



# Audio Programming in C++

## The Beginner Level

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Håkan Blomqvist

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*Dedicated to all my friends around me. You know who you are.*

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# Introduction

Welcome to the beginner book about audio Programming. In this book we will get to learn the different types of sound synthesis there are (a least the popular ones) and go through some example code using the different synthesis techniques. All code examples are written using C++20 and Clang++ as a compiler and linker.

You'll find all code examples in Appendix A and all code will include what to type in a terminal to compile and link the program. After compilation you can run the program by typing `./` (dot slash) before the name of the program. This book assumes that you already know how to program in C++ and what a compiler and linker is. Fokus for the book is about the different synthesis methods and examples of them in C++20. The examples are not optimized for real time playback just to show how the synthesis method works. Therefore the sounds generated will be saved as Wave och Aiff file format. The examples will also be visualized as waveforms in a .png image. Just so that we can both hear and see the result.

Although this book is about audio programming in C++20, we suggest getting hardware synths or building hardware synths/modules to understand the software side of things better. Especially for Virtual Analog Synthesis where you emulate/simulate the components in the hardware synths. A modular synth is what I recommend and I know that it can be expensive to build your own. Playing around with software like the free VCV Rack (from [VCV Rack](https://vcvrack.com/)<sup>1</sup>) instead of your own modular is an alternative, although only for the sound and not so much for the hardware.

## Why C++20?

Well, you could probably compile and run the examples as C++17. But as C++20 is the latest standard, then that should be used. So when C++23 is out, this book will be updated to support that, if needed.

## So what is the definition of synthesis?

It's a word used in just two major contexts: the creation of chemical compounds and production of electronic sounds.

There are a large number of types of synthesis. Like Texture synthesizers, Video synthesizers, Color synthesizers, and Speech synthesizers. None of these will be discussed in this book.

Cambridge Dictionary says regarding Synthesis: “the act of combining different ideas or things to make a whole that is new and different from the items considered separately.”

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<sup>1</sup><https://vcvrack.com/>

In this book Synthesis is about Sound Synthesis, as in the process of producing sound. Sound where we can reuse existing sounds by processing them, or we can generate sound electronically or mechanically. Sound synthesis can use mathematics, physics or even biology; and it brings together art and science in a mix of musical skill and technical expertise.

## What was the first Sound Synthesizer?

In 1896, Telharmonium was realized and patented in 1897. Telharmonium was also called Dynamophone and used tone wheels to generate musical sounds as electrical signals by Additive synthesis.

Inventor behind the Telharmonium is Thaddeus Cahill, the eldest son of Dr. Timothy and Ellen Harrington Cahill, both immigrants from Ireland. Thaddeus was born in Iowa, USA on June 18:th 1867. He studied physics of music at Oberlin Conservatory in Oberlin, Ohio. He graduated from Columbian Law school. Thaddeus started to invent things at age 13 or 14, after Bell Company had refused to sell him telephone instruments to experiment with.

At age 18, he had invented and patented mechanism for typewriters. In 1893, at 26 years of age, he was struck by a grand new idea. The idea was to use electrical dynamos to build up complex tones such as instruments. An invention like this would replace the piano, organ, and violin as the preferred parlor instruments. A Piano could not be kept in tune. Furthermore, the volume of a note or chord was soon lost. The organ was better in these respects, but had limited power of expression. The violin did not possess these problems, but it had little chord capacity. So Cahill's idea would fix that and could even have the power to send music anywhere electricity could be transported. Over a network of telephone lines, Cahill could transmit music from his central station to tens of thousands of places at once.

Cahill wanted Telharmonium music to be broadcast into hotels, restaurants, theaters, and even houses via the telephone line.

He built three versions: the first version weighed 7 tons. The second and third version weighed almost 210 tons. Each was a considerable advancement over the features of its predecessor. Telharmonium had a hefty price tag of \$200,000 (approximately \$5,514,000 today).

Performances of Telharmonium were made at the "Telharmonic Hall", on 39:th street and Broadway. Mark Twain was among the appreciative of its audiences.

The Telharmonium foreshadowed modern electronic musical equipment in a number of ways. For instance, its sound output came in the form of connecting ordinary telephone receivers to large paper cones—a primitive form of loudspeaker. Cahill was noted for saying that electromagnetic diaphragms were the most preferable means of outputting its distinctive sound. There are sadly no known recordings of music made with the Telharmonium.

Telharmonium tones were described as "clear and pure" — referring to the electronic sine wave tones it was capable of producing. However, it was not restricted to such simple sounds. Each tonewheel of the instrument corresponded to a single note, and, to broaden its possibilities, Cahill added several extra tonewheels to add harmonics to each note. This, combined with organ-like stops and multiple

keyboards, as well as a number of foot pedals, meant that every sound could be sculpted and reshaped — the instrument was noted for its ability to reproduce the sounds of common orchestral woodwind instruments such as the flute, bassoon, clarinet, and also the cello. The Telharmonium.

Mr. Cahill's Telharmonium company was declared bankruptcy in 1914.

Thaddeus Cahill passed at age 66 (12:th of April, 1934) in New York City, USA. The last Telharmonium to be scrapped was the 7 ton version, in 1962.

## What is sound?

A sound is a continuous and regular vibrating of air molecules that is heard when they reach a person's ear.

Air molecules are the molecules that make up air, which include particles of oxygen and nitrogen. Air is a materia and even though it is invisible, and it is constantly effecting the world around us.

Audio reproduction can be made with speakers and headphones. The sound moves in pressure waves. When air molecules are compressed and rarified fast enough, we hear it as sound. The faster the air pressure changes, the higher the "frequency" of the sound we hear. When a speaker moves back and forth it pushes on air molecules which changes the air pressure and creates sound waves.

Speakers work by converting electrical energy into mechanical energy (motion). The mechanical energy compresses air and converts the motion into sound energy or sound pressure level (SPL).

When an electric current is sent through a coil of wire, it induces a magnetic field. In speakers, a current is sent through the voice coil which produces an electric field that interacts with the magnetic field of the permanent magnet attached to the speaker. Like charges repel each other and different charges attract. As an audio signal is sent through the voice coil and the musical waveform moves up and down, the voice coil is attracted and repelled by the permanent magnet.

This makes the cone that the voice coil is attached to move back and forth. The back and forth motion creates pressure waves in the air that we perceive as sound.

Frequency response is how loud the output of a speaker will be at different frequencies. A typical test for frequency response sends out a sweep of frequencies from the bass to the mid-range, and up to the treble range to see if the sound from the speaker is the same in all these areas. The ideal frequency response for a speaker is very flat. This means the speaker would be the same level at low frequency as it is in the mid-range or highs.

Connecting a computer to a device that is digital is quite straightforward.

However, when analog devices are involved (speakers, headphones or microphones), interfacing becomes much more complex. What is needed is a way to electronically translate analog signals into digital (binary) quantities, and vice versa.

For input of a analog signal, an analog-to-digital converter, or ADC is used. For speakers or headphones (if not built in) a digital-to-analog converter, or DAC, is used.



An ADC inputs an analog electrical signal such as voltage or current and outputs a binary number. A DAC, on the other hand, inputs a binary number and outputs an analog voltage or current signal.

## What is Timbre?

Timbre, also called timber, quality of auditory sensations produced by the tone of a sound wave.

The timbre of a sound depends on its wave form, which varies with the number of overtones, or harmonics, that are present, their frequencies, and their relative intensities.

In music timbre is the characteristic tone color of an instrument or voice, arising from reinforcement by individual singers or instruments of different harmonics, or overtones (q.v.), of a fundamental pitch. Extremely nasal timbre thus stresses different overtones than mellow timbre. The timbre of the tuning fork and of the stopped diapason organ pipe is clear and pure because the sound they produce is almost without overtones. Timbre is determined by an instrument's shape (e.g., the conical or cylindrical pipe of a wind instrument), by the frequency range within which the instrument can produce overtones, and by the envelope of the instrument's sound. The timbre of spoken vowels or of a singing voice is modified by constricting or opening various parts of the vocal tract, such as the lips, tongue, or throat.

## Synthesis Chapters in the book

This is a beginner level book and aimed at that. As with many things in life — we need to learn how to stand before we can learn how to walk — Learn to walk before we can learn how to run and so forth.

Additive Synthesis

Subtractive Synthesis

Formant Synthesis

Granular Synthesis

(FM) Frequency Modulation Synthesis

Linear Arithmetic Synthesis

Phase Distortion Synthesis

Scanned Synthesis

Vector Synthesis

Virtual Analog Synthesis

Wavetable Synthesis

Physical modelling

Appendix B: Audio Engineering Terminology (0 - 9 - A - Z)

You need to know Audio Engineering Terminology in order to know what is what when developing software for Audio Engineers.

The Beginner Level Book in Audio Programming is the first book in a series of at least three in total. Part two and three are work in progress and TBA

# The Physics of string, wind and percussion Instruments

Instruments have been classified in various ways, some of which overlap. The Chinese divide them according to the material of which they are made—as, for example, stone, wood, silk, and metal. Writers in the Greco-Roman world distinguished three main types of instruments: wind, stringed, and percussion. This classification was retained in the Middle Ages and persisted for several centuries: it is the one preferred by some, with the addition of electronic instruments, at the present day.

## Strings

Pythagoras experimented with the tones produced when plucking strings of different lengths. He found that some specific ratios of string lengths created pleasing combinations (“harmonies”) and others did not. Based on his careful observations, Pythagoras identified the physics of intervals, or distances between notes, that form the primary harmonic system which is still used today.

The mathematical structure of harmonic sound begins with a single naturally occurring tone, which contains within it a series of additional frequencies above its fundamental frequency (“overtones”), of which we are normally unaware on a conscious level. Within this harmonic or overtone series, there is a mathematical relationship between the frequencies – they are specific integer multiples of each other. For example, if the slowest frequency (the “fundamental”) were 100 Hz, then the overtones would be  $2 \times 100$  (200 Hz),  $3 \times 100$  (300 Hz) and so forth. (The overtone series is often referred to as harmonics.)

Pythagoras observed several ratios of sound wave frequencies and the corresponding intervals between them, including 4:3 (known to musicians as the interval of a perfect fourth, or two pitches that are five semitones apart from each other) and 3:2 (a perfect fifth, seven semitones apart). Note that pitch is the frequency or rate of vibration of a physical source such as a plucked string.

The most prominent interval that Pythagoras observed highlights the universality of his findings. The ratio of 2:1 is known as the octave (8 tones apart within a musical scale).

When the frequency of one tone is twice the rate of another, the first tone is said to be an octave higher than the second tone, yet interestingly the tones are often perceived as being almost identical. For example, a woman’s voice may fluctuate around 220 Hertz while a man’s voice is around 110 Hertz, approximately half the frequency of the woman’s. However, if they sing together, it may sound as though they are singing the same melody together in unison, even though they are actually an octave apart. This 2:1 ratio is so elemental to what humans consider to be music, that the octave

is the basis of all musical systems that have been documented – despite the diversity of musical cultures around the world.

## Wind

Wind instruments are vibrating column of air in pipes. As mentioned about strings, that they are stretched between two points and prefers to vibrate at certain frequencies that are called Harmonics. We noted that strings in the second harmonic oscillates at the twice frequency than the first and that the third oscillates at three times the frequency than the first. In theory, a perfect stretched string can generate the complete harmonic series from 1 to infinity, just not in reality.

The amplitude in each vibrating string will determine the shape of the audio waveform produced. If the amplitude of any harmonic 'n' is 1/n times that of the fundamental, we obtain a sawtooth wave. Alternatively, if the amplitude of any odd-numbered harmonic 'n' is 1/n times that of the fundamental, but all the even harmonics are missing, we obtain a square wave.

An open rigid pipe that is not suspended in vacuum, has a column of air inside it. It might appear that the air can enter and exit without anything special happening. But, if you blow across the top of such pipe, it will generate a pleasant note. You may therefore assume that the air inside it is oscillating in such way as to produce a harmonic series.

By blowing in one end of the pipe, you create a pulse of higher pressure at one end. The high-pressure pulse passes through the pipe until it reaches the far end, at which the most of the energy is selected back into the pipe. It's almost as if it had bounced off an invisible wall. This bouncing back and fourth will continue until all the energy is dissipated.

The reason pipes produce a harmonic series is very similar to the reason why strings do. A string has to be fixed at both ends for harmonic motion to occur. It's called a boundary condition. The pipe has an analogous condition. The pressure of the atmosphere outside the pipe must be the same as the inside pressure. So by the right type of blowing – the maximum positive pressure in the pipe will occur at the center, as will the maximum negative pressure of the reflected pulse. These are the conditions under which a standing wave occur. The fundamental frequency wavelength for a pipe is the twice length of the pipe (it has to move back and fourth)

So, if the pipe is 0.34 meters in length, and given that the speed of sound is 340 meters per second, the pulse in the pipe will travel down the pipe in a thousandth of a second (1 millisecond). The pulse will then move back up, which also takes a thousands of a second. So the period is 2 milliseconds and the frequency of the fundamental is therefore 500 HZ.

The means to give energy to the column of air, and to sustain the standing wave within a pipe is by blowing into a mouthpiece. With the mouthpiece a stream of pressure pulses are formed. The lips around the mouthpiece are working as a valve, creating short pulses go high-pressure air. If the timing go these pulses is appropriate to the length of the pipe (wavelength) a note will be produced. The timing is regulated by change in tension go the lips and the air pressure in the mouth.

An internal duct flute like the Recorder generates more of a sawtooth wave or a triangle wave while a woodwind family instrument with a single-reed mouthpiece, like the Clarinet, is more of a square wave ('hollow' sound).

The pipe and the string is also similar in that both can form both a saw and a square wave.

## Percussion

String and wind instruments are more of a one dimensional oscillators (a string is stretched between two points and a wind instrument is a pipe with vibrating column of air).

A drum skin has two dimensions, as it's a circular membrane that is stretched with an equal tension at all points, and which is fixed at all points around a circumference. Drums are a different type of oscillator than the string and wind instruments. The equation to describe a vibrating membrane is quite complex and harder synthesize the string and wind instruments.

The fundamental frequency of a vibrating circular membrane is excited by hitting the drum skin exactly in its centre (this is called the  $w_{01}$  mode of the membrane).

If you hit the drum skin away from the centre, the skin will vibrate in completely different ways, just to make things even more complex.

To get the second harmonic and third and so on, you can't do as with a string, and get an overtone of exactly three times the frequency of the fundamental. By using the Bessel function we can calculate  $w_{02}$  and  $w_{03}$ , as of the fundamental and get the second and third harmonic.

Other factors are for the tension across the skins surface. All this makes drums really hard to synthesize well, and this is why samples of drums or a recording of real drums.

# Additive Synthesis

In additive synthesis, you start out with nothing and build a sound by combining multiple sine waves of differing levels and frequencies. As more sine waves are combined, they begin to generate additional harmonics. In most additive synthesizers, each set of sine waves is viewed and used much like an oscillator.

Depending on the sophistication of the additive synthesizer you are using, you will either have individual envelope control over the level and pitch of each sine wave, or you will be limited to envelope control over groups of sine waves—one envelope per sound and its harmonics, or all odd or all even harmonics, for example. In practical terms, working with groups of related harmonics is the best approach due to the mathematical relationships between them and the impact this has on the overall tone when adjusting them en masse, rather than individually.

Additive synthesis is to be considered as the reverse approach to subtractive synthesis.

A true additive synthesizer will allow us to manipulate individually the amplitudes of 32, 64, 128, or even 256 harmonics, and that's something that no pre-patched analogue synthesizer can do. That's why modern-day implementations of additive synthesis are mainly digital.

Additive synthesis is also called Fourier Synthesis in honor of Joseph Fourier, the mathematician who discovered the basis of what we call Fourier analysis - the mathematical method used to break up a sounds into sine waves - and with Fourier analysis - building them back up again.

Additive Synthesis is not limited to using only sine waves, though you can make all types of sound with a sine wave. Square waves, sawtooth waves, or more complex waves like pulse width modulated wave or the outputs of a ring modulator to create extremely time varying spectra.

Also, to make an additive synthesized sound more playable for music, a velocity- and pressure-sensitivity will give expression and character.

For instance the sounds of orchestral instruments such as flutes and trumpets has a residual element of noise. This noise may not be very loud or intrusive, but it's there nonetheless. So, many synthesized sounds will remain unconvincing if they lack a little noise within them. By adding a noise generator we remedy that. Though a 'white' or 'pink' noise is far from that of what get's filtered by the nature of the instrument in itself. So, despite everything, we need at least one filter with it's own contour generator to ensure that the noise color changes realistically over time. The noise generator will also need a VCA and its associated contour generator.

If this analysis seems a little arcane, it isn't. In fact, this extension to pure additive synthesis even has a name: if the analysis is performed beforehand it's called Spectral Modeling Synthesis. Without the signal analysis, you could just call it the 'sinusoids plus noise' model of sound generation.

Additive synthesis most directly generates sound by adding the output of multiple sine wave generators. Alternative implementations may use-pre-computed wavetables or the Inverse Fast Fourier transform.

The timbre of musical instruments can be considered in the light of Fourier theory to consists of multiple harmonic or inharmonic partials or overtones. Each partial is a sine wave of different frequency and amplitude the swells and decays over time due to modulation from an ADSR envelope or low frequency oscillator.

The sounds that are heard in everyday life are not characterized by a single frequency. Instead, they consist of a sum of pure sine frequencies, each one at a different amplitude. When humans hear these frequencies simultaneously, we can recognize the sound. This is true for both “non-musical” sounds (e.g. water splashing, leaves rustling, etc.) and for “musical sounds” (e.g. a piano note, a bird’s tweet, etc.). This set of parameters (frequencies, their relative amplitudes, and how the relative amplitudes change over time) are encapsulated by the timbre of the sound. Fourier analysis is the technique that is used to determine these exact timbre parameters from an overall sound signal; conversely, the resulting set of frequencies and amplitudes is called the Fourier series of the original sound signal.

Noise has, by definition, no harmonic structure, although it may be present only in specific parts of the spectrum: colored noise. So any noise which is present in a sound will appear as random additional frequencies within those bands, and whose level and phase are also random.

Synthesizers using Additive Synthesis are Telharmonium, the Fairlight CMI, Synclavier II, Kawai K5 and K5000S, and the Hammond organ from 1935 with a tone tonewheel.

## Additive Synthesis in C++

In the example code presented in the file `01-additive.cpp` we set up and generate 32 sine oscillators. Oscillator #1 starts at 55 Hz (the fundamental) and the other 32 oscillators are in 55 Hz steps from the fundamental (31 harmonics) up to 1760 Hz. Each Sine oscillator has different gain to change the amplitude of the sine wave before adding with the mixer.

Each Sine Oscillator are saved to disk as both Aiff and a PNG representation. Oscillator 33 is a White Noise oscillator and applied last to add some noise to the sound. Normalization is also done to normalize the amplitude from 1.0 to -1.0. At the end we shape mixed sound after an ADSR Envelope that we also generate with the help of the `Envelope.hpp` class.

If you prefer saving the audio as Wave instead of Aiff, then change line number 523 from:

```
a.save(path+filename,AudioFileFormat::Aiff);
```

To:

```
a.save(path+filename,AudioFileFormat::Wave);
```

## Additive Synthesis with a KAWAI K5

KAWAI K5 Synthesizer 1987 - Sounds Presets Demo (NO TALKING, ONLY PLAYING) 80s Vintage synthesizer<sup>2</sup>

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<sup>2</sup><https://www.youtube.com/watch?v=jXQJBL3iPd0>

# Subtractive Synthesis

Subtractive synthesis is based around the idea that real instruments can be broken down into three major parts: a source of sound, a modifier (which processes the output of the source) and some controllers (which act as the interface between the performer and the instrument). This is most obviously apparent in many wind instruments, where the individual parts can be examined in isolation.

There are numerous differences between synthesis methods, but most follow a fundamentally similar architecture and signal flow that is based on subtractive synthesis principles.

According to legend, when Michelangelo was asked how he managed to carve David out of a block of stone, he replied, "I just cut away everything that doesn't look like David."

In essence, this is how subtractive synthesis works. You filter, or cut away, parts of the sound that you don't want to hear. In other words, you subtract parts of the frequency spectrum, consisting of the fundamental tone and associated harmonics.

Subtractive synthesis assumes that an acoustic instrument can be approximated with a simple oscillator that can produce waveforms with different frequency spectrums. The signal is sent from the oscillator to a filter that represents the frequency-dependent losses and resonances in the body of the instrument. The filtered (or unfiltered) signal is shaped over time by the amplifier section of the synthesizer.

The distinctive timbre, intonation, and volume characteristics of a real instrument can theoretically be recreated by combining these components in a way that resembles the natural behavior of the instrument you are trying to emulate.

In reality, however, subtractive synthesizers aren't perfect at emulating real-world instruments. No synthesized clarinet is going to be mistaken for a real clarinet. The true strength of subtractive synthesis is that it offers a unique sound palette on its own.

There are a few different kinds of filters, but most synths rely on two key ones - a Low Pass Filter (LPF) and a High Pass Filter (HPF).

The Low Pass Filter allows the low frequencies to pass through, cutting off the high frequencies and creating bass sounds. The High Pass Filter does the opposite. Turning the CUTOFF knob shifts the point at which the filter starts to act on the sound.

A filter does not just stop the sound abruptly, it slopes down from the original volume. The Slope is measured in decibels (dB) per octave, which is why you might see a -12dB or -24dB filter on a synthesizer. The higher the number, the steeper the slope and the stronger the filter.

Along with the Cutoff, the other key part of a filter is the Resonance. Most filters have a Resonance or Q control. Resonance occurs when the sound in the same range as the cutoff frequency is routed back to the filter, creating feedback.



At very high levels, this feedback can cause the filter to self-oscillate and generate a Sine wave. Turning up the Resonance results in brighter and harsher tones around the cutoff frequency and can be used for a range of techniques including to "squeal" the synth, or create a classic way effect.

While less common, you also sometimes get Bandpass and Band Reject (or Notch) filters. A bandpass is a combination of both High and Low Pass, leaving only the central part of the audio unfiltered. Band Reject is the opposite, leaving the higher and lower frequencies unfiltered.

Subtractive synthesis uses different types of oscillators, some are "lighter" and some are "darker".

The four main types of oscillators are: Sine, Sawtooth, Square and Triangle.

A Sine wave is one of the most basic waveforms. Its edges are smooth, like ripples in a pond, with no sharp changes.

This produces a smooth, mellow sound. Sine waves are often used to recreate pipe sounds, like a flute or an organ, or for smooth pads.

The Sawtooth, named as it looks like the teeth on a saw. Sawtooth waves have very sharp points and abrupt changes, which create a strong "buzz" sound.

Because of this, a Sawtooth waveform has harmonic frequencies at regular intervals that get progressively quieter.

It's these harmonic frequencies that make Sawtooth waves sound rich and full, great for powerful synth bass and lead sounds.

Square waves have a sound that is rich in harmonics, that is not as "buzzy" as a Sawtooth, but not as smooth as a Sine. They have half as many harmonic frequencies as a Sawtooth does, which repeat every second cycle.

Square waves are a particular type of pulse-wave. As the name suggests, the signal pulses on and off. Pulse waves are commonly used for pulse-width modulation (PWM), which was used in early synthesizers to replicate two oscillators interacting with each other.

Square waves are often described as sounding "hollow" or "nasal". This means that they are good for creating wind instruments, like a clarinet.

Triangle waves lie between a Square wave and a Sine wave. Similar to a Square wave, they contain the odd harmonics of the original frequency. However, the volume of these harmonics drops much more quickly.

This causes the waveform to look and sound much closer to a Sine wave, but still retain some of the "buzzing" quality of a Square wave.

In the Additive synthesis part we mentioned that we used an ADSR Envelope with `Envelope.hpp`. ADSR stands for Attack, Decay, Sustain and Release.

The Envelope is key in determining what sound you want to create. Consider for instance, the difference between a drum sound and a violin sound.

The drum has a sharp, sudden increase in volume (Attack), with almost no Sustain and quite a short Release time.

The violin has a longer, slower Attack, building to maximum volume, a long Sustain and the sound tails off slowly.

Envelopes don't just apply to the amplifier, many synthesizers have an envelope on the filter or the pitch. If you're using modular synths, you can set an envelope to virtually any parameter you like.

For example, on a filter envelope, the amplifier works similarly to if you were turning the cutoff knob over time, gradually turning it up and back again.

To animate the sound, An LFO is used. LFO (Low Frequency Oscillator) is exactly what it sounds like. It's another oscillator, with similar waveforms to the ones we spoke about earlier in this piece. However, it occurs at such a low frequency that you can't hear it.

Rather than using an LFO to create a tone, you can use it to manipulate other parameters. Think of it as automatic knob twiddling.

LFOs have a Rate control that change the speed at which it oscillates and Depth controls for each parameter that you can affect on the synth.

These depth controls allow you to set the LFO to twiddle your chosen parameter by varying amounts, in a positive or negative direction.

Many synths include controls to set the LFO to Pitch (which creates Vibrato), Volume (which creates Tremolo) or Filter Cutoff.

Having the LFO control filter cutoff is a highly popular synthesis technique used to modulate tone and can be used to create the wah-wah sound mentioned earlier. At extreme amounts, it can produce a "wobble" that is popular in dubstep basslines.

Three examples of analog synthesizers using Subtractive Synthesis are: Moog Minimoog Model D, Sequential Circuits Prophet-5, and Roland Jupiter-8.

## Subtractive Synthesis in C++

`02-subtractive.cpp` is loosely inspired by Minimoog Model D and has the `MoogFilter.hpp` included. `MoogFilter` class is an implementation of a 4-pole ladder filter (a 24dB per Octave slope) for you to play around with.

Just as the Minimoog we have 7 oscillators to pick from (actually 5 oscillators; but one oscillator changes from triangle to sawtooth instead. + the are a noise oscillator). But in this code example you can pick freely between: `Triangle`, `Inverse Sawtooth`, `Saw`, `Square`, `30% Pulse Width`, `20% Pulse Width` and `White Noise`.

There are even a LFO for use to modulate a variable in the filter or anything else.

Anyway, the code is straightforward with it's three oscillators, 4-Pole Filter, Mixer and Envelope.

# Subtractive Synthesis with a Sequential Circuits Prophet 5

Sequential Circuits Prophet 5 Sounds<sup>3</sup>

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<sup>3</sup><https://www.youtube.com/watch?v=YHNRR01xK68>

# Formant Synthesis

Formants are one of the hardest concept to grasp in acoustics. They are ranges of frequencies that are amplified by the shape of your vocal tract, the shape of your throat and mouth that give your particular voice its distinct characteristics and also is responsible for vowel (A, E, I, O, U) sounds.

In theory, a voice produce an infinite number of formants. Only four or five are relevant to the way we hear things. Higher ones are out of bound of human hearing. The most important are the first two formants. The lowest two. Those are the formants responsible for vowel sounds and the third and upwards are responsible for timbre or the color of your voice.

The formants in your voice are at the same frequency even if you whisper. By whispering you use noise to make a sound through you vocal cords (vocal folds) the lowest two formants are at the same frequency.

A vocoder works by taking the formants from your voice and adding it to a sound from a synthesizer.

Formant synthesis works by using a formant filter that adds the lowest four to five the vowel frequency as peaks to the synthesized sound to make it kinda talk.

We all push air over our vocal cords to generate a pitched signal with a definable fundamental and multiple harmonics. We can all tighten and relax these cords to change the pitch of this signal. Furthermore, we can all produce vocal noise. The pitched sounds are generated deep in our larynx, so they must pass through our throats, mouths, and noses before they reach the outside world through our lips and nostrils. And, like any other cavity, this 'vocal tract' exhibits resonant modes that emphasize some frequencies while suppressing others. In other words, the human vocal system comprises a pitch-controlled oscillator, a noise generator, and a set of band-pass filters. The resonances of the vocal tract, and the spectral peaks that they produce, are called 'formants', a word derived from the Latin 'formare', meaning 'to shape'.

Measurement and acoustic theory have demonstrated that the centre frequencies of these formants are related to simple anatomical properties such as the length and cross-section of the tube of air that comprises the vocal tract. And, since longer tubes have lower fundamentals than shorter ones, it's a fair generalization to suppose that adult human males will have deeper voices than adult human females or human children.

Example of a synthesizer based on Formant Synthesis are the Yamaha FS1R.

## Formant Synthesis in C++

`03-formant.cpp` code example is straightforward. It will generate a vowel sound based on the table of Formant 1 and 2 in the `generateformant` function.

## Formant Synthesis with a Yamaha FS1R

Yamaha FS1R | The Hidden Treasure!<sup>4</sup>

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<sup>4</sup><https://www.youtube.com/watch?v=Ul-V498IpbQ>

# Granular Synthesis

The basic premise behind granular synthesis is that a sound can be broken down into tiny particles, or grains. In many respects, granular synthesis is similar to wavetable synthesis, but it works on a much finer scale. This method is ideal for creating constantly evolving sounds and truly unique tones.

A number of interesting manipulations are possible using this synthesis method.

- Time-stretching. Grains can be sent out at a faster or slower rate than their counterparts in the original sample, allowing faster or slower playback—without the changes to pitch that occur with traditional sample playback. You can even “freeze” a sample at a certain position by extracting multiple grains from a single point. On this latter point, you could repeat a drum hit “grain” multiple times in a time-stretched loop to create a different drum pattern, for example.
- Pitch-shifting. Modifications to the pitch of each grain allow you to vary the pitch of a sample without affecting its timing. By modulating the pitch or pan position of each grain, you can also create spatial and “blurring” effects.
- You can also scramble the order in which grains are played back to produce effects ranging from mild fuzziness to extreme mangling.

Greek composer Iannis Xenakis is known as the inventor of the granular synthesis technique.

Canadian composer Barry Truax was one of the first to implement real-time versions of this synthesis technique.

## Granular Synthesis in C++

In the `04-granular.cpp` code example we load a generated sound from a previous synthesis tech, we extend the length of the loaded sample and the randomly adding grains from the loaded sample to the extended version.

## Granular Synthesis with a Portal by Output

[Portal by Output - Granular FX Plugin - First Look<sup>5</sup>](#)

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<sup>5</sup><https://www.youtube.com/watch?v=DScdFZcR2QE>

# Frequency Modulation (FM) Synthesis

FM synthesis uses a modulator oscillator and a sine wave carrier oscillator. The modulator oscillator modulates the frequency of the waveform generated by the carrier oscillator within the audio range, thus producing new harmonics. These harmonics are known as sidebands.

Where there is a mathematical relationship between the carrier and modulator waveforms, the sound produced is harmonic. Where the modulator is a non-integer multiple of the carrier waveform, inharmonic sidebands are produced, resulting in an inharmonic sound.

Typically, FM synthesizers don't incorporate a filter. You can generate some subtractive synthesizer style sounds with FM synthesis, but it is difficult to recreate the sound of a resonant subtractive synthesizer filter using this method. FM synthesis is extremely good, however, at creating sounds that are difficult to achieve with subtractive synthesizers—sounds such as bell timbres, metallic tones, and the tine tones of electric pianos. Another strength of FM synthesis is punchy bass and synthetic brass sounds.

FM synthesis was developed in the 1960s at Stanford University, California, by John Chowning, who was trying to create sounds different from analog synthesis. His algorithm was licensed to Japanese company Yamaha in 1973.

Yamaha's engineers began adapting Chowning's algorithm for use in a commercial digital synthesizer, adding improvements such as the "key scaling" method to avoid the introduction of distortion that normally occurred in analog systems during frequency modulation, though it would take several years before Yamaha released their FM digital synthesizers.

The most known and used FM based synthesizer is the Yamaha DX7.

## Frequency Modulation (FM) Synthesis in C++

In the `05-fm.cpp` example there is a function called `generatefrequencymodulation` that is one carrier and one modulator. By adding more carriers and modulators we can generate more complex sounds. By adding an Envelope Generator to each Carrier and Modulator will help to shape sounds with a new timbre.

## Frequency Modulation (FM) Synthesis with a Yamaha DX7

The sounds of 1983 | Yamaha DX7 demo <sup>6</sup>

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<sup>6</sup><https://www.youtube.com/watch?v=11bWvQaQhrM>

# Linear Arithmetic Synthesis

Linear arithmetic synthesis, or LA synthesis, is a means of sound synthesis invented by the Roland Corporation when they released their D-50 synthesizer in 1987.

LA synthesis combines traditional subtractive synthesis with PCM-based samples. The term linear arithmetic refers to synthesis that puts sounds together in a timeline. Typically a PCM transient begins a note, which is then continued with a subtractive synthesis prolongation.

This technology first appeared in 1987, in the Roland D-50 synthesizer. At the time, re-synthesizing samplers were very expensive, so Roland set out to produce a machine that would be easy to program, sound realistic, and still sound like a synthesizer.

## Linear Arithmetic Synthesis in C++

In the `06-1a.cpp` example we load a sample that is used for the `Attack` part of the envelope and the rest is generated with `subtractive synthesis`.

## Linear Arithmetic Synthesis with a Roland D-50

[Roland D-50 | The King is back!](#)<sup>7</sup>

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<sup>7</sup><https://www.youtube.com/watch?v=zfMkR3JHAWo>



# Phase Distortion Synthesis

Phase distortion (PD) synthesis is a synthesis method introduced in 1984 by Casio in its CZ range of synthesizers. In outline, it is similar to phase modulation synthesis as championed by Yamaha Corporation (under the name of frequency modulation), in the sense that both methods dynamically change the harmonic content of a carrier waveform by influence of another waveform (modulator) in the time domain. However, the application and results of the two methods are quite distinct.

This tech was introduced in 1984 by Casio in its CZ range of synthesizers.

## Phase Distortion Synthesis in C++

`07-pd.cpp` is an example of Phase Distortion Synthesis. The function `generatephasedistortionwave` contains all the logic for the synthesis method.

## Phase Distortion Synthesis with a Casio CZ-1000

[Casio CZ-1000 | The Cosmo Synthesizer](#)<sup>8</sup>

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<sup>8</sup><https://www.youtube.com/watch?v=NxWiuJ8R1-Y>

# Scanned Synthesis

Scanned synthesis involves a slow dynamic system whose frequencies of vibration are below about 15 Hz. The ear cannot hear the low frequencies of the dynamic system. So, to make audible frequencies, the “shape” of the dynamic system, along a closed path, is scanned periodically. The “shape” is converted to a sound wave whose pitch is determined by the speed of the scanning function. Pitch control is completely separate from the dynamic system control. Thus timbre and pitch are independent. This system can be looked upon as a dynamic wave table. The model can be compared to a slowly vibrating string, or a two dimensional surface obeying the wave equation.

## Scanned Synthesis in C++

generatescannedwave function in 08-scanned.cpp contains an example of scanned synthesis.

## Scanned Synthesis with a Qu-Bit Scanned

[Qu-Bit Scanned, A Scanned Synthesis Eurorack Oscillator<sup>9</sup>](#)

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<sup>9</sup>[https://www.youtube.com/watch?v=p\\_AboqfHAQE](https://www.youtube.com/watch?v=p_AboqfHAQE)

# Vector Synthesis

Vector Synthesis is a type of audio synthesis introduced by Sequential Circuits in the Prophet VS synthesizer in 1986.

Vector synthesis provides movement in a sound by providing dynamic cross-fading between four sound sources. The four sound sources are conceptually arranged as the extreme points of X and Y axes, and typically labelled A, B, C and D. A given mix of the four sound sources can be represented by a single point in this 'vector plane'. Movement of the point provides sonic interest and is the power of this technique. Mixing is frequently done using a joystick, although the point can be controlled using envelope generators or LFOs.

The term simply refers to the ability to crossfade between 4 timbres via joystick.

## Vector Synthesis in C++

In the `09-vectorsynth.cpp` example we load 4 samples and cross fade between them.

## Vector Synthesis with a Sequential Circuits Prophet VS

[Sequential Circuits Prophet VS Vintage Vector Synthesizer](#)<sup>10</sup>

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<sup>10</sup>[https://www.youtube.com/watch?v=H4gSZ7\\_AWOk](https://www.youtube.com/watch?v=H4gSZ7_AWOk)

# Virtual Analog (VA) Synthesis

Using a Digital Signal Processor as its own ASIC or one implemented in an FPGA can help emulate/simulate the analog hardware components (resistors, capacitors...) that produces the synthesized sound.

This type of synthesis is also called analog modeling synthesis. Analog modeling synthesizers can be more reliable than their true analog counterparts since the oscillator pitch is ultimately maintained by a digital clock, and the digital hardware is typically less susceptible to temperature changes.

While analog synthesizers need an oscillator circuit for each voice of polyphony, analog modeling synthesizers don't face this problem. This means that many of them, especially the more modern models, can produce as many polyphonic voices as the CPU on which they run can handle.

Modeling synths also provide patch storage capabilities and MIDI support not found on most true analog instruments. Analog modeling synthesizers that run entirely within a host computer operating system are typically referred to as analog software synthesizers.

The term was not used until the 1990s when the Nord Lead came out.

The most popular and known VA synthesizer is the Access Virus TI.

## Virtual Analog (VA) Synthesis in C++

(No example is included as this synthesis method is too complex for a beginner level and therefore out of scope.)

## Virtual Analog (VA) Synthesis with a Access Virus TI

[Access Virus TI - Patches<sup>11</sup>](#)

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<sup>11</sup>[https://www.youtube.com/watch?v=-S\\_wzWBxMzA](https://www.youtube.com/watch?v=-S_wzWBxMzA)

# Wavetable Synthesis

Wavetable synthesis is fundamentally based on periodic reproduction of an arbitrary, single-cycle waveform. In wavetable synthesis, some method is employed to vary or modulate the selected waveform in the wavetable. The position in the wavetable selects the single cycle waveform. Digital interpolation between adjacent waveforms allows for dynamic and smooth changes of the timbre of the tone produced. Sweeping the wavetable in either direction can be controlled in a number of ways, for example, by use of an LFO, envelope, pressure or velocity.

Wavetable synthesis isn't well-suited for emulating acoustic instruments. It is noted for producing constantly evolving sounds; harsh and metallic, or bell-like sounds; punchy basses; and other digital tones.

Wavetable synthesis is also called "Table-lookup synthesis".

The German PPG company did a lot of research into Wavetable Synthesis in the 1970s. The PPG wave 2.2 Wavetable Synthesizer from 1982 are for instance a known synthesizer using the Wavetable synthesis method.

## Wavetable Synthesis in C++

11-wavetable.cpp has a Wavetable Synthesis function called `generatewavetable`, that is one example of the method.

## Wavetable Synthesis with a PPG wave 2.2

[PPG wave 2.2 Synthesizer \(1982\) - RetroSound soundscapes - sound demo](#)<sup>12</sup>

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<sup>12</sup>[https://www.youtube.com/watch?v=9bZ\\_VGFt5X0](https://www.youtube.com/watch?v=9bZ_VGFt5X0)

# Physical Modeling

Also known as Component modeling synthesis, this synthesis method uses mathematical models to simulate instruments. Parameters are used to describe the physical characteristics of an instrument, such as the materials the instrument is made of, the dimensions of the instrument, and the environment it is played in—under water, or in the air, for example. Equally important are descriptions of how the player would interact with the instrument—whether it is played by blowing; by plucking, bowing, or strumming strings; by hitting it with sticks; by placing fingers on sound holes, and so on.

To model a drum sound, for example, the following aspects need to be taken into account. Of primary importance is the actual drum strike—how hard it is and whether the drumhead is struck with a wooden stick, a mallet, a beater, and so on. The properties of the drumhead (the skin or membrane) include the kind of material, its degree of stiffness, its density, its diameter, and the way it is attached to the shell of the drum. The volume of the drum cylinder itself, its material, and the resonance characteristics of all of the above need to be mathematically described.

To model a violin, you need to take into account the bow against the string, the bow width and material, the bow tension, the string material, the string density, the string tension, the resonance and damping behavior of the strings, the transfer of string vibrations through the bridge (materials, size, and shape of the bridge), and the materials, size, and resonance characteristics of the violin body. Further considerations include the environment that your modeled violin is played in and the playing style—“hammering” or tapping with the bow as opposed to drawing it across the strings.

## Physical Modeling in C++

The `12-physicalmodelling_karplus_strong.cpp` example is based on the Karplus-Strong string synthesis, that loops a short waveform through a filtered delay line to simulate the sound of a hammered or plucked string or some types of percussion.

*Alexander Strong* invented the algorithm, and *Kevin Karplus* did the first analysis of how it worked.

## Physical Modeling with a Aodyo Anyma Phi

[Aodyo Anyma Phi - Physical Modeling Synth - presets and sound tweaking](#)<sup>13</sup>

<sup>13</sup><https://www.youtube.com/watch?v=oe6GMKSNO0k>

# In the Rearview Mirror

As mentioned in the introduction, this is a beginner level book and aimed at that. As with many things in life — we need to learn how to stand before we can learn how to walk — Learn to walk before we can learn how to run and so forth.

So what have we learned? We have learned that the different synthesis methods really are different in how they work and different in how they sound.

It's clear that some synthesis methods are more useful than others. There is no chance in that Subtractive (Virtual Analog or not), FM and wavetable are the most used synthesis methods. And with the performance you'll get with a DSPs at a very low price point today, just makes Physical Modeling a more and more conceivable method to use for music production.

Appendix C is more important than you might think. You should study the terminology to become more proficient with what and how an Audio Engineer work. As an Audio Programmer you'll make tools for an Audio Engineer, and not the other way around. You shouldn't just study what's in this book. If you get a chance to visit a studio and even get a chance to hang out with an Audio Engineer to learn more, than do.

As mentioned in the introduction, you should get your self a modular synthesizer (or at least VCV Rack) to do sound experiments with.

Thank you for your time.

# Appendix A: C++20 Code

00-signalgenerators.cpp - 36691 bytes.

---

```
1 // Signal Generators
2 //
3 // Compile: clang++ -std=c++20 -lpng 00-signalgenerators.cpp -o 00-signalgenerators
4 //
5 // If #include <png++/png.hpp> is missing:
6 // > brew install png++
7 // png++ (header-only) is used to render audio as images to visualize the waveform.
8 // Install with homebrew on macOS or with other package manager.
9 // https://savannah.nongnu.org/projects/pngpp/
10 //
11 // Uses Adam Stark's AudioFile (header-only) to write AIFF-files.
12 // Check out the code from github link below.
13 // https://github.com/adamstark/AudioFile
14 //
15 // Uses p-ranav Indicators
16 // > git clone https://github.com/p-ranav/indicators
17 // > cd indicators
18 // > mkdir build && cd build
19 // > cmake -DINDICATORS_SAMPLES=ON -DINDICATORS_DEMO=ON ..
20 // > make
21 //
22 // python3 utils/amalgamate/amalgamate.py -c single_include.json -s .
23 //
24 // If clang++ is missing:
25 // clang++ from XCode:
26 // > xcode-select --install
27 // > xcode-select -p
28 //
29 // clang++ from homebrew:
30 // > brew install llvm
31 // If clang++ is needed, install it with homebrew for macOS or...
32 // https://clang.llvm.org/get_started.html
33
34 #define _USE_MATH_DEFINES
35 #include <cmath>
36 #include <vector>
```



```
37 #include <random>
38 #include <filesystem>
39 #include "indicators.hpp"
40 #include <png++/png.hpp>
41 #include "AudioFile/AudioFile.h"
42
43 namespace Render
44 {
45     void fillbackground(png::image<png::rgb_pixel>& image);
46     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
47     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
48     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
49 }
50
51 namespace SignalGenerators
52 {
53     void generatesinwaveaiff();
54     void rendersinwavepng();
55     void generatesumoffirsteightpartialsaiff();
56     void rendersumoffirsteightpartialspng();
57     void generatesawtoothwaveaiff();
58     void rendersawtoothwavepng();
59     void generatesquarewaveaiff();
60     void rendersquarewavepng();
61     void generatetrianglewaveaiff();
62     void rendertrianglewavepng();
63     void generatetwentyfivepulsewaveaiff();
64     void rendertwentyfivepulsewavepng();
65     void generateconvexcurveaiff();
66     void renderconvexcurvepng();
67     void generateconcavecurveaiff();
68     void renderconcavecurvepng();
69     void generatefrequencymodulationaiff();
70     void renderfrequencymodulationpng();
71     void generatephasemodulationaiff();
72     void renderphasemodulationpng();
73     void generateamplitudedmodulationaiff();
74     void renderamplitudedmodulationpng();
75     void generateringmodulationaiff();
76     void renderringmodulationpng();
77     void generatenoiseaiff();
78     void rendernoisepng();
79     void generatepulsewaveaiff();
```

```
80     void renderpulsewavepng();
81 }
82
83 int main()
84 {
85     namespace fs = std::filesystem;
86     fs::create_directory("signalgenerators");
87
88     using namespace indicators;
89     // Hide cursor
90     show_console_cursor(false);
91
92     // Setup ProgressBar
93     ProgressBar bar{
94         option::BarWidth{50},
95         option::Start{"["},
96         option::Fill{"█"},
97         option::Lead{"█"},
98         option::Remainder{"-"},
99         option::End{" ]"},
100        option::PostfixText{"Generating: Sine Wave 1/28"},
101        option::ForegroundColor{Color::cyan},
102        option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
103    };
104
105    // Update progress
106    bar.set_progress(13);
107
108    SignalGenerators::generatesinewaveaiff();
109
110    // Update progress
111    bar.set_progress(19);
112    bar.set_option(option::PostfixText{"Rendering: Sine Wave 2/28"});
113
114    SignalGenerators::rendersinewavepng();
115
116    // Update progress
117    bar.set_progress(22);
118    bar.set_option(option::PostfixText{"Generating: Sum of first eight partials 3/28"});
119
120    SignalGenerators::generatesumoffirsteightpartialsaiff();
121
122    // Update progress
```

```
123     bar.set_progress(25);
124     bar.set_option(option::PostfixText{"Rendering: Sum of first eight partials 4/28"});
125
126     SignalGenerators::rendersumoffirsteightpartialspng();
127
128     // Update progress
129     bar.set_progress(28);
130     bar.set_option(option::PostfixText{"Generating: Saw Tooth Wave 5/28"});
131
132     SignalGenerators::generatesawtoothwaveaiff();
133
134     // Update progress
135     bar.set_progress(31);
136     bar.set_option(option::PostfixText{"Rendering: Saw Tooth Wave 6/28"});
137
138     SignalGenerators::rendersawtoothwavepng();
139
140     // Update progress
141     bar.set_progress(34);
142     bar.set_option(option::PostfixText{"Generating: Square Wave 7/28"});
143
144     SignalGenerators::generatesquarewaveaiff();
145
146     // Update progress
147     bar.set_progress(37);
148     bar.set_option(option::PostfixText{"Rendering: Square Wave 8/28"});
149
150     SignalGenerators::rendersquarewavepng();
151
152     // Update progress
153     bar.set_progress(40);
154     bar.set_option(option::PostfixText{"Generating: Triangle Wave 9/28"});
155
156     SignalGenerators::generatetrianglewaveaiff();
157
158     // Update progress
159     bar.set_progress(43);
160     bar.set_option(option::PostfixText{"Rendering: Triangle Wave 10/28"});
161
162     SignalGenerators::rendertrianglewavepng();
163
164     // Update progress
165     bar.set_progress(46);
```

```
166     bar.set_option(option::PostfixText{"Generating: 25% Pulse Wave 11/28"});
167
168     SignalGenerators::generatetwentyfivepulsewaveaiff();
169
170     // Update progress
171     bar.set_progress(49);
172     bar.set_option(option::PostfixText{"Rendering: 25% Pulse Wave 12/28"});
173
174     SignalGenerators::rendertwentyfivepulsewavepng();
175
176     // Update progress
177     bar.set_progress(52);
178     bar.set_option(option::PostfixText{"Generating: Convex Curve 13/28"});
179
180     SignalGenerators::generateconvexcurveaiff();
181
182     // Update progress
183     bar.set_progress(55);
184     bar.set_option(option::PostfixText{"Rendering: Convex Curve 14/28"});
185
186     SignalGenerators::renderconvexcurvepng();
187
188     // Update progress
189     bar.set_progress(58);
190     bar.set_option(option::PostfixText{"Generating: Concave Curve 15/28"});
191
192     SignalGenerators::generateconcavecurveaiff();
193
194     // Update progress
195     bar.set_progress(61);
196     bar.set_option(option::PostfixText{"Rendering: Concave Curve 16/28"});
197
198     SignalGenerators::renderconcavecurvepng();
199
200     // Update progress
201     bar.set_progress(64);
202     bar.set_option(option::PostfixText{"Generating: Frequency Modulation 17/28"});
203
204
205     SignalGenerators::generatefrequencymodulationaiff();
206
207     // Update progress
208     bar.set_progress(67);
```

```
209     bar.set_option(option::PostfixText{"Rendering: Frequency Modulation 18/28"});
210
211     SignalGenerators::renderfrequencymodulationpng();
212
213     // Update progress
214     bar.set_progress(70);
215     bar.set_option(option::PostfixText{"Generating: Phase Modulation 19/28"});
216
217     SignalGenerators::generatephasemodulationaiiff();
218
219     // Update progress
220     bar.set_progress(73);
221     bar.set_option(option::PostfixText{"Rendering: Phase Modulation 20/28"});
222
223     SignalGenerators::renderphasemodulationpng();
224
225     // Update progress
226     bar.set_progress(76);
227     bar.set_option(option::PostfixText{"Generating: Amplitude Modulation 21/28"});
228
229     SignalGenerators::generateamplitudedmodulationaiiff();
230
231     // Update progress
232     bar.set_progress(79);
233     bar.set_option(option::PostfixText{"Rendering: Amplitude Modulation 22/28"});
234
235     SignalGenerators::renderamplitudedmodulationpng();
236
237     // Update progress
238     bar.set_progress(82);
239     bar.set_option(option::PostfixText{"Generating: Ring Modulation 23/28"});
240
241     SignalGenerators::generateringmodulationaiiff();
242
243     // Update progress
244     bar.set_progress(85);
245     bar.set_option(option::PostfixText{"Rendering: Ring Modulation 24/28"});
246
247     SignalGenerators::renderringmodulationpng();
248
249     // Update progress
250     bar.set_progress(88);
251     bar.set_option(option::PostfixText{"Generating: White Noise 25/28"});
```

```
252
253     SignalGenerators::generatenoiseaiff();
254
255     // Update progress
256     bar.set_progress(91);
257     bar.set_option(option::PostfixText{"Rendering: White Noise 26/28"});
258
259     SignalGenerators::rendernoisepng();
260
261     // Update progress
262     bar.set_progress(94);
263     bar.set_option(option::PostfixText{"Generating: Pulse Wave 27/28"});
264
265     SignalGenerators::generatepulsewaveaiff();
266
267     // Update progress
268     bar.set_progress(97);
269     bar.set_option(option::PostfixText{"Rendering: Pulse Wave 28/28"});
270
271     SignalGenerators::renderpulsewavepng();
272
273     // Update progress
274     bar.set_progress(100);
275     bar.set_option(option::PostfixText{"Done 28/28"});
276
277     // Show cursor
278     show_console_cursor(true);
279
280     return 0;
281 }
282
283 namespace SignalGenerators
284 {
285     void generatesinewaveaiff()
286     {
287         const std::string sineName="signalgenerators/01-sine-wave.aiff";
288         const double sampleRate=44100.0;
289         const double frequencyInHz=440.0;
290
291         // Setup the audio file
292         AudioFile<float> a;
293         a.setNumChannels(1);
294         a.setBitDepth(24);
```

```

295         a.setNumSamplesPerChannel(44100);
296
297         // Genrate Sine wave
298         for (int i=0;i<a.getNumSamplesPerChannel();++i)
299         {
300             for (int channel=0;channel<a.getNumChannels();++channel)
301             {
302                 a.samples[channel][i]=sin((static_cast<double> (i) / sampleRate) *
303 * 2.0 * M_PI);
304             }
305         }
306
307         a.save(sineName,AudioFileFormat::Aiff);
308     }
309
310     void rendersinewavepng()
311     {
312         const std::string sineName="signalgenerators/02-sine-wave.png";
313         const double sampleRate=44100.0;
314         const double frequencyInHz=440.0;
315         std::vector<uint32_t> signalY;
316         signalY.resize(44100);
317
318         for (int i=0;i<sampleRate;++i)
319         {
320             double value=sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2
321 _PI);
322             if (value >= 0.0) {
323                 signalY[i]=300-(300*value);
324             } else if (value < 0.0)
325             {
326                 signalY[i]=300+(300*fabs(value));
327             }
328         }
329
330         png::image<png::rgb_pixel> image(44100,600);
331         Render::fillbackground(image);
332         Render::drawwave(image,signalY);
333         image.write(sineName);
334     }
335
336     void generatesumoffirsteightpartialsaiff()
337     {

```

```
338         const std::string sineName="signalgenerators/03-sine-wave-first-eight-partials.aif\
339 f";
340         const double sampleRate=44100.0;
341         const double frequencyInHz=440.0;
342
343         // Setup the audio file
344         AudioFile<float> a;
345         a.setNumChannels(1);
346         a.setBitDepth(24);
347         a.setNumSamplesPerChannel(44100);
348
349         double s1=0.0,s2=0.0,s3=0.0,s4=0.0,s5=0.0,s6=0.0,s7=0.0,s8=0.0,ss=0.0;
350
351         // Genrate Sine wave
352         for (int i=0;i<a.getNumSamplesPerChannel();++i)
353         {
354             for (int channel=0;channel<a.getNumChannels();++channel)
355             {
356                 s1=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*1) *
357                 s2=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*2) *
358                 s3=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*3) *
359                 s4=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*4) *
360                 s5=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*5) *
361                 s6=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*6) *
362                 s7=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*7) *
363                 s8=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*8) *
364                 ss=(s1+s2+s3+s4+s5+s6+s7+s8)/8;
365                 a.samples[channel][i]=ss;
366             }
367         }
368
369         a.save(sineName,AudioFileFormat::Aiff);
370     }
371
372     void rendersumoffirsteightpartialspng()
373     {
374         const std::string sineName="signalgenerators/04-sine-wave-first-eight-partials.png\
375 ";
376         const double sampleRate=44100.0;
377         const double frequencyInHz=440.0;
378         double s1=0.0,s2=0.0,s3=0.0,s4=0.0,s5=0.0,s6=0.0,s7=0.0,s8=0.0,ss=0.0;
379         std::vector<uint32_t> signalY;
380         signalY.resize(44100);
```



```

381
382     // Genrate Sine waves
383     for (int i=0;i<sampleRate;++i)
384     {
385         s1=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*1) * 2.0 * M_
386             s2=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*2) *
387             s3=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*3) *
388             s4=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*4) *
389             s5=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*5) *
390             s6=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*6) *
391             s7=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*7) *
392             s8=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*8) *
393             ss=(s1+s2+s3+s4+s5+s6+s7+s8)/8;
394             if (ss >= 0.0) {
395                 signalY[i]=300-(300*ss);
396             } else if (ss < 0.0)
397             {
398                 signalY[i]=300+(300*fabs(ss));
399             }
400     }
401
402     png::image<png::rgb_pixel> image(44100,600);
403     Render::fillbackground(image);
404     Render::drawwave(image,signalY);
405     image.write(sineName);
406 }
407
408 void generatesawtoothwaveaiff()
409 {
410     const std::string sineName="signalgenerators/05-saw-tooth-wave.aiff";
411     const double sampleRate=44100.0;
412     const double frequencyInHz=440.0;
413     double ss;
414
415     // Setup the audio file
416     AudioFile<float> a;
417     a.setNumChannels(1);
418     a.setBitDepth(24);
419     a.setNumSamplesPerChannel(44100);
420
421     for (int i=0;i<a.getNumSamplesPerChannel();++i)
422     {
423         for (int channel=0;channel<a.getNumChannels();++channel)

```

```
424         {
425             ss=-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate));
426             a.samples[channel][i]=ss;
427         }
428     }
429
430     a.save(sineName,AudioFileFormat::Aiff);
431 }
432
433 void rendersawtoothwavepng()
434 {
435     const std::string sineName="signalgenerators/06-saw-tooth-wave.png";
436     const double sampleRate=44100.0;
437     const double frequencyInHz=440.0;
438     double ss;
439     std::vector<uint32_t> signalY;
440     signalY.resize(44100);
441
442     for (int i=0;i<44100;++i)
443     {
444         ss=-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate));
445         if (ss >= 0.0) {
446             signalY[i]=300-(300*ss);
447         } else if (ss < 0.0)
448         {
449             signalY[i]=300+(300*fabs(ss));
450         }
451     }
452
453     png::image<png::rgb_pixel> image(44100,600);
454     Render::fillbackground(image);
455     Render::drawwave(image,signalY);
456     image.write(sineName);
457 }
458
459 void generatesquarewaveaiff()
460 {
461     const std::string sineName="signalgenerators/07-square-wave.aiff";
462     const double sampleRate=44100.0;
463     const double frequencyInHz=440.0;
464     double period=sampleRate/frequencyInHz;
465     double dutyCycle=period*0.5;
466     double ss;
```

```

467
468     // Setup the audio file
469     AudioFile<float> a;
470     a.setNumChannels(1);
471     a.setBitDepth(24);
472     a.setNumSamplesPerChannel(44100);
473
474     for (int i=0; i<a.getNumSamplesPerChannel();++i)
475     {
476         for (int channel=0; channel<a.getNumChannels();++channel)
477         {
478             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
479             {
480                 ss=0.7;
481             } else {
482                 ss=-0.7;
483             }
484             a.samples[channel][i]=ss;
485         }
486     }
487     a.samples[0][0]=0.0;
488     a.samples[0][44099]=0.0;
489     a.save(sineName,AudioFileFormat::Aiff);
490 }
491
492 void rendersquarewavepng()
493 {
494     const std::string sineName="signalgenerators/08-square-wave.png";
495     const double sampleRate=44100.0;
496     const double frequencyInHz=440.0;
497     double period=sampleRate/frequencyInHz;
498     double dutyCycle=period*0.5;
499     std::vector<uint32_t> signalY;
500     signalY.resize(44100);
501
502     for (int i=0; i<44100;++i)
503     {
504         if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
505         {
506             signalY[i]=0;
507         } else {
508             signalY[i]=600;
509         }

```

```

510         }
511
512         signalY[0]=300;
513         signalY[44099]=300;
514
515         png::image<png::rgb_pixel> image(44100,600);
516         Render::fillbackground(image);
517         Render::drawwave(image,signalY);
518         image.write(sineName);
519     }
520
521     void generatetrianglewaveaiff()
522     {
523         const std::string sineName="signalgenerators/09-trangle-wave.aiff";
524         const double sampleRate=44100.0;
525         const double frequencyInHz=440.0;
526         double ss;
527
528         // Setup the audio file
529         AudioFile<float> a;
530         a.setNumChannels(1);
531         a.setBitDepth(24);
532         a.setNumSamplesPerChannel(44100);
533
534         for (int i=0;i<a.getNumSamplesPerChannel();++i)
535         {
536             for (int channel=0;channel<a.getNumChannels();++channel)
537             {
538                 ss=M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate));
539                 a.samples[channel][i]=ss;
540             }
541         }
542         a.save(sineName,AudioFileFormat::Aiff);
543     }
544
545     void rendertrianglewavepng()
546     {
547         const std::string sineName="signalgenerators/10-trangle-wave.png";
548         const double sampleRate=44100.0;
549         const double frequencyInHz=440.0;
550         double ss;
551         std::vector<uint32_t> signalY;
552         signalY.resize(44100);

```

```

553
554     for (int i=0;i<44100;++i)
555     {
556         ss=M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate));
557         if (ss >= 0.0) {
558             signalY[i]=300-(300*ss);
559         } else if (ss < 0.0)
560         {
561             signalY[i]=300+(300*fabs(ss));
562         }
563     }
564
565     png::image<png::rgb_pixel> image(44100,600);
566     Render::fillbackground(image);
567     Render::drawwave(image,signalY);
568     image.write(sineName);
569 }
570
571 void generatetwentyfivepulsewaveaiff()
572 {
573     const std::string sineName="signalgenerators/11-twentyfive-pulse-wave.aiff";
574     const double sampleRate=44100.0;
575     const double frequencyInHz=440.0;
576     double period=sampleRate/frequencyInHz;
577     double dutyCycle=period*0.25;
578     double ss;
579
580     // Setup the audio file
581     AudioFile<float> a;
582     a.setNumChannels(1);
583     a.setBitDepth(24);
584     a.setNumSamplesPerChannel(44100);
585
586     for (int i=0;i<a.getNumSamplesPerChannel();++i)
587     {
588         for (int channel=0;channel<a.getNumChannels();++channel)
589         {
590             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
591             {
592                 ss=0.7;
593             } else {
594                 ss=-0.7;
595             }

```

```

596             a.samples[channel][i]=ss;
597         }
598     }
599     a.save(sineName,AudioFileFormat::Aiff);
600 }
601
602 void rendertwentyfivepulsewavepng()
603 {
604     const std::string sineName="signalgenerators/12-twentyfive-pulse-wave.png";
605     const double sampleRate=44100.0;
606     const double frequencyInHz=440.0;
607     double period=sampleRate/frequencyInHz;
608     double dutyCycle=period*0.25;
609     std::vector<uint32_t> signalY;
610     signalY.resize(44100);
611
612     for (int i=0;i<44100;++i)
613     {
614         if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
615         {
616             signalY[i]=0;
617         } else {
618             signalY[i]=600;
619         }
620     }
621
622     signalY[0]=300;
623     signalY[44099]=300;
624
625     png::image<png::rgb_pixel> image(44100,600);
626     Render::fillbackground(image);
627     Render::drawwave(image,signalY);
628     image.write(sineName);
629 }
630
631 void generateconvexcurveaiff()
632 {
633     const std::string sineName="signalgenerators/13-convex-curve.aiff";
634     const double sampleRate=44100.0;
635     const double frequencyInHz=440.0;
636     double period=sampleRate/frequencyInHz;
637     uint32_t maxValue=0,temp=0;
638     double ss,sss;

```

```

639
640     // Setup the audio file
641     AudioFile<float> a;
642     a.setNumChannels(1);
643     a.setBitDepth(24);
644     a.setNumSamplesPerChannel(44100);
645
646     for (int i=0;i<44100;++i)
647     {
648         temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
649         if (temp > maxValue) {maxValue=temp;}
650     }
651
652     for (int i=0;i<a.getNumSamplesPerChannel();++i)
653     {
654         for (int channel=0;channel<a.getNumChannels();++channel)
655         {
656             temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
657             ss=double(temp)/double(maxValue);
658             sss=(ss-0.5)*1.5;
659             a.samples[channel][i]=sss;
660         }
661     }
662     a.save(sineName,AudioFileFormat::Aiff);
663 }
664
665 void renderconvexcurvepng()
666 {
667     const std::string sineName="signalgenerators/14-convex-curve.png";
668     const double sampleRate=44100.0;
669     const double frequencyInHz=440.0;
670     double period=sampleRate/frequencyInHz;
671     double ss;
672     uint32_t maxValue=0,temp=0;
673     std::vector<uint32_t> signalY;
674     signalY.resize(44100);
675
676     for (int i=0;i<44100;++i)
677     {
678         temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
679         if (temp > maxValue) {maxValue=temp;}
680     }
681

```

```

682         for (int i=0;i<44100;++i)
683         {
684             temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
685             ss=double(temp)/double(maxValue);
686             signalY[i]=600-(600*ss);
687         }
688
689         png::image<png::rgb_pixel> image(44100,600);
690         Render::fillbackground(image);
691         Render::drawwave(image,signalY);
692         image.write(sineName);
693     }
694
695     void generateconcavecurveaiff()
696     {
697         const std::string sineName="signalgenerators/15-concave-curve.aiff";
698         const double sampleRate=44100.0;
699         const double frequencyInHz=440.0;
700         double period=sampleRate/frequencyInHz;
701         uint32_t maxValue=0;
702         double ss,sss,temp;
703
704         // Setup the audio file
705         AudioFile<float> a;
706         a.setNumChannels(1);
707         a.setBitDepth(24);
708         a.setNumSamplesPerChannel(44100);
709
710         for (int i=0;i<44100;++i)
711         {
712             temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
713             if (int(temp) > maxValue) {maxValue=int(temp);}
714         }
715
716         for (int i=0;i<a.getNumSamplesPerChannel();++i)
717         {
718             for (int channel=0;channel<a.getNumChannels();++channel)
719             {
720                 temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
721                 ss=temp/double(maxValue);
722                 sss=(ss-0.5)*1.5;
723                 a.samples[channel][i]=sss;
724             }

```



```
725         }
726         a.save(sineName,AudioFileFormat::Aiff);
727     }
728
729     void renderconcavecurvepng()
730     {
731         const std::string sineName="signalgenerators/16-concave-curve.png";
732         const double sampleRate=44100.0;
733         const double frequencyInHz=440.0;
734         double period=sampleRate/frequencyInHz;
735         double ss,temp;
736         uint32_t maxValue=0;
737         std::vector<uint32_t> signalY;
738         signalY.resize(44100);
739
740         for (int i=0;i<44100;++i)
741         {
742             temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
743             if (int(temp) > maxValue) {maxValue=int(temp);}
744         }
745
746         for (int i=0;i<44100;++i)
747         {
748             temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
749             ss=temp/double(maxValue);
750             signalY[i]=600-(600*ss);
751         }
752
753         png::image<png::rgb_pixel> image(44100,600);
754         Render::fillbackground(image);
755         Render::drawwave(image,signalY);
756         image.write(sineName);
757     }
758
759     void generatefrequencymodulationaiff()
760     {
761         const std::string sineName="signalgenerators/17-frequency-modulation.aiff";
762         const double twoPI=2*M_PI;
763         const double sampleRate=44100.0;
764         const double frequencyInHz=440.0;
765         const double frequencyRadian = twoPI / sampleRate;
766         double modifierFrequency = frequencyInHz * 3;
767         double modifierIncrement = frequencyRadian * modifierFrequency;
```

```

768         double modifierPhase = 0;
769         double carrierIncrement = 0;
770         double carrierPhase = 0;
771         double modifieramplitude = 2 * modifierFrequency;
772         double modifierValue = 0;
773
774         // Setup the audio file
775         AudioFile<float> a;
776         a.setNumChannels(1);
777         a.setBitDepth(24);
778         a.setNumSamplesPerChannel(44100);
779
780         for (int i=0;i<a.getNumSamplesPerChannel();++i)
781         {
782             for (int channel=0;channel<a.getNumChannels();++channel)
783             {
784                 a.samples[channel][i]=sinf(carrierPhase);
785                 modifierValue = modifieramplitude * sinf(modifierPhase);
786                 carrierIncrement = frequencyRadian * (frequencyInHz + modifierValue);
787                 carrierPhase = carrierPhase + carrierIncrement;
788                 modifierPhase = modifierPhase + modifierIncrement;
789                 if (carrierPhase >= twoPI) {
790                     carrierPhase -= twoPI;
791                 }
792                 else if (carrierPhase < 0) {
793                     carrierPhase += twoPI;
794                 }
795                 if (modifierPhase >= twoPI) {
796                     modifierPhase -= twoPI;
797                 }
798             }
799         }
800         a.save(sineName,AudioFileFormat::Aiff);
801     }
802
803     void renderfrequencymodulationpng()
804     {
805         const std::string sineName="signalgenerators/18-frequency-modulation.png";
806         const double twoPI=2*M_PI;
807         const double sampleRate=44100.0;
808         const double frequencyInHz=440.0;
809         const double frequencyRadian = twoPI / sampleRate;
810         double modifierFrequency = frequencyInHz * 3;

```

```

811         double modifierIncrement = frequencyRadian * modifierFrequency;
812         double modifierPhase = 0;
813         double carrierIncrement = 0;
814         double carrierPhase = 0;
815         double modifieramplitude = 2 * modifierFrequency;
816         double modifierValue = 0;
817         double value=0;
818         std::vector<uint32_t> signalY;
819         signalY.resize(44100);
820
821         for (uint32_t i=0;i<sampleRate;++i)
822         {
823             value=sinf(carrierPhase);
824             if (value >= 0.0) {
825                 signalY[i]=300-(300*value);
826             } else if (value < 0.0)
827             {
828                 signalY[i]=300+(300*fabs(value));
829             }
830             modifierValue = modifieramplitude * sinf(modifierPhase);
831             carrierIncrement = frequencyRadian * (frequencyInHz + modifierValue);
832             carrierPhase = carrierPhase + carrierIncrement;
833             modifierPhase = modifierPhase + modifierIncrement;
834             if (carrierPhase >= twoPI) {
835                 carrierPhase -= twoPI;
836             }
837             else if (carrierPhase < 0) {
838                 carrierPhase += twoPI;
839             }
840             if (modifierPhase >= twoPI) {
841                 modifierPhase -= twoPI;
842             }
843         }
844         png::image<png::rgb_pixel> image(44100,600);
845         Render::fillbackground(image);
846         Render::drawwave(image,signalY);
847         image.write(sineName);
848     }
849
850     void generatephasemodulationaiff()
851     {
852         const std::string sineName="signalgenerators/19-phase-modulation.aiff";
853         const double twoPI=2*M_PI;

```

```

854     const double sampleRate=44100.0;
855     const double frequencyInHz=440.0;
856     const double frequencyRadian = twoPI / sampleRate;
857     double modifierFrequency = frequencyInHz * 3;
858     double carrierIncrement = frequencyRadian * frequencyInHz;
859     double modifieramplitude = frequencyRadian * (2 * modifierFrequency);
860     double modifierValue = 0;
861     double carrierPhase = 0;
862     double modifierPhase = 0;
863     double modifierIncrement = frequencyRadian * modifierFrequency;
864
865     // Setup the audio file
866     AudioFile<float> a;
867     a.setNumChannels(1);
868     a.setBitDepth(24);
869     a.setNumSamplesPerChannel(44100);
870
871     for (int i=0;i<a.getNumSamplesPerChannel();++i)
872     {
873         for (int channel=0;channel<a.getNumChannels();++channel)
874         {
875             a.samples[channel][i]=sinf(carrierPhase);
876             modifierValue = modifieramplitude * sinf(modifierPhase);
877             carrierPhase = carrierPhase + carrierIncrement + modifierValue;
878             modifierPhase = modifierPhase + modifierIncrement;
879             if (carrierPhase >= twoPI) {
880                 carrierPhase -= twoPI;
881             }
882             else if (carrierPhase < 0) {
883                 carrierPhase += twoPI;
884             }
885             if (modifierPhase >= twoPI) {
886                 modifierPhase -= twoPI;
887             }
888         }
889     }
890     a.save(sineName,AudioFileFormat::Aiff);
891 }
892
893 void renderphasemodulationpng()
894 {
895     const std::string sineName="signalgenerators/20-phase-modulation.png";
896     const double twoPI=2*M_PI;

```

```

897     const double sampleRate=44100.0;
898     const double frequencyInHz=440.0;
899     const double frequencyRadian = twoPI / sampleRate;
900     double modifierFrequency = frequencyInHz * 3;
901     double carrierIncrement = frequencyRadian * frequencyInHz;
902     double modifieramplitude = frequencyRadian * (2 * modifierFrequency);
903     double modifierValue = 0;
904     double carrierPhase = 0;
905     double modifierPhase = 0;
906     double modifierIncrement = frequencyRadian * modifierFrequency;
907     double value=0;
908     std::vector<uint32_t> signalY;
909     signalY.resize(44100);
910
911     for (uint32_t i=0;i<sampleRate;++i)
912     {
913         value=sinf(carrierPhase);
914         if (value >= 0.0) {
915             signalY[i]=300-(300*value);
916         } else if (value < 0.0)
917         {
918             signalY[i]=300+(300*fabs(value));
919         }
920         modifierValue = modifieramplitude * sinf(modifierPhase);
921         carrierPhase = carrierPhase + carrierIncrement + modifierValue;
922         modifierPhase = modifierPhase + modifierIncrement;
923         if (carrierPhase >= twoPI) {
924             carrierPhase -= twoPI;
925         }
926         else if (carrierPhase < 0) {
927             carrierPhase += twoPI;
928         }
929         if (modifierPhase >= twoPI){
930             modifierPhase -= twoPI;
931         }
932     }
933     png::image<png::rgb_pixel> image(44100,600);
934     Render::fillbackground(image);
935     Render::drawwave(image,signalY);
936     image.write(sineName);
937 }
938
939 void generateamplitudedmodulationaiff()

```

```

940     {
941         const std::string sineName="signalgenerators/21-amplituded-modulation.aiff";
942         const double twoPI=2*M_PI;
943         const double sampleRate=44100.0;
944         const double frequencyInHz=440.0;
945         const double frequencyRadian = twoPI / sampleRate;
946         double modifierFrequency = frequencyInHz * 2.5;
947         double modifierIncrement = frequencyRadian * modifierFrequency;
948         double modifieramplitude = 1.0;
949         double carrierPhase = 0;
950         double modifierPhase = 0;
951         double modifierScale = 1 / (1 + modifieramplitude);
952         double modifierValue = 0;
953         double carrierIncrement = frequencyRadian * frequencyInHz;
954
955         // Setup the audio file
956         AudioFile<float> a;
957         a.setNumChannels(1);
958         a.setBitDepth(24);
959         a.setNumSamplesPerChannel(44100);
960
961         for (int i=0;i<a.getNumSamplesPerChannel();++i)
962         {
963             for (int channel=0;channel<a.getNumChannels();++channel)
964             {
965                 modifierValue = 1.0 + (modifieramplitude * sinf(modifierPhase));
966                 a.samples[channel][i]=((sinf(carrierPhase) * modifierValue) * modifierScale);
967                 carrierPhase += carrierIncrement;
968                 if (carrierPhase >= twoPI) {
969                     carrierPhase -= twoPI;
970                 }
971                 modifierPhase += modifierIncrement;
972                 if (modifierPhase >= twoPI) {
973                     modifierPhase -= twoPI;
974                 }
975             }
976         }
977         a.save(sineName,AudioFileFormat::Aiff);
978     }
979
980 void renderamplitudedmodulationpng()
981 {
982     const std::string sineName="signalgenerators/22-amplituded-modulation.png";

```

```

983     const double twoPI=2*M_PI;
984     const double sampleRate=44100.0;
985     const double frequencyInHz=440.0;
986     const double frequencyRadian = twoPI / sampleRate;
987     double modifierFrequency = frequencyInHz * 2.5;
988     double modifierIncrement = frequencyRadian * modifierFrequency;
989     double modifieramplitude = 1.0;
990     double carrierPhase = 0;
991     double modifierPhase = 0;
992     double modifierScale = 1 / (1 + modifieramplitude);
993     double modifierValue = 0;
994     double carrierIncrement = frequencyRadian * frequencyInHz;
995     double value=0;
996     std::vector<uint32_t> signalY;
997     signalY.resize(44100);
998
999     for (uint32_t i=0;i<sampleRate;++i)
1000    {
1001        modifierValue = 1.0 + (modifieramplitude * sinf(modifierPhase));
1002        value=((sinf(carrierPhase) * modifierValue) * modifierScale);
1003        if (value >= 0.0) {
1004            signalY[i]=300-(300*value);
1005        } else if (value < 0.0)
1006        {
1007            signalY[i]=300+(300*fabs(value));
1008        }
1009        carrierPhase += carrierIncrement;
1010        if (carrierPhase >= twoPI) {
1011            carrierPhase -= twoPI;
1012        }
1013        modifierPhase += modifierIncrement;
1014        if (modifierPhase >= twoPI) {
1015            modifierPhase -= twoPI;
1016        }
1017    }
1018    png::image<png::rgb_pixel> image(44100,600);
1019    Render::fillbackground(image);
1020    Render::drawwave(image,signalY);
1021    image.write(sineName);
1022 }
1023
1024 void generateringmodulationaiiff()
1025 {

```

```

1026         const std::string sineName="signalgenerators/23-ring-modulation.aiff";
1027         const double twoPI=2*M_PI;
1028         const double sampleRate=44100.0;
1029         const double frequencyInHz=440.0;
1030         const double frequencyRadian = twoPI / sampleRate;
1031         double modifieramplitude = 1.0;
1032         double modifierFrequency = frequencyInHz * 2.5;
1033         double modifierPhase = 0;
1034         double modifierIncrement = frequencyRadian * modifierFrequency;
1035         double carrierPhase = 0;
1036         double carrierIncrement = frequencyRadian * frequencyInHz;
1037
1038         // Setup the audio file
1039         AudioFile<float> a;
1040         a.setNumChannels(1);
1041         a.setBitDepth(24);
1042         a.setNumSamplesPerChannel(44100);
1043
1044         for (int i=0;i<a.getNumSamplesPerChannel();++i)
1045         {
1046             for (int channel=0;channel<a.getNumChannels();++channel)
1047             {
1048                 a.samples[channel][i]=(sinf(carrierPhase) * modifieramplitude * sin(
1049 se));
1050                 carrierPhase += carrierIncrement;
1051                 if (carrierPhase >= twoPI) {
1052                     carrierPhase -= twoPI;
1053                 }
1054                 modifierPhase += modifierIncrement;
1055                 if (modifierPhase >= twoPI) {
1056                     modifierPhase -= twoPI;
1057                 }
1058             }
1059         }
1060         a.save(sineName,AudioFileFormat::Aiff);
1061     }
1062
1063     void renderringmodulationpng()
1064     {
1065         const std::string sineName="signalgenerators/24-ring-modulation.png";
1066         const double twoPI=2*M_PI;
1067         const double sampleRate=44100.0;
1068         const double frequencyInHz=440.0;

```



```

1069     const double frequencyRadian = twoPI / sampleRate;
1070     double modifieramplitude = 1.0;
1071     double modifierFrequency = frequencyInHz * 2.5;
1072     double modifierPhase = 0;
1073     double modifierIncrement = frequencyRadian * modifierFrequency;
1074     double carrierPhase = 0;
1075     double carrierIncrement = frequencyRadian * frequencyInHz;
1076     double value=0;
1077     std::vector<uint32_t> signalY;
1078     signalY.resize(44100);
1079
1080     for (uint32_t i=0;i<sampleRate;++i)
1081     {
1082         value=(sinf(carrierPhase) * modifieramplitude * sinf(modifierPhase));
1083         if (value >= 0.0) {
1084             signalY[i]=300-(300*value);
1085         } else if (value < 0.0)
1086         {
1087             signalY[i]=300+(300*fabs(value));
1088         }
1089         carrierPhase += carrierIncrement;
1090         if (carrierPhase >= twoPI) {
1091             carrierPhase -= twoPI;
1092         }
1093         modifierPhase += modifierIncrement;
1094         if (modifierPhase >= twoPI) {
1095             modifierPhase -= twoPI;
1096         }
1097     }
1098     png::image<png::rgb_pixel> image(44100,600);
1099     Render::fillbackground(image);
1100     Render::drawwave(image,signalY);
1101     image.write(sineName);
1102 }
1103
1104 void generatenoiseaiff()
1105 {
1106     const std::string sineName="signalgenerators/25-white-noise.aiff";
1107     std::random_device rd;
1108     std::mt19937 gen(rd());
1109     std::uniform_real_distribution<> dis(-1.0, 1.0);
1110
1111     // Setup the audio file

```

```

1112     AudioFile<float> a;
1113     a.setNumChannels(1);
1114     a.setBitDepth(24);
1115     a.setNumSamplesPerChannel(44100);
1116
1117     for (int i=0;i<a.getNumSamplesPerChannel();++i)
1118     {
1119         for (int channel=0;channel<a.getNumChannels();++channel)
1120         {
1121             a.samples[channel][i]=dis(gen);
1122         }
1123     }
1124     a.save(sineName,AudioFileFormat::Aiff);
1125 }
1126
1127 void rendernoisepng()
1128 {
1129     const std::string sineName="signalgenerators/26-white-noise.png";
1130     const double sampleRate=44100.0;
1131     std::random_device rd;
1132     std::mt19937 gen(rd());
1133     std::uniform_real_distribution<> dis(-1.0, 1.0);
1134     double value=0;
1135     std::vector<uint32_t> signalY;
1136     signalY.resize(44100);
1137
1138     for (uint32_t i=0;i<sampleRate;++i)
1139     {
1140         value=dis(gen);
1141         if (value >= 0.0) {
1142             signalY[i]=300-(300*value);
1143         } else if (value < 0.0)
1144         {
1145             signalY[i]=300+(300*fabs(value));
1146         }
1147     }
1148     png::image<png::rgb_pixel> image(44100,600);
1149     Render::fillbackground(image);
1150     Render::drawwave(image,signalY);
1151     image.write(sineName);
1152 }
1153
1154 void generatepulsewaveaiff()

```

```

1155     {
1156         const std::string sineName="signalgenerators/27-pulse-wave.aiff";
1157         const double twoPI=2*M_PI;
1158         const double sampleRate=44100.0;
1159         const double frequencyInHz=440.0;
1160         const double frequencyRadian = twoPI / sampleRate;
1161         double maxNumber = floor(sampleRate / (2.0 * frequencyInHz) - 0.5) - 1;
1162         double phaseIncrementDenominator = (M_PI / sampleRate) * frequencyInHz;
1163         double phaseIncrementNumerator = phaseIncrementDenominator * ((2.0 * maxNumber) + \
1164 1);
1165         double phaseDenominator = 0.0;
1166         double phaseNumerator = 0.0;
1167         double amplitudeScale = 1.0 / (2.0 * maxNumber);
1168         double value = 0;
1169
1170         // Setup the audio file
1171         AudioFile<float> a;
1172         a.setNumChannels(1);
1173         a.setBitDepth(24);
1174         a.setNumSamplesPerChannel(44100);
1175
1176         for (int i=0;i<a.getNumSamplesPerChannel();++i)
1177         {
1178             for (int channel=0;channel<a.getNumChannels();++channel)
1179             {
1180                 double Denominator = sin(phaseDenominator);
1181                 if (Denominator == 0.0) {
1182                     value = 1.0;
1183                 } else {
1184                     value = amplitudeScale * ((sin(phaseNumerator) / Denominator)
1185                 }
1186                 a.samples[channel][i]=value;
1187                 phaseDenominator += phaseIncrementDenominator;
1188                 if (phaseDenominator >= twoPI) {
1189                     phaseDenominator -= twoPI;
1190                 }
1191                 phaseNumerator += phaseIncrementNumerator;
1192                 if (phaseNumerator >= twoPI) {
1193                     phaseNumerator -= twoPI;
1194                 }
1195             }
1196         }
1197         a.save(sineName,AudioFileFormat::Aiff);

```

```

1198     }
1199
1200     void renderpulsewavepng()
1201     {
1202         const std::string sineName="signalgenerators/28-pulse-wave.png";
1203         const double twoPI=2*M_PI;
1204         const double sampleRate=44100.0;
1205         const double frequencyInHz=440.0;
1206         const double frequencyRadian = twoPI / sampleRate;
1207         double maxNumber = floor(sampleRate / (2.0 * frequencyInHz) - 0.5) - 1;
1208         double phaseIncrementDenominator = (M_PI / sampleRate) * frequencyInHz;
1209         double phaseIncrementNumerator = phaseIncrementDenominator * ((2.0 * maxNumber) + \
1210 1);
1211         double phaseDenominator = 0.0;
1212         double phaseNumerator = 0.0;
1213         double amplitudeScale = 1.0 / (2.0 * maxNumber);
1214         double value = 0;
1215         std::vector<uint32_t> signalY;
1216         signalY.resize(44100);
1217
1218         for (uint32_t i=0;i<sampleRate;++i)
1219         {
1220             double Denominator = sin(phaseDenominator);
1221             if (Denominator == 0.0) {
1222                 value = 1.0;
1223             } else {
1224                 value = amplitudeScale * ((sin(phaseNumerator) / Denominator) - 1);
1225             }
1226
1227             if (value >= 0.0) {
1228                 signalY[i]=300-(300*value);
1229             } else if (value < 0.0)
1230             {
1231                 signalY[i]=300+(300*fabs(value));
1232             }
1233
1234             phaseDenominator += phaseIncrementDenominator;
1235             if (phaseDenominator >= twoPI) {
1236                 phaseDenominator -= twoPI;
1237             }
1238             phaseNumerator += phaseIncrementNumerator;
1239             if (phaseNumerator >= twoPI) {
1240                 phaseNumerator -= twoPI;

```

```

1241         }
1242     }
1243     png::image<png::rgb_pixel> image(44100,600);
1244     Render::fillbackground(image);
1245     Render::drawwave(image,signalY);
1246     image.write(sineName);
1247 }
1248 }
1249
1250 namespace Render
1251 {
1252     void fillbackground(png::image<png::rgb_pixel>& image)
1253     {
1254         png::rgb_pixel px(0x04,0x13,0x31);
1255         for (uint32_t y=0;y<image.get_height();y++) {
1256             for (uint32_t x=0;x<image.get_width();++x) {
1257                 image.set_pixel(x,y,px);
1258             }
1259         }
1260     }
1261
1262     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
1263     {
1264         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
1265     ))
1266         {
1267             png::rgb_pixel px(0x7a,0xb1,0xe3);
1268             image.set_pixel(x,y,px);
1269         }
1270     }
1271
1272     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
1273     {
1274         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
1275         dx = x2 - x1; dy = y2 - y1;
1276         if (dx == 0)
1277         {
1278             if (y2 < y1) std::swap(y1, y2);
1279             for (y = y1; y <= y2; y++)
1280                 drawpx(image, x1, y);
1281             return;
1282         }
1283         if (dy == 0)

```

```

1284         {
1285             if (x2 < x1) std::swap(x1, x2);
1286             for (x = x1; x <= x2; x++)
1287                 drawpx(image, x, y1);
1288             return;
1289         }
1290     dx1 = abs(dx); dy1 = abs(dy);
1291     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
1292     if (dy1 <= dx1)
1293     {
1294         if (dx >= 0)
1295         {
1296             x = x1; y = y1; xe = x2;
1297         }
1298         else
1299         {
1300             x = x2; y = y2; xe = x1;
1301         }
1302         drawpx(image, x, y);
1303         for (i = 0; x<xe; i++)
1304         {
1305             x = x + 1;
1306             if (px<0)
1307                 px = px + 2 * dy1;
1308             else
1309             {
1310                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
1311                 px = px + 2 * (dy1 - dx1);
1312             }
1313             drawpx(image, x, y);
1314         }
1315     }
1316     else
1317     {
1318         if (dy >= 0)
1319         {
1320             x = x1; y = y1; ye = y2;
1321         }
1322         else
1323         {
1324             x = x2; y = y2; ye = y1;
1325         }
1326         drawpx(image, x, y);

```

```

1327         for (i = 0; y<ye; i++)
1328         {
1329             y = y + 1;
1330             if (py <= 0)
1331                 py = py + 2 * dx1;
1332             else
1333             {
1334                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
1335                 py = py + 2 * (dx1 - dy1);
1336             }
1337             drawpx(image, x, y);
1338         }
1339     }
1340 }
1341
1342 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
1343 {
1344     uint32_t y=0, ox=0, oy=0;
1345     for (uint32_t x=0; x<image.get_width(); ++x)
1346     {
1347         y=signalY[x];
1348         if (x == 0) {ox=x; oy=y;}
1349         drawline(image, x, y, ox, oy);
1350         ox=x; oy=y;
1351     }
1352 }
1353 }

```

---

**Envelope.hpp - 3356 bytes.**

```

1  #ifndef ADSR_H
2  #define ADSR_H
3
4  #include <vector>
5  #include <cmath>
6
7  namespace ADSR
8  {
9      class Envelope
10     {
11     public:
12         Envelope(float sampleRate, uint32_t duration)
13         {

```

```

14         sampleLength=sampleRate*duration;
15         A_l=0.2;A_r=0.9;D_l=0.1;D_r=0.7;S_l=0.4;S_r=D_r;R_l=0.3;R_r=0.0;
16     }
17     ~Envelope() {}
18     void generateenvelope(std::vector<float>& env)
19     {
20         env.resize(sampleLength);
21         uint32_t height=600;
22         coords[0]=0;// x1
23         coords[1]=height;// y1
24         coords[2]=sampleLength*A_l;// x2
25         coords[3]=height-(height*A_r);// y2
26         coords[4]=sampleLength*(A_l+D_l);// x3
27         coords[5]=height-(height*D_r);// y3
28         coords[6]=sampleLength*(A_l+D_l+S_l);// x4
29         coords[7]=height-(height*S_r);// y4
30         coords[8]=sampleLength;// x5
31         coords[9]=height;// y5
32         bridge(env,coords[0],coords[1],coords[2],coords[3]);
33         bridge(env,coords[2],coords[3],coords[4],coords[5]);
34         bridge(env,coords[4],coords[5],coords[6],coords[7]);
35         bridge(env,coords[6],coords[7],coords[8],coords[9]);
36     }
37     void generateenvelope2(std::vector<float>& env)
38     {
39         env.resize(sampleLength);
40         uint32_t height=600;
41         coords[0]=0;// x1
42         coords[1]=height;// y1
43         coords[2]=sampleLength;// x5
44         coords[3]=0;// y5
45         bridge(env,coords[0],coords[1],coords[2],coords[3]);
46     }
47     void generateenvelope3(std::vector<float>& env)
48     {
49         env.resize(sampleLength);
50         uint32_t height=600;
51         coords[0]=0;// x1
52         coords[1]=0;// y1
53         coords[2]=sampleLength;// x5
54         coords[3]=height;// y5
55         bridge(env,coords[0],coords[1],coords[2],coords[3]);
56     }

```



```

57     void applyenvelope(std::vector<float>& v, std::vector<float>& env)
58     {
59         for (uint32_t i=0; i<env.size(); ++i)
60         {
61             v[i]=v[i]*env[i];
62         }
63     }
64     private:
65     void setpt(std::vector<float>& env, uint32_t x, uint32_t y)
66     {
67         if (((x >= 0) && (x < sampleLength)) && ((y >= 0) && (y < 600)))
68         {
69             env[x]=((600.0-float(y))/600.0);
70         }
71     }
72
73     void bridge(std::vector<float>& env, uint32_t x1, uint32_t y1, uint32_t x2, uint32_t
74     _t y2)
75     {
76         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
77         dx = x2 - x1; dy = y2 - y1;
78         if (dx == 0)
79         {
80             if (y2 < y1) std::swap(y1, y2);
81             for (y = y1; y <= y2; y++)
82                 setpt(env, x1, y);
83             return;
84         }
85         if (dy == 0)
86         {
87             if (x2 < x1) std::swap(x1, x2);
88             for (x = x1; x <= x2; x++)
89                 setpt(env, x, y1);
90             return;
91         }
92         dx1 = std::abs(dx); dy1 = std::abs(dy);
93         px = 2 * dy1 - dx1; py = 2 * dx1 - dy1;
94         if (dy1 <= dx1)
95         {
96             if (dx >= 0)
97             {
98                 x = x1; y = y1; xe = x2;
99             }

```

```

100         else
101         {
102             x = x2; y = y2; xe = x1;
103         }
104         setpt(env, x, y);
105         for (i = 0; x < xe; i++)
106         {
107             x = x + 1;
108             if (px < 0)
109                 px = px + 2 * dy1;
110             else
111             {
112                 if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) y = y + 1; else
113                 px = px + 2 * (dy1 - dx1);
114             }
115             setpt(env, x, y);
116         }
117     }
118     else
119     {
120         if (dy >= 0)
121         {
122             x = x1; y = y1; ye = y2;
123         }
124         else
125         {
126             x = x2; y = y2; ye = y1;
127         }
128         setpt(env, x, y);
129         for (i = 0; y < ye; i++)
130         {
131             y = y + 1;
132             if (py <= 0)
133                 py = py + 2 * dx1;
134             else
135             {
136                 if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) x = x + 1; else
137                 py = py + 2 * (dx1 - dy1);
138             }
139             setpt(env, x, y);
140         }
141     }
142 }

```

```

143
144         float A_l;
145         float A_r;
146         float D_l;
147         float D_r;
148         float S_l;
149         float S_r;
150         float R_l;
151         float R_r;
152     protected:
153         uint32_t sampleLength;
154         uint32_t coords[10];
155     };
156 }
157 #endif

```

---

#### MoogFilter.hpp - 1628 bytes.

---

```

1  #ifndef MOOG_H
2  #define MOOG_H
3
4  #define MOOG_PI      3.14159265358979323846264338327950288
5  #include <vector>
6
7  namespace Moog
8  {
9      class MoogFilter
10     {
11     public:
12         MoogFilter(float sampleRate) : sampleRate(sampleRate)
13         {
14             memset(stage, 0, sizeof(stage));
15             memset(delay, 0, sizeof(delay));
16             SetCutoff(1000.0f);
17             SetResonance(0.10f);
18         }
19         ~MoogFilter() {}
20
21         void Process(std::vector<float>& samples, uint32_t n)
22         {
23             for (int s = 0; s < n; ++s)
24             {
25                 float x = samples[s] - resonance * stage[3];

```

```

26
27 // Four cascaded one-pole filters (bilinear transform)
28 stage[0] = x * p + delay[0] * p - k * stage[0];
29 stage[1] = stage[0] * p + delay[1] * p - k * stage[1];
30 stage[2] = stage[1] * p + delay[2] * p - k * stage[2];
31 stage[3] = stage[2] * p + delay[3] * p - k * stage[3];
32
33 // Clipping band-limited sigmoid
34 stage[3] -= (stage[3] * stage[3] * stage[3]) / 6.0;
35
36 delay[0] = x;
37 delay[1] = stage[0];
38 delay[2] = stage[1];
39 delay[3] = stage[2];
40
41 samples[s] = stage[3];
42     }
43 }
44
45 void SetResonance(float r)
46 {
47     resonance = r * (t2 + 6.0 * t1) / (t2 - 6.0 * t1);
48 }
49
50 void SetCutoff(float c)
51 {
52     cutoff = 2.0 * c / sampleRate;
53
54     p = cutoff * (1.8 - 0.8 * cutoff);
55     k = 2.0 * sin(cutoff * MOOG_PI * 0.5) - 1.0;
56     t1 = (1.0 - p) * 1.386249;
57     t2 = 12.0 + t1 * t1;
58
59     SetResonance(resonance);
60 }
61
62 float GetResonance() { return resonance; }
63 float GetCutoff() { return cutoff; }
64 private:
65     double stage[4];
66     double delay[4];
67
68     double p;

```

```

69         double k;
70         double t1;
71         double t2;
72     protected:
73         float cutoff;
74         float resonance;
75         float sampleRate;
76     };
77 }
78 #endif

```

---

### 01-additive.cpp - 21478 bytes.

```

1 // compile: clang++ -std=c++20 -lpng 01-additive.cpp -o 01-additive
2
3 #define _USE_MATH_DEFINES
4 #include <cmath>
5 #include <vector>
6 #include <random>
7 #include <filesystem>
8 #include "indicators.hpp"
9 #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "Envelope.hpp"
12
13 namespace Render
14 {
15     void fillbackground(png::image<png::rgb_pixel>& image);
16     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
17     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
18     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
19     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std::\
20 :vector<uint32_t>& v2);
21     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
22     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
23     void saveimagefile(std::vector<float>& v2, std::string filename);
24     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
25 v1, std::vector<uint32_t>& v2);
26     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
27     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
28 }
29
30 namespace SignalGenerators

```

```
31 {
32     void gain(std::vector<float>& v, double gain);
33     void normalize(std::vector<float>& v);
34     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
35     void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz);
36     void generatenoise(std::vector<float>& v, int duration);
37     void saveaudiofile(std::vector<float>& v, std::string filename);
38 }
39
40 int main()
41 {
42     namespace fs = std::filesystem;
43     fs::create_directory("additive");
44
45     const int duration=1;
46     const double sampleRate=44100.0;
47     std::vector<float> osc1;
48     std::vector<float> osc2;
49     std::vector<float> osc3;
50     std::vector<float> osc4;
51     std::vector<float> osc5;
52     std::vector<float> osc6;
53     std::vector<float> osc7;
54     std::vector<float> osc8;
55     std::vector<float> osc9;
56     std::vector<float> osc10;
57     std::vector<float> osc11;
58     std::vector<float> osc12;
59     std::vector<float> osc13;
60     std::vector<float> osc14;
61     std::vector<float> osc15;
62     std::vector<float> osc16;
63     std::vector<float> osc17;
64     std::vector<float> osc18;
65     std::vector<float> osc19;
66     std::vector<float> osc20;
67     std::vector<float> osc21;
68     std::vector<float> osc22;
69     std::vector<float> osc23;
70     std::vector<float> osc24;
71     std::vector<float> osc25;
72     std::vector<float> osc26;
73     std::vector<float> osc27;
```

```
74     std::vector<float> osc28;
75     std::vector<float> osc29;
76     std::vector<float> osc30;
77     std::vector<float> osc31;
78     std::vector<float> osc32;
79     std::vector<float> osc33;
80     std::vector<float> mixer;
81     std::vector<float> envelope;
82
83     using namespace indicators;
84     // Hide cursor
85     show_console_cursor(false);
86
87     // Setup ProgressBar
88     ProgressBar bar{
89         option::BarWidth{50},
90         option::Start{"["},
91         option::Fill{"█"},
92         option::Lead{"█"},
93         option::Remainder{"-"},
94         option::End{" ]"},
95         option::PostfixText{"Setting: Sine Oscillator1 @ 55 Hz 1/35"},
96         option::ForegroundColor{Color::cyan},
97         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
98     };
99
100    // Update progress
101    bar.set_progress(0);
102
103    SignalGenerators::generatesinewave(osc1,duration,55);
104    SignalGenerators::gain(osc1,0.5);
105    SignalGenerators::saveaudiofile(osc1,"01-osc1.aiff");
106    Render::saveimagefile(osc1,"02-osc1.png");
107
108    // Update progress
109    bar.set_progress(3);
110    bar.set_option(option::PostfixText{"Setting: Sine Oscillator2 @ 110 Hz 2/35"});
111
112    SignalGenerators::generatesinewave(osc2,duration,110);
113    SignalGenerators::gain(osc2,0.8);
114    SignalGenerators::saveaudiofile(osc2,"03-osc2.aiff");
115    Render::saveimagefile(osc2,"04-osc2.png");
116
```

```
117     // Update progress
118     bar.set_progress(6);
119     bar.set_option(option::PostfixText{"Setting: Sine Oscillator3 @ 165 Hz 3/35"});
120
121     SignalGenerators::generatesinewave(osc3,duration,165);
122     SignalGenerators::gain(osc3,0.7);
123     SignalGenerators::saveaudiofile(osc3,"05-osc3.aiff");
124     Render::saveimagefile(osc3,"06-osc3.png");
125
126     // Update progress
127     bar.set_progress(8);
128     bar.set_option(option::PostfixText{"Setting: Sine Oscillator4 @ 220 Hz 4/35"});
129
130     SignalGenerators::generatesinewave(osc4,duration,220);
131     SignalGenerators::gain(osc4,0.8);
132     SignalGenerators::saveaudiofile(osc4,"07-osc4.aiff");
133     Render::saveimagefile(osc4,"08-osc4.png");
134
135     // Update progress
136     bar.set_progress(11);
137     bar.set_option(option::PostfixText{"Setting: Sine Oscillator5 @ 275 Hz 5/35"});
138
139     SignalGenerators::generatesinewave(osc5,duration,275);
140     SignalGenerators::gain(osc5,0.4);
141     SignalGenerators::saveaudiofile(osc5,"09-osc5.aiff");
142     Render::saveimagefile(osc5,"10-osc5.png");
143
144     // Update progress
145     bar.set_progress(14);
146     bar.set_option(option::PostfixText{"Setting: Sine Oscillator6 @ 330 Hz 6/35"});
147
148     SignalGenerators::generatesinewave(osc6,duration,330);
149     SignalGenerators::gain(osc6,0.7);
150     SignalGenerators::saveaudiofile(osc6,"11-osc6.aiff");
151     Render::saveimagefile(osc6,"12-osc6.png");
152
153     // Update progress
154     bar.set_progress(17);
155     bar.set_option(option::PostfixText{"Setting: Sine Oscillator7 @ 385 Hz 7/35"});
156
157     SignalGenerators::generatesinewave(osc7,duration,385);
158     SignalGenerators::gain(osc7,0.3);
159     SignalGenerators::saveaudiofile(osc7,"13-osc7.aiff");
```



```
160     Render::saveimagefile(osc7, "14-osc7.png");
161
162     // Update progress
163     bar.set_progress(20);
164     bar.set_option(option::PostfixText{"Setting: Sine Oscillator8 @ 440 Hz 8/35"});
165
166     SignalGenerators::generatesinewave(osc8,duration,440);
167     SignalGenerators::gain(osc8,0.9);
168     SignalGenerators::saveaudiofile(osc8, "15-osc8.aiff");
169     Render::saveimagefile(osc8, "16-osc8.png");
170
171     // Update progress
172     bar.set_progress(23);
173     bar.set_option(option::PostfixText{"Setting: Sine Oscillator9 @ 495 Hz 9/35"});
174
175     SignalGenerators::generatesinewave(osc9,duration,495);
176     SignalGenerators::gain(osc9,0.5);
177     SignalGenerators::saveaudiofile(osc9, "17-osc8.aiff");
178     Render::saveimagefile(osc9, "18-osc8.png");
179
180     // Update progress
181     bar.set_progress(26);
182     bar.set_option(option::PostfixText{"Setting: Sine Oscillator10 @ 550 Hz 10/35"});
183
184     SignalGenerators::generatesinewave(osc10,duration,550);
185     SignalGenerators::gain(osc10,0.5);
186     SignalGenerators::saveaudiofile(osc10, "19-osc8.aiff");
187     Render::saveimagefile(osc10, "20-osc8.png");
188
189     // Update progress
190     bar.set_progress(29);
191     bar.set_option(option::PostfixText{"Setting: Sine Oscillator11 @ 605 Hz 11/35"});
192
193     SignalGenerators::generatesinewave(osc11,duration,605);
194     SignalGenerators::gain(osc11,0.5);
195     SignalGenerators::saveaudiofile(osc11, "21-osc8.aiff");
196     Render::saveimagefile(osc11, "22-osc8.png");
197
198     // Update progress
199     bar.set_progress(31);
200     bar.set_option(option::PostfixText{"Setting: Sine Oscillator12 @ 660 Hz 12/35"});
201
202     SignalGenerators::generatesinewave(osc12,duration,660);
```

```
203     SignalGenerators::gain(osc12,0.5);
204     SignalGenerators::saveaudiofile(osc12,"23-osc8.aiff");
205     Render::saveimagefile(osc12,"24-osc8.png");
206
207     // Update progress
208     bar.set_progress(34);
209     bar.set_option(option::PostfixText{"Setting: Sine Oscillator13 @ 715 Hz 13/35"});
210
211     SignalGenerators::generatesinewave(osc13,duration,715);
212     SignalGenerators::gain(osc13,0.7);
213     SignalGenerators::saveaudiofile(osc13,"25-osc8.aiff");
214     Render::saveimagefile(osc13,"26-osc8.png");
215
216     // Update progress
217     bar.set_progress(37);
218     bar.set_option(option::PostfixText{"Setting: Sine Oscillator14 @ 770 Hz 14/35"});
219
220     SignalGenerators::generatesinewave(osc14,duration,770);
221     SignalGenerators::gain(osc14,0.7);
222     SignalGenerators::saveaudiofile(osc14,"27-osc8.aiff");
223     Render::saveimagefile(osc14,"28-osc8.png");
224
225     // Update progress
226     bar.set_progress(40);
227     bar.set_option(option::PostfixText{"Setting: Sine Oscillator15 @ 825 Hz 15/35"});
228
229     SignalGenerators::generatesinewave(osc15,duration,825);
230     SignalGenerators::gain(osc15,0.9);
231     SignalGenerators::saveaudiofile(osc15,"29-osc8.aiff");
232     Render::saveimagefile(osc15,"30-osc8.png");
233
234     // Update progress
235     bar.set_progress(43);
236     bar.set_option(option::PostfixText{"Setting: Sine Oscillator16 @ 880 Hz 16/35"});
237
238     SignalGenerators::generatesinewave(osc16,duration,880);
239     SignalGenerators::gain(osc16,0.9);
240     SignalGenerators::saveaudiofile(osc16,"31-osc8.aiff");
241     Render::saveimagefile(osc16,"32-osc8.png");
242
243     // Update progress
244     bar.set_progress(46);
245     bar.set_option(option::PostfixText{"Setting: Sine Oscillator17 @ 935 Hz 17/35"});
```

```
246
247     SignalGenerators::generatesinewave(osc17,duration,935);
248     SignalGenerators::gain(osc17,0.8);
249     SignalGenerators::saveaudiofile(osc17,"33-osc8.aiff");
250     Render::saveimagefile(osc17,"34-osc8.png");
251
252     // Update progress
253     bar.set_progress(49);
254     bar.set_option(option::PostfixText{"Setting: Sine Oscillator18 @ 990 Hz 18/35"});
255
256     SignalGenerators::generatesinewave(osc18,duration,990);
257     SignalGenerators::gain(osc18,0.8);
258     SignalGenerators::saveaudiofile(osc18,"35-osc8.aiff");
259     Render::saveimagefile(osc18,"36-osc8.png");
260
261     // Update progress
262     bar.set_progress(51);
263     bar.set_option(option::PostfixText{"Setting: Sine Oscillator19 @ 1045 Hz 19/35"});
264
265     SignalGenerators::generatesinewave(osc19,duration,1045);
266     SignalGenerators::gain(osc19,0.7);
267     SignalGenerators::saveaudiofile(osc19,"37-osc8.aiff");
268     Render::saveimagefile(osc19,"38-osc8.png");
269
270     // Update progress
271     bar.set_progress(54);
272     bar.set_option(option::PostfixText{"Setting: Sine Oscillator20 @ 1100 Hz 20/35"});
273
274     SignalGenerators::generatesinewave(osc20,duration,1100);
275     SignalGenerators::gain(osc20,0.7);
276     SignalGenerators::saveaudiofile(osc20,"39-osc8.aiff");
277     Render::saveimagefile(osc20,"40-osc8.png");
278
279     // Update progress
280     bar.set_progress(57);
281     bar.set_option(option::PostfixText{"Setting: Sine Oscillator21 @ 1155 Hz 21/35"});
282
283     SignalGenerators::generatesinewave(osc21,duration,1155);
284     SignalGenerators::gain(osc21,0.6);
285     SignalGenerators::saveaudiofile(osc21,"41-osc8.aiff");
286     Render::saveimagefile(osc21,"42-osc8.png");
287
288     // Update progress
```

```
289     bar.set_progress(60);
290     bar.set_option(option::PostfixText{"Setting: Sine Oscillator22 @ 1210 Hz 22/35"});
291
292     SignalGenerators::generatesinewave(osc22,duration,1210);
293     SignalGenerators::gain(osc22,0.6);
294     SignalGenerators::saveaudiofile(osc22,"43-osc8.aiff");
295     Render::saveimagefile(osc22,"44-osc8.png");
296
297     // Update progress
298     bar.set_progress(63);
299     bar.set_option(option::PostfixText{"Setting: Sine Oscillator23 @ 1265 Hz 23/35"});
300
301     SignalGenerators::generatesinewave(osc23,duration,1265);
302     SignalGenerators::gain(osc23,0.5);
303     SignalGenerators::saveaudiofile(osc23,"45-osc8.aiff");
304     Render::saveimagefile(osc23,"46-osc8.png");
305
306     // Update progress
307     bar.set_progress(66);
308     bar.set_option(option::PostfixText{"Setting: Sine Oscillator24 @ 1320 Hz 24/35"});
309
310     SignalGenerators::generatesinewave(osc24,duration,1320);
311     SignalGenerators::gain(osc24,0.5);
312     SignalGenerators::saveaudiofile(osc24,"47-osc8.aiff");
313     Render::saveimagefile(osc24,"48-osc8.png");
314
315     // Update progress
316     bar.set_progress(69);
317     bar.set_option(option::PostfixText{"Setting: Sine Oscillator25 @ 1375 Hz 25/35"});
318
319     SignalGenerators::generatesinewave(osc25,duration,1375);
320     SignalGenerators::gain(osc25,0.4);
321     SignalGenerators::saveaudiofile(osc25,"49-osc8.aiff");
322     Render::saveimagefile(osc25,"50-osc8.png");
323
324     // Update progress
325     bar.set_progress(71);
326     bar.set_option(option::PostfixText{"Setting: Sine Oscillator26 @ 1430 Hz 26/11"});
327
328     SignalGenerators::generatesinewave(osc26,duration,1430);
329     SignalGenerators::gain(osc26,0.4);
330     SignalGenerators::saveaudiofile(osc26,"51-osc8.aiff");
331     Render::saveimagefile(osc26,"52-osc8.png");
```

```
332
333     // Update progress
334     bar.set_progress(74);
335     bar.set_option(option::PostfixText{"Setting: Sine Oscillator27 @ 1485 Hz 27/35"});
336
337     SignalGenerators::generatesinewave(osc27,duration,1485);
338     SignalGenerators::gain(osc27,0.3);
339     SignalGenerators::saveaudiofile(osc27,"53-osc8.aiff");
340     Render::saveimagefile(osc27,"54-osc8.png");
341
342     // Update progress
343     bar.set_progress(77);
344     bar.set_option(option::PostfixText{"Setting: Sine Oscillator28 @ 1540 Hz 28/35"});
345
346     SignalGenerators::generatesinewave(osc28,duration,1540);
347     SignalGenerators::gain(osc28,0.3);
348     SignalGenerators::saveaudiofile(osc28,"55-osc8.aiff");
349     Render::saveimagefile(osc28,"56-osc8.png");
350
351     // Update progress
352     bar.set_progress(80);
353     bar.set_option(option::PostfixText{"Setting: Sine Oscillator29 @ 1595 Hz 29/35"});
354
355     SignalGenerators::generatesinewave(osc29,duration,1595);
356     SignalGenerators::gain(osc29,0.3);
357     SignalGenerators::saveaudiofile(osc29,"57-osc8.aiff");
358     Render::saveimagefile(osc29,"58-osc8.png");
359
360     // Update progress
361     bar.set_progress(83);
362     bar.set_option(option::PostfixText{"Setting: Sine Oscillator30 @ 1650 Hz 30/35"});
363
364     SignalGenerators::generatesinewave(osc30,duration,1650);
365     SignalGenerators::gain(osc30,0.3);
366     SignalGenerators::saveaudiofile(osc30,"59-osc8.aiff");
367     Render::saveimagefile(osc30,"60-osc8.png");
368
369     // Update progress
370     bar.set_progress(86);
371     bar.set_option(option::PostfixText{"Setting: Sine Oscillator31 @ 1705 Hz 31/35"});
372
373     SignalGenerators::generatesinewave(osc31,duration,1705);
374     SignalGenerators::gain(osc31,0.3);
```

```
375     SignalGenerators::saveaudiofile(osc31, "61-osc8.aiff");
376     Render::saveimagefile(osc31, "62-osc8.png");
377
378     // Update progress
379     bar.set_progress(89);
380     bar.set_option(option::PostfixText{"Setting: Sine Oscillator32 @ 1760 Hz 32/35"});
381
382     SignalGenerators::generatesinewave(osc32, duration, 1760);
383     SignalGenerators::gain(osc32, 0.3);
384     SignalGenerators::saveaudiofile(osc32, "63-osc8.aiff");
385     Render::saveimagefile(osc32, "64-osc8.png");
386
387     // Update progress
388     bar.set_progress(91);
389     bar.set_option(option::PostfixText{"Setting: Noise Oscillator33 33/35"});
390
391     SignalGenerators::generatenoise(osc33, duration);
392     SignalGenerators::gain(osc33, 0.1);
393     SignalGenerators::saveaudiofile(osc33, "65-osc9.aiff");
394     Render::saveimagefile(osc33, "66-osc9.png");
395
396     // Update progress
397     bar.set_progress(94);
398     bar.set_option(option::PostfixText{"Mixing all 33 oscillators 34/35"});
399
400     mixer.resize(duration*sampleRate);
401     SignalGenerators::addwaves(osc1, osc2, mixer);
402     SignalGenerators::addwaves(mixer, osc3, mixer);
403     SignalGenerators::addwaves(mixer, osc4, mixer);
404     SignalGenerators::addwaves(mixer, osc5, mixer);
405     SignalGenerators::addwaves(mixer, osc6, mixer);
406     SignalGenerators::addwaves(mixer, osc7, mixer);
407     SignalGenerators::addwaves(mixer, osc8, mixer);
408     SignalGenerators::addwaves(mixer, osc9, mixer);
409     SignalGenerators::addwaves(mixer, osc10, mixer);
410     SignalGenerators::addwaves(mixer, osc11, mixer);
411     SignalGenerators::addwaves(mixer, osc12, mixer);
412     SignalGenerators::addwaves(mixer, osc13, mixer);
413     SignalGenerators::addwaves(mixer, osc14, mixer);
414     SignalGenerators::addwaves(mixer, osc15, mixer);
415     SignalGenerators::addwaves(mixer, osc16, mixer);
416     SignalGenerators::addwaves(mixer, osc17, mixer);
417     SignalGenerators::addwaves(mixer, osc18, mixer);
```

```
418     SignalGenerators::addwaves(mixer,osc19,mixer);
419     SignalGenerators::addwaves(mixer,osc20,mixer);
420     SignalGenerators::addwaves(mixer,osc21,mixer);
421     SignalGenerators::addwaves(mixer,osc22,mixer);
422     SignalGenerators::addwaves(mixer,osc23,mixer);
423     SignalGenerators::addwaves(mixer,osc24,mixer);
424     SignalGenerators::addwaves(mixer,osc25,mixer);
425     SignalGenerators::addwaves(mixer,osc26,mixer);
426     SignalGenerators::addwaves(mixer,osc27,mixer);
427     SignalGenerators::addwaves(mixer,osc28,mixer);
428     SignalGenerators::addwaves(mixer,osc29,mixer);
429     SignalGenerators::addwaves(mixer,osc30,mixer);
430     SignalGenerators::addwaves(mixer,osc31,mixer);
431     SignalGenerators::addwaves(mixer,osc32,mixer);
432     SignalGenerators::addwaves(mixer,osc33,mixer);
433     SignalGenerators::normalize(mixer);
434     SignalGenerators::saveaudiofile(mixer,"67-mixed.aiff");
435     Render::saveimagefile(mixer,"68-mixed.png");
436
437     // Update progress
438     bar.set_progress(97);
439     bar.set_option(option::PostfixText{"Adding ADSR Envelope 35/35"});
440
441     ADSR::Envelope env(sampleRate,duration);
442     env.generateenvelope(envelope);
443     env.applyenvelope(mixer,envelope);
444     SignalGenerators::normalize(mixer);
445     Render::saveenvelopeimage(envelope,"69-envelope.png");
446     SignalGenerators::saveaudiofile(mixer,"70-mixednveloped.aiff");
447     Render::saveimagefile(mixer,"71-mixednveloped.png");
448
449     // Update progress
450     bar.set_progress(100);
451     bar.set_option(option::PostfixText{"Done 35/35"});
452
453     // Show cursor
454     show_console_cursor(true);
455     return 0;
456 }
457
458 namespace SignalGenerators
459 {
460     void gain(std::vector<float>& v, double gain)
```

```

461     {
462         for (uint32_t i=0;i<v.size();++i) {
463             v[i]=v[i]*gain;
464         }
465     }
466
467 void normalize(std::vector<float>& v)
468 {
469     float max=0.0,value=0.0;
470     for (uint32_t i=0;i<v.size();++i) {
471         value=v[i];
472         if (value > max) {max=value;}
473     }
474     // max=std::ceil(max);
475     for (uint32_t i=0;i<v.size();++i) {
476         // v[i]=v[i]/max;
477         v[i]=(v[i]/max)*0.707;
478     }
479 }
480
481 void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
482 {
483     for (uint32_t i=0;i<v1.size();++i)
484     {
485         v3[i]=v1[i]+v2[i];
486     }
487 }
488
489 void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz)
490 {
491     const double sampleRate=44100.0;
492     for (uint32_t i=0;i<sampleRate*duration;++i) {
493         v.push_back(sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2.0
494 PI));
495     }
496 }
497
498 void generatenoise(std::vector<float>& v, int duration)
499 {
500     const double sampleRate=44100.0;
501     std::random_device rd;
502     std::mt19937 gen(rd());
503     std::uniform_real_distribution<> dis(-1.0, 1.0);

```



```

504
505         for (uint32_t i=0;i<sampleRate*duration;++i) {
506             v.push_back(dis(gen));
507         }
508     }
509
510     void saveaudiofile(std::vector<float>& v, std::string filename)
511     {
512         const std::string path="additive/";
513         // Setup the audio file
514         AudioFile<float> a;
515         a.setNumChannels(1);
516         a.setBitDepth(24);
517         a.setNumSamplesPerChannel(44100);
518
519         for (int i=0;i<a.getNumSamplesPerChannel();++i)
520         {
521             for (int channel=0;channel<a.getNumChannels();++channel)
522             {
523                 a.samples[channel][i]=v[i];
524             }
525         }
526         a.save(path+filename,AudioFileFormat::Aiff);
527     }
528 }
529
530 namespace Render
531 {
532     void fillbackground(png::image<png::rgb_pixel>& image)
533     {
534         png::rgb_pixel px(0x04,0x13,0x31);
535         for (uint32_t y=0;y<image.get_height();y++) {
536             for (uint32_t x=0;x<image.get_width();++x) {
537                 image.set_pixel(x,y,px);
538             }
539         }
540     }
541
542     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
543     {
544         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
545     ))
546         {

```

```
547         png::rgb_pixel px(0x7a,0xb1,0xe3);
548         image.set_pixel(x,y,px);
549     }
550 }
551
552 void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
553 {
554     int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
555     dx = x2 - x1; dy = y2 - y1;
556     if (dx == 0)
557     {
558         if (y2 < y1) std::swap(y1, y2);
559         for (y = y1; y <= y2; y++)
560             drawpx(image, x1, y);
561         return;
562     }
563     if (dy == 0)
564     {
565         if (x2 < x1) std::swap(x1, x2);
566         for (x = x1; x <= x2; x++)
567             drawpx(image, x, y1);
568         return;
569     }
570     dx1 = abs(dx); dy1 = abs(dy);
571     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
572     if (dy1 <= dx1)
573     {
574         if (dx >= 0)
575         {
576             x = x1; y = y1; xe = x2;
577         }
578         else
579         {
580             x = x2; y = y2; xe = x1;
581         }
582         drawpx(image, x, y);
583         for (i = 0; x<xe; i++)
584         {
585             x = x + 1;
586             if (px<0)
587                 px = px + 2 * dy1;
588             else
589                 {
```

```

590         if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
591         px = px + 2 * (dy1 - dx1);
592     }
593     drawpx(image, x, y);
594 }
595 }
596 else
597 {
598     if (dy >= 0)
599     {
600         x = x1; y = y1; ye = y2;
601     }
602     else
603     {
604         x = x2; y = y2; ye = y1;
605     }
606     drawpx(image, x, y);
607     for (i = 0; y<ye; i++)
608     {
609         y = y + 1;
610         if (py <= 0)
611             py = py + 2 * dx1;
612         else
613         {
614             if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
615             py = py + 2 * (dx1 - dy1);
616         }
617         drawpx(image, x, y);
618     }
619 }
620 }
621
622 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
623 {
624     uint32_t y=0, ox=0, oy=0;
625     for (uint32_t x=0; x<image.get_width(); ++x)
626     {
627         y=signalY[x];
628         if (x == 0) {ox=x;oy=y;}
629         drawline(image, x, y, ox, oy);
630         ox=x;oy=y;
631     }
632 }

```

```
633
634     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
635 :vector<uint32_t>& v2)
636     {
637         uint32_t halfHeight=image.get_height()/2;
638         double value=0.0;
639         if (v2.size() == 0 || v2.size() > v1.size()) {
640             v2.resize(v1.size());
641         }
642
643         for (uint32_t i=0;i<v1.size();++i)
644         {
645             value=v1[i];
646             if (value >= 0.0) {
647                 v2[i]=halfHeight-(halfHeight*value);
648             } else if (value < 0.0)
649             {
650                 v2[i]=halfHeight+(halfHeight*fabs(value));
651             }
652         }
653     }
654
655     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
656     {
657         fillbackground(image);
658         drawwave(image,v);
659     }
660
661     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
662     {
663         const std::string path="additive/";
664         png::image<png::rgb_pixel> image(44100,600);
665         renderimage(image,v);
666         image.write(path+filename);
667     }
668
669     void saveimagefile(std::vector<float>& v2, std::string filename)
670     {
671         const std::string path="additive/";
672         png::image<png::rgb_pixel> image(44100,600);
673         std::vector<uint32_t> v;
674         normalizedtoimg(image,v2,v);
675         renderimage(image,v);
```

```
676         image.write(path+filename);
677     }
678
679     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
680 v1, std::vector<uint32_t>& v2)
681     {
682         uint32_t height=image.get_height();
683         if (v2.size() == 0 || v2.size() > v1.size()) {
684             v2.resize(v1.size());
685         }
686         for (uint32_t i=0; i<v1.size(); ++i)
687         {
688             v2[i]=height-(height*v1[i]);
689         }
690     }
691
692     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
693     {
694         const std::string path="additive/";
695         png::image<png::rgb_pixel> image(44100,600);
696         renderimage(image,v);
697         image.write(path+filename);
698     }
699
700     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
701     {
702         const std::string path="additive/";
703         png::image<png::rgb_pixel> image(44100,600);
704         std::vector<uint32_t> v;
705         normalizedenvelopetoimg(image,v2,v);
706         renderimage(image,v);
707         image.write(path+filename);
708     }
709
710 }
```

---

**02-subtractive.cpp - 14546 bytes.**


---

```

1 // compile: clang++ -std=c++20 -lpng 02-subtractive.cpp -o 02-subtractive
2
3 #define _USE_MATH_DEFINES
4 #include <cmath>
5 #include <vector>
6 #include <random>
7 #include <filesystem>
8 #include "indicators.hpp"
9 #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "MoogFilter.hpp"
12 #include "Envelope.hpp"
13
14
15 #include <iostream>
16
17 namespace Render
18 {
19     void fillbackground(png::image<png::rgb_pixel>& image);
20     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
21     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
22     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
23     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
24 :vector<uint32_t>& v2);
25     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
26     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
27     void saveimagefile(std::vector<float>& v2, std::string filename);
28     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
29 v1, std::vector<uint32_t>& v2);
30     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
31     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
32 }
33
34 namespace SignalGenerators
35 {
36     void gain(std::vector<float>& v, double gain);
37     void normalize(std::vector<float>& v);
38     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
39     void generatetrianglwave(std::vector<float>& v, int duration, double frequencyInHz\
40 );
41     void generateinversesawtoothwave(std::vector<float>& v, int duration, double freque\
42 ncyInHz);

```

```
43     void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz\  
44 );  
45     void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz);  
46     void generatethirtyfivesquarewave(std::vector<float>& v, int duration, double frequ\  
47 encyInHz);  
48     void generatetwentyfivesquarewave(std::vector<float>& v, int duration, double frequ\  
49 encyInHz);  
50     void generatenoise(std::vector<float>& v, int duration);  
51     void generatelfo(std::vector<float>& v, int osc, int duration, double frequencyInHz\  
52 );  
53     void generateoscillator(std::vector<float>& v, int osc, int duration, double freque\  
54 ncyInHz);  
55     void saveaudiofile(std::vector<float>& v, std::string filename);  
56 }  
57  
58 int main()  
59 {  
60     namespace fs = std::filesystem;  
61     fs::create_directory("subtractive");  
62  
63     enum osc: int { triangle, inversesaw, saw, square, thirty, twenty, noise };  
64     const int duration=1;  
65     const double sampleRate=44100.0;  
66     const double noiseGain=0.5;  
67     std::vector<float> osc1;  
68     std::vector<float> osc2;  
69     std::vector<float> osc3;  
70     std::vector<float> osc4;  
71     std::vector<float> mixer;  
72     std::vector<float> envelope;  
73     Moog::MoogFilter mf(44100.0);  
74  
75     using namespace indicators;  
76     // Hide cursor  
77     show_console_cursor(false);  
78  
79     // Setup ProgressBar  
80     ProgressBar bar{  
81         option::BarWidth{50},  
82         option::Start{"["},  
83         option::Fill{"█"},  
84         option::Lead{"█"},  
85         option::Remainder{"-"},
```

```

86         option::End{ " ]"},
87         option::PostfixText{"Setting: Oscillator 1 to Triangle Wave @ 440 Hz 1/8"},
88         option::ForegroundColor{Color::cyan},
89         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
90     };
91
92     // Update progress
93     bar.set_progress(0);
94
95     SignalGenerators::generateoscillator(osc1, triangle, duration, 440);
96
97     // Update progress
98     bar.set_progress(12);
99     bar.set_option(option::PostfixText{"Setting: Oscillator 2 to Triangle Wave @ 880 Hz\
100 2/8"});
101
102     SignalGenerators::generateoscillator(osc2, triangle, duration, 440*2);
103
104     // Update progress
105     bar.set_progress(25);
106     bar.set_option(option::PostfixText{"Setting: Oscillator 3 to Triangle Wave @ 1320 H\
107 z 3/8"});
108
109     SignalGenerators::generateoscillator(osc3, triangle, duration, 440*3);
110
111     // Update progress
112     bar.set_progress(37);
113     bar.set_option(option::PostfixText{"Setting: Oscillator 4 (The White Noise Oscillat\
114 or) 4/8"});
115
116     SignalGenerators::generateoscillator(osc4, noise, duration, 440);
117
118     // SignalGenerators::generatelfo(lfo, inversesaw, duration, 15);
119     // SignalGenerators::addwaves(osc3, lfo, osc3);
120
121     SignalGenerators::gain(osc4, noiseGain);
122
123     // Update progress
124     bar.set_progress(50);
125     bar.set_option(option::PostfixText{"Setting Oscillator 4 Gain to: 50% 5/8"});
126
127     mixer.resize(duration*sampleRate);
128     SignalGenerators::addwaves(osc1, osc2, mixer);

```



```
129     SignalGenerators::normalize(mixer);
130     SignalGenerators::addwaves(mixer,osc3,mixer);
131     SignalGenerators::normalize(mixer);
132     Render::saveimagefile(mixer,"01-mixerpreosc4.png");
133     SignalGenerators::saveaudiofile(mixer,"02-mixerpreosc4.aiff");
134     SignalGenerators::addwaves(mixer,osc4,mixer);
135     SignalGenerators::normalize(mixer);
136     Render::saveimagefile(mixer,"03-mixerpostosc4.png");
137     SignalGenerators::saveaudiofile(mixer,"04-mixerpostosc4.aiff");
138
139     // Update progress
140     bar.set_progress(62);
141     bar.set_option(option::PostfixText{"Mixed Oscillator 1 to 4 together 6/8"});
142
143     Render::saveimagefile(mixer,"05-prefilter.png");
144     SignalGenerators::saveaudiofile(mixer,"06-prefilter.aiff");
145
146     mf.Process(mixer,mixer.size());
147
148     SignalGenerators::normalize(mixer);
149     Render::saveimagefile(mixer,"07-postfilter.png");
150     SignalGenerators::saveaudiofile(mixer,"08-postfilter.aiff");
151
152     // Update progress
153     bar.set_progress(75);
154     bar.set_option(option::PostfixText{"Adding MoogFilter to the mixed signal 7/8"});
155
156     ADSR::Envelope env(sampleRate,duration);
157     env.generateenvelope(envelope);
158     Render::saveenvelopeimage(envelope,"09-envelope.png");
159
160     // Update progress
161     bar.set_progress(87);
162     bar.set_option(option::PostfixText{"Adding ADSR Envelope to the mixed signal 8/8"});
163
164     env.applyenvelope(mixer,envelope);
165     Render::saveimagefile(mixer,"10-envelopedmix.png");
166     SignalGenerators::saveaudiofile(mixer,"11-envelopedmix.aiff");
167
168     // Update progress
169     bar.set_progress(100);
170     bar.set_option(option::PostfixText{"Done 8/8"});
171
```

```
172     // Show cursor
173     show_console_cursor(true);
174     return 0;
175 }
176
177 namespace SignalGenerators
178 {
179     void gain(std::vector<float>& v, double gain)
180     {
181         for (uint32_t i=0;i<v.size();++i) {
182             v[i]=v[i]*gain;
183         }
184     }
185
186     void normalize(std::vector<float>& v)
187     {
188         float max=0.0,value=0.0;
189         for (uint32_t i=0;i<v.size();++i) {
190             value=v[i];
191             if (value > max) {max=value;}
192         }
193         // max=std::ceil(max);
194         for (uint32_t i=0;i<v.size();++i) {
195             // v[i]=v[i]/max;
196             v[i]=(v[i]/max)*0.707;
197         }
198     }
199
200     void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
201     {
202         for (uint32_t i=0;i<v1.size();++i)
203         {
204             v3[i]=v1[i]+v2[i];
205         }
206     }
207
208     void generatetrianglwave(std::vector<float>& v, int duration, double frequencyInHz)
209     {
210         const double sampleRate=44100.0;
211
212         for (uint32_t i=0;i<sampleRate*duration;++i) {
213             v.push_back(M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate)));
214         }
215     }
216 }
```

```

215     }
216
217     void generateinversesawtoothwave(std::vector<float>& v, int duration, double frequen\
218 ncyInHz)
219     {
220         const double sampleRate=44100.0;
221         double period=sampleRate/frequencyInHz;
222         std::vector<float> temp;
223
224         for (uint32_t i=0;i<period;++i) {
225             temp.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
226         }
227         for (uint32_t i=0;i<sampleRate*duration;++i) {
228             v.push_back(temp[temp.size()-(i%temp.size())]);
229         }
230     }
231
232     void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz)
233     {
234         const double sampleRate=44100.0;
235
236         for (uint32_t i=0;i<sampleRate*duration;++i) {
237             v.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
238         }
239     }
240
241     void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz)
242     {
243         const double sampleRate=44100.0;
244         double period=sampleRate/frequencyInHz;
245         double dutyCycle=period*0.5;
246         double ss=0.0;
247
248         for (uint32_t i=0;i<sampleRate*duration;++i) {
249             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
250             {
251                 ss=0.7;
252             } else {
253                 ss=-0.7;
254             }
255             v.push_back(ss);
256         }
257         v[0]=0.0;

```

```
258         v[v.size()-1]=0.0;
259     }
260
261     void generatethirtyfivesquarewave(std::vector<float>& v, int duration, double frequ\
262 encyInHz)
263     {
264         const double sampleRate=44100.0;
265         double period=sampleRate/frequencyInHz;
266         double dutyCycle=period*0.35;
267         double ss;
268
269         for (uint32_t i=0;i<sampleRate*duration;++i) {
270             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
271             {
272                 ss=0.7;
273             } else {
274                 ss=-0.7;
275             }
276             v.push_back(ss);
277         }
278     }
279
280     void generatetwentyfivesquarewave(std::vector<float>& v, int duration, double frequ\
281 encyInHz)
282     {
283         const double sampleRate=44100.0;
284         double period=sampleRate/frequencyInHz;
285         double dutyCycle=period*0.25;
286         double ss;
287
288         for (uint32_t i=0;i<sampleRate*duration;++i) {
289             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
290             {
291                 ss=0.7;
292             } else {
293                 ss=-0.7;
294             }
295             v.push_back(ss);
296         }
297     }
298
299     void generatenoise(std::vector<float>& v, int duration)
300     {
```

```

301         const double sampleRate=44100.0;
302         std::random_device rd;
303         std::mt19937 gen(rd());
304         std::uniform_real_distribution<> dis(-1.0, 1.0);
305
306         for (uint32_t i=0;i<sampleRate*duration;++i) {
307             v.push_back(dis(gen));
308         }
309     }
310
311     void generatelfo(std::vector<float>& v, int osc, int duration, double frequencyInHz)
312     {
313         if (frequencyInHz >= 0.0 && frequencyInHz <= 20.0) {
314             switch(osc) {
315                 case 0: generatetrianglewave(v,duration,frequencyInHz);
316                     break;
317                 case 1: generateinversesawtoothwave(v,duration,frequencyInHz);
318                     break;
319                 case 2: generatesawtoothwave(v,duration,frequencyInHz);
320                     break;
321                 case 3: generatesquarewave(v,duration,frequencyInHz);
322                     break;
323                 case 4: generatethirtyfivesquarewave(v,duration,frequencyInHz);
324                     break;
325                 case 5: generatetwentyfivesquarewave(v,duration,frequencyInHz);
326                     break;
327             }
328         }
329     }
330
331     void generateoscillator(std::vector<float>& v, int osc, int duration, double frequen\
332     ncyInHz)
333     {
334         if (frequencyInHz >= 20.0 && frequencyInHz <= 20000.0) {
335             switch(osc) {
336                 case 0: generatetrianglewave(v,duration,frequencyInHz);
337                     break;
338                 case 1: generateinversesawtoothwave(v,duration,frequencyInHz);
339                     break;
340                 case 2: generatesawtoothwave(v,duration,frequencyInHz);
341                     break;
342                 case 3: generatesquarewave(v,duration,frequencyInHz);
343                     break;

```

```

344         case 4: generatethirtyfivesquarewave(v,duration,frequencyInHz);
345             break;
346         case 5: generatetwentyfivesquarewave(v,duration,frequencyInHz);
347             break;
348         case 6: generatenoise(v,duration);
349             break;
350     }
351 }
352 }
353
354 void saveaudiofile(std::vector<float>& v, std::string filename)
355 {
356     const std::string path="subtractive/";
357     // Setup the audio file
358     AudioFile<float> a;
359     a.setNumChannels(1);
360     a.setBitDepth(24);
361     a.setNumSamplesPerChannel(44100);
362
363     for (int i=0;i<a.getNumSamplesPerChannel();++i)
364     {
365         for (int channel=0;channel<a.getNumChannels();++channel)
366         {
367             a.samples[channel][i]=v[i];
368         }
369     }
370     a.save(path+filename,AudioFileFormat::Aiff);
371 }
372 }
373
374 namespace Render
375 {
376     void fillbackground(png::image<png::rgb_pixel>& image)
377     {
378         png::rgb_pixel px(0x04,0x13,0x31);
379         for (uint32_t y=0;y<image.get_height();y++) {
380             for (uint32_t x=0;x<image.get_width();++x) {
381                 image.set_pixel(x,y,px);
382             }
383         }
384     }
385
386     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)

```

```

387     {
388         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
389     ))
390         {
391             png::rgb_pixel px(0x7a,0xb1,0xe3);
392             image.set_pixel(x,y,px);
393         }
394     }
395
396 void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
397 {
398     int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
399     dx = x2 - x1; dy = y2 - y1;
400     if (dx == 0)
401     {
402         if (y2 < y1) std::swap(y1, y2);
403         for (y = y1; y <= y2; y++)
404             drawpx(image, x1, y);
405         return;
406     }
407     if (dy == 0)
408     {
409         if (x2 < x1) std::swap(x1, x2);
410         for (x = x1; x <= x2; x++)
411             drawpx(image, x, y1);
412         return;
413     }
414     dx1 = abs(dx); dy1 = abs(dy);
415     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
416     if (dy1 <= dx1)
417     {
418         if (dx >= 0)
419         {
420             x = x1; y = y1; xe = x2;
421         }
422         else
423         {
424             x = x2; y = y2; xe = x1;
425         }
426         drawpx(image, x, y);
427         for (i = 0; x<xe; i++)
428         {
429             x = x + 1;

```

```

430         if (px<0)
431             px = px + 2 * dy1;
432         else
433         {
434             if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y - 1;
435             px = px + 2 * (dy1 - dx1);
436         }
437         drawpx(image, x, y);
438     }
439 }
440 else
441 {
442     if (dy >= 0)
443     {
444         x = x1; y = y1; ye = y2;
445     }
446     else
447     {
448         x = x2; y = y2; ye = y1;
449     }
450     drawpx(image, x, y);
451     for (i = 0; y<ye; i++)
452     {
453         y = y + 1;
454         if (py <= 0)
455             py = py + 2 * dx1;
456         else
457         {
458             if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x - 1;
459             py = py + 2 * (dx1 - dy1);
460         }
461         drawpx(image, x, y);
462     }
463 }
464 }
465
466 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
467 {
468     uint32_t y=0, ox=0, oy=0;
469     for (uint32_t x=0; x<image.get_width(); ++x)
470     {
471         y=signalY[x];
472         if (x == 0) {ox=x;oy=y;}

```



```
473         drawline(image,x,y,ox,oy);
474         ox=x;oy=y;
475     }
476 }
477
478 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
479 :vector<uint32_t>& v2)
480 {
481     uint32_t halfHeight=image.get_height()/2;
482     double value=0.0;
483     if (v2.size() == 0 || v2.size() > v1.size()) {
484         v2.resize(v1.size());
485     }
486
487     for (uint32_t i=0;i<v1.size();++i)
488     {
489         value=v1[i];
490         if (value >= 0.0) {
491             v2[i]=halfHeight-(halfHeight*value);
492         } else if (value < 0.0)
493         {
494             v2[i]=halfHeight+(halfHeight*fabs(value));
495         }
496     }
497 }
498
499 void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
500 {
501     fillbackground(image);
502     drawwave(image,v);
503 }
504
505 void saveimagefile(std::vector<uint32_t>& v, std::string filename)
506 {
507     const std::string path="subtractive/";
508     png::image<png::rgb_pixel> image(44100,600);
509     renderimage(image,v);
510     image.write(path+filename);
511 }
512
513 void saveimagefile(std::vector<float>& v2, std::string filename)
514 {
515     const std::string path="subtractive/";
```

```
516         png::image<png::rgb_pixel> image(44100,600);
517         std::vector<uint32_t> v;
518         normalizedtoimg(image,v2,v);
519         renderimage(image,v);
520         image.write(path+filename);
521     }
522
523     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
524 v1,std::vector<uint32_t>& v2)
525     {
526         uint32_t height=image.get_height();
527         if (v2.size() == 0 || v2.size() > v1.size()) {
528             v2.resize(v1.size());
529         }
530         for (uint32_t i=0;i<v1.size();++i)
531         {
532             v2[i]=height-(height*v1[i]);
533         }
534     }
535
536     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
537     {
538         const std::string path="subtractive/";
539         png::image<png::rgb_pixel> image(44100,600);
540         renderimage(image,v);
541         image.write(path+filename);
542     }
543
544     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
545     {
546         const std::string path="subtractive/";
547         png::image<png::rgb_pixel> image(44100,600);
548         std::vector<uint32_t> v;
549         normalizedenvelopetoimg(image,v2,v);
550         renderimage(image,v);
551         image.write(path+filename);
552     }
553
554 }
```

---

**03-formant.cpp - 16411 bytes.**

```
1 // compile: clang++ -std=c++20 -lpng 03-formant.cpp -o 03-formant
2
3 #define _USE_MATH_DEFINES
4 #include <cmath>
5 #include <vector>
6 #include <random>
7 #include <filesystem>
8 #include "indicators.hpp"
9 #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "Envelope.hpp"
12
13 namespace Render
14 {
15     void fillbackground(png::image<png::rgb_pixel>& image);
16     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
17     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
18     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
19     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
20 :vector<uint32_t>& v2);
21     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
22     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
23     void saveimagefile(std::vector<float>& v2, std::string filename);
24     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
25 v1, std::vector<uint32_t>& v2);
26     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
27     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
28 }
29
30 namespace SignalGenerators
31 {
32     void gain(std::vector<float>& v, double gain);
33     void normalize(std::vector<float>& v);
34     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
35     void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz);
36     void generatenoise(std::vector<float>& v, int duration);
37     void generateformant(std::vector<float>& v, int formantnumber, int duration);
38     void saveaudiofile(std::vector<float>& v, std::string filename);
39 }
40
41 int main()
42 {
```

```
43     namespace fs = std::filesystem;
44     fs::create_directory("formant");
45
46     const int duration=1;
47     const double sampleRate=44100.0;
48     std::vector<float> formant0;
49     std::vector<float> formant1;
50     std::vector<float> formant2;
51     std::vector<float> formant3;
52     std::vector<float> formant4;
53     std::vector<float> formant5;
54     std::vector<float> formant6;
55     std::vector<float> formant7;
56     std::vector<float> formant8;
57     std::vector<float> formant9;
58     std::vector<float> formant10;
59     std::vector<float> envelope;
60
61     using namespace indicators;
62     // Hide cursor
63     show_console_cursor(false);
64
65     // Setup ProgressBar
66     ProgressBar bar{
67         option::BarWidth{50},
68         option::Start{"["},
69         option::Fill{"█"},
70         option::Lead{"█"},
71         option::Remainder{"-"},
72         option::End{" ]"},
73         option::PostfixText{"Generate Formant0 1/23"},
74         option::ForegroundColor{Color::cyan},
75         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
76     };
77
78     // Update progress
79     bar.set_progress(0);
80
81     // Vowel [i]
82     SignalGenerators::generateformant(formant0,0,duration);
83     SignalGenerators::saveaudiofile(formant0,"01-formant0-vowel-i.aiff");
84     Render::saveimagefile(formant0,"02-formant0-vowel-i.png");
85
```

```
86     // Update progress
87     bar.set_progress(4);
88     bar.set_option(option::PostfixText{"Generate Formant1 2/23"});
89
90     // Vowel [ɔ]
91     SignalGenerators::generateformant(formant1,1,duration);
92     SignalGenerators::saveaudiofile(formant1,"03-formant1-vowel-ɔ.aiff");
93     Render::saveimagefile(formant0,"04-formant1-vowel-ɔ.png");
94
95     // Update progress
96     bar.set_progress(9);
97     bar.set_option(option::PostfixText{"Generate Formant2 3/23"});
98
99     // Vowel [e]
100    SignalGenerators::generateformant(formant2,2,duration);
101    SignalGenerators::saveaudiofile(formant2,"05-formant2-vowel-e.aiff");
102    Render::saveimagefile(formant0,"06-formant2-vowel-e.png");
103
104    // Update progress
105    bar.set_progress(13);
106    bar.set_option(option::PostfixText{"Generate Formant3 4/23"});
107
108    // Vowel [ɔ]
109    SignalGenerators::generateformant(formant3,3,duration);
110    SignalGenerators::saveaudiofile(formant3,"07-formant3-vowel-ɔ.aiff");
111    Render::saveimagefile(formant0,"08-formant3-vowel-ɔ.png");
112
113    // Update progress
114    bar.set_progress(17);
115    bar.set_option(option::PostfixText{"Generate Formant4 5/23"});
116
117    // Vowel [æ]
118    SignalGenerators::generateformant(formant4,4,duration);
119    SignalGenerators::saveaudiofile(formant4,"09-formant4-vowel-æ.aiff");
120    Render::saveimagefile(formant0,"10-formant4-vowel-æ.png");
121
122    // Update progress
123    bar.set_progress(22);
124    bar.set_option(option::PostfixText{"Generate Formant5 6/23"});
125
126    // Vowel [ɔ]
127    SignalGenerators::generateformant(formant5,5,duration);
128    SignalGenerators::saveaudiofile(formant5,"11-formant5-vowel-ɔ.aiff");
```

```
129     Render::saveimagefile(formant0, "12-formant5-vowel-□.png");
130
131     // Update progress
132     bar.set_progress(26);
133     bar.set_option(option::PostfixText{"Generate Formant6 7/23"});
134
135     // Vowel [□]
136     SignalGenerators::generateformant(formant6,6,duration);
137     SignalGenerators::saveaudiofile(formant6, "13-formant6-vowel-□.aiff");
138     Render::saveimagefile(formant0, "14-formant6-vowel-□.png");
139
140     // Update progress
141     bar.set_progress(30);
142     bar.set_option(option::PostfixText{"Generate Formant7 8/23"});
143
144     // Vowel [o]
145     SignalGenerators::generateformant(formant7,7,duration);
146     SignalGenerators::saveaudiofile(formant7, "15-formant7-vowel-o.aiff");
147     Render::saveimagefile(formant0, "16-formant7-vowel-o.png");
148
149     // Update progress
150     bar.set_progress(35);
151     bar.set_option(option::PostfixText{"Generate Formant8 9/23"});
152
153     // Vowel [□]
154     SignalGenerators::generateformant(formant8,8,duration);
155     SignalGenerators::saveaudiofile(formant8, "17-formant8-vowel-□.aiff");
156     Render::saveimagefile(formant0, "18-formant8-vowel-□.png");
157
158     // Update progress
159     bar.set_progress(39);
160     bar.set_option(option::PostfixText{"Generate Formant9 10/23"});
161
162     // Vowel [u]
163     SignalGenerators::generateformant(formant9,9,duration);
164     SignalGenerators::saveaudiofile(formant9, "19-formant9-vowel-u.aiff");
165     Render::saveimagefile(formant0, "20-formant9-vowel-u.png");
166
167     // Update progress
168     bar.set_progress(43);
169     bar.set_option(option::PostfixText{"Generate Formant10 11/23"});
170
171     // Vowel [□]
```

```
172     SignalGenerators::generateformant(formant10,10,duration);
173     SignalGenerators::saveaudiofile(formant10,"21-formant10-vowel-□.aiff");
174     Render::saveimagefile(formant0,"22-formant10-vowel-□.png");
175
176     // Update progress
177     bar.set_progress(48);
178     bar.set_option(option::PostfixText{"Generate Envelope 12/23"});
179
180     ADSR::Envelope env(sampleRate,duration);
181     env.generateenvelope(envelope);
182     Render::saveenvelopeimage(envelope,"23-envelope.png");
183
184     // Update progress
185     bar.set_progress(52);
186     bar.set_option(option::PostfixText{"Apply Envelope to Formant0 13/23"});
187
188     env.applyenvelope(formant0,envelope);
189     SignalGenerators::normalize(formant0);
190     SignalGenerators::saveaudiofile(formant0,"24-env-formant0-vowel-i.aiff");
191     Render::saveimagefile(formant0,"25-env-formant0-vowel-i.png");
192
193     // Update progress
194     bar.set_progress(57);
195     bar.set_option(option::PostfixText{"Apply Envelope to Formant1 14/23"});
196
197     env.applyenvelope(formant1,envelope);
198     SignalGenerators::normalize(formant1);
199     SignalGenerators::saveaudiofile(formant1,"26-env-formant1-vowel-□.aiff");
200     Render::saveimagefile(formant1,"27-env-formant1-vowel-□.png");
201
202     // Update progress
203     bar.set_progress(61);
204     bar.set_option(option::PostfixText{"Apply Envelope to Formant2 15/23"});
205
206     env.applyenvelope(formant2,envelope);
207     SignalGenerators::normalize(formant2);
208     SignalGenerators::saveaudiofile(formant2,"28-env-formant2-vowel-e.aiff");
209     Render::saveimagefile(formant2,"29-env-formant2-vowel-e.png");
210
211     // Update progress
212     bar.set_progress(65);
213     bar.set_option(option::PostfixText{"Apply Envelope to Formant3 16/23"});
214
```

```
215     env.applyenvelope(formant3, envelope);
216     SignalGenerators::normalize(formant3);
217     SignalGenerators::saveaudiofile(formant3, "30-env-formant3-vowel-□.aiff");
218     Render::saveimagefile(formant3, "31-env-formant3-vowel-□.png");
219
220     // Update progress
221     bar.set_progress(70);
222     bar.set_option(option::PostfixText{"Apply Envelope to Formant4 17/23"});
223
224     env.applyenvelope(formant4, envelope);
225     SignalGenerators::normalize(formant4);
226     SignalGenerators::saveaudiofile(formant4, "32-env-formant4-vowel-æ.aiff");
227     Render::saveimagefile(formant4, "33-env-formant4-vowel-æ.png");
228
229     // Update progress
230     bar.set_progress(74);
231     bar.set_option(option::PostfixText{"Apply Envelope to Formant5 18/23"});
232
233     env.applyenvelope(formant5, envelope);
234     SignalGenerators::normalize(formant5);
235     SignalGenerators::saveaudiofile(formant5, "34-env-formant5-vowel-□.aiff");
236     Render::saveimagefile(formant5, "35-env-formant5-vowel-□.png");
237
238     // Update progress
239     bar.set_progress(78);
240     bar.set_option(option::PostfixText{"Apply Envelope to Formant6 19/23"});
241
242     env.applyenvelope(formant6, envelope);
243     SignalGenerators::normalize(formant6);
244     SignalGenerators::saveaudiofile(formant6, "36-env-formant6-vowel-□.aiff");
245     Render::saveimagefile(formant6, "37-env-formant6-vowel-□.png");
246
247     // Update progress
248     bar.set_progress(83);
249     bar.set_option(option::PostfixText{"Apply Envelope to Formant7 20/23"});
250
251     env.applyenvelope(formant7, envelope);
252     SignalGenerators::normalize(formant7);
253     SignalGenerators::saveaudiofile(formant7, "38-env-formant7-vowel-o.aiff");
254     Render::saveimagefile(formant7, "39-env-formant7-vowel-o.png");
255
256     // Update progress
257     bar.set_progress(87);
```



```
258     bar.set_option(option::PostfixText{"Apply Envelope to Formant8 21/23"});
259
260     env.applyenvelope(formant8, envelope);
261     SignalGenerators::normalize(formant8);
262     SignalGenerators::saveaudiofile(formant8, "40-env-formant8-vowel-□.aiff");
263     Render::saveimagefile(formant8, "41-env-formant8-vowel-□.png");
264
265     // Update progress
266     bar.set_progress(91);
267     bar.set_option(option::PostfixText{"Apply Envelope to Formant9 22/23"});
268
269     env.applyenvelope(formant9, envelope);
270     SignalGenerators::normalize(formant9);
271     SignalGenerators::saveaudiofile(formant9, "42-env-formant9-vowel-u.aiff");
272     Render::saveimagefile(formant9, "43-env-formant9-vowel-u.png");
273
274     // Update progress
275     bar.set_progress(96);
276     bar.set_option(option::PostfixText{"Apply Envelope to Formant10 23/23"});
277
278     env.applyenvelope(formant10, envelope);
279     SignalGenerators::normalize(formant10);
280     SignalGenerators::saveaudiofile(formant10, "44-env-formant10-vowel-□.aiff");
281     Render::saveimagefile(formant10, "45-env-formant10-vowel-□.png");
282
283     // Update progress
284     bar.set_progress(100);
285     bar.set_option(option::PostfixText{"Done 23/23"});
286
287     // Show cursor
288     show_console_cursor(true);
289     return 0;
290 }
291
292 namespace SignalGenerators
293 {
294     void gain(std::vector<float>& v, double gain)
295     {
296         for (uint32_t i=0; i<v.size(); ++i) {
297             v[i]=v[i]*gain;
298         }
299     }
300 }
```

```

301     void normalize(std::vector<float>& v)
302     {
303         float max=0.0,value=0.0;
304         for (uint32_t i=0;i<v.size();++i) {
305             value=v[i];
306             if (value > max) {max=value;}
307         }
308         // max=std::ceil(max);
309         for (uint32_t i=0;i<v.size();++i) {
310             // v[i]=v[i]/max;
311             v[i]=(v[i]/max)*0.707;
312         }
313     }
314
315     void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
316     {
317         for (uint32_t i=0;i<v1.size();++i)
318         {
319             v3[i]=v1[i]+v2[i];
320         }
321     }
322
323     void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz)
324     {
325         const double sampleRate=44100.0;
326         for (uint32_t i=0;i<sampleRate*duration;++i) {
327             v.push_back(sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2.0
328 PI));
329         }
330     }
331
332     void generatenoise(std::vector<float>& v, int duration)
333     {
334         const double sampleRate=44100.0;
335         std::random_device rd;
336         std::mt19937 gen(rd());
337         std::uniform_real_distribution<> dis(-1.0, 1.0);
338
339         for (uint32_t i=0;i<sampleRate*duration;++i) {
340             v.push_back(dis(gen));
341         }
342     }
343

```

```

344     void generateformant(std::vector<float>& mixer, int formantnumber, int duration)
345     {
346         // Vowel      [i]      [ɔ]      [e]      [ɒ]      [æ]      [ɑ]
347         // F1          280      370      405      600      860      830
348         // F2          2230     2090     2080     1930     1550     1300
349         const double sampleRate=44100.0;
350         float f1[]={280.0,370.0,405.0,600.0,860.0,830.0,560.0,430.0,400.0,330.0,680.0};
351         float f2[]={2230.0,2090.0,2080.0,1930.0,1550.0,1170.0,820.0,980.0,1100.0,1260.0,1300.0};
352     };
353
354     mixer.resize(duration*sampleRate);
355     std::vector<float> osc1;
356     std::vector<float> osc2;
357     std::vector<float> osc3;
358     generatesinewave(osc1,duration,f1[formantnumber]);
359     generatesinewave(osc2,duration,f2[formantnumber]);
360     generatenoise(osc3,duration);
361     gain(osc3,0.2);
362     addwaves(osc1,osc2,mixer);
363     addwaves(mixer,osc3,mixer);
364     normalize(mixer);
365
366     void saveaudiofile(std::vector<float>& v, std::string filename)
367     {
368         const std::string path="formant/";
369         // Setup the audio file
370         AudioFile<float> a;
371         a.setNumChannels(1);
372         a.setBitDepth(24);
373         a.setNumSamplesPerChannel(44100);
374
375         for (int i=0;i<a.getNumSamplesPerChannel();++i)
376         {
377             for (int channel=0;channel<a.getNumChannels();++channel)
378             {
379                 a.samples[channel][i]=v[i];
380             }
381         }
382         a.save(path+filename,AudioFileFormat::Aiff);
383     }
384 }
385
386 namespace Render

```

```

387 {
388     void fillbackground(png::image<png::rgb_pixel>& image)
389     {
390         png::rgb_pixel px(0x04,0x13,0x31);
391         for (uint32_t y=0;y<image.get_height();y++) {
392             for (uint32_t x=0;x<image.get_width();++x) {
393                 image.set_pixel(x,y,px);
394             }
395         }
396     }
397
398     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
399     {
400         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
401     ))
402         {
403             png::rgb_pixel px(0x7a,0xb1,0xe3);
404             image.set_pixel(x,y,px);
405         }
406     }
407
408     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
409     {
410         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
411         dx = x2 - x1; dy = y2 - y1;
412         if (dx == 0)
413         {
414             if (y2 < y1) std::swap(y1, y2);
415             for (y = y1; y <= y2; y++)
416                 drawpx(image, x1, y);
417             return;
418         }
419         if (dy == 0)
420         {
421             if (x2 < x1) std::swap(x1, x2);
422             for (x = x1; x <= x2; x++)
423                 drawpx(image, x, y1);
424             return;
425         }
426         dx1 = abs(dx); dy1 = abs(dy);
427         px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
428         if (dy1 <= dx1)
429         {

```

```

430         if (dx >= 0)
431         {
432             x = x1; y = y1; xe = x2;
433         }
434         else
435         {
436             x = x2; y = y2; xe = x1;
437         }
438         drawpx(image, x, y);
439         for (i = 0; x<xe; i++)
440         {
441             x = x + 1;
442             if (px<0)
443                 px = px + 2 * dy1;
444             else
445             {
446                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y;
447                 px = px + 2 * (dy1 - dx1);
448             }
449             drawpx(image, x, y);
450         }
451     }
452     else
453     {
454         if (dy >= 0)
455         {
456             x = x1; y = y1; ye = y2;
457         }
458         else
459         {
460             x = x2; y = y2; ye = y1;
461         }
462         drawpx(image, x, y);
463         for (i = 0; y<ye; i++)
464         {
465             y = y + 1;
466             if (py <= 0)
467                 py = py + 2 * dx1;
468             else
469             {
470                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x;
471                 py = py + 2 * (dx1 - dy1);
472             }

```

```
473             drawpx(image, x, y);
474         }
475     }
476 }
477
478 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
479 {
480     uint32_t y=0, ox=0, oy=0;
481     for (uint32_t x=0; x<image.get_width(); ++x)
482     {
483         y=signalY[x];
484         if (x == 0) {ox=x; oy=y;}
485         drawline(image, x, y, ox, oy);
486         ox=x; oy=y;
487     }
488 }
489
490 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
491 :vector<uint32_t>& v2)
492 {
493     uint32_t halfHeight=image.get_height()/2;
494     double value=0.0;
495     if (v2.size() == 0 || v2.size() > v1.size()) {
496         v2.resize(v1.size());
497     }
498
499     for (uint32_t i=0; i<v1.size(); ++i)
500     {
501         value=v1[i];
502         if (value >= 0.0) {
503             v2[i]=halfHeight-(halfHeight*value);
504         } else if (value < 0.0)
505         {
506             v2[i]=halfHeight+(halfHeight*fabs(value));
507         }
508     }
509 }
510
511 void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
512 {
513     fillbackground(image);
514     drawwave(image, v);
515 }
```

```
516
517     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
518     {
519         const std::string path="formant/";
520         png::image<png::rgb_pixel> image(44100,600);
521         renderimage(image,v);
522         image.write(path+filename);
523     }
524
525     void saveimagefile(std::vector<float>& v2, std::string filename)
526     {
527         const std::string path="formant/";
528         png::image<png::rgb_pixel> image(44100,600);
529         std::vector<uint32_t> v;
530         normalizedtoimg(image,v2,v);
531         renderimage(image,v);
532         image.write(path+filename);
533     }
534
535     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
536 v1,std::vector<uint32_t>& v2)
537     {
538         uint32_t height=image.get_height();
539         if (v2.size() == 0 || v2.size() > v1.size()) {
540             v2.resize(v1.size());
541         }
542         for (uint32_t i=0;i<v1.size();++i)
543         {
544             v2[i]=height-(height*v1[i]);
545         }
546     }
547
548     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
549     {
550         const std::string path="formant/";
551         png::image<png::rgb_pixel> image(44100,600);
552         renderimage(image,v);
553         image.write(path+filename);
554     }
555
556     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
557     {
558         const std::string path="formant/";
```

```

559         png::image<png::rgb_pixel> image(44100,600);
560         std::vector<uint32_t> v;
561         normalizedenvelopetoimg(image,v2,v);
562         renderimage(image,v);
563         image.write(path+filename);
564     }
565
566 }

```

---

#### 04-granular.cpp - 9996 bytes.

```

1  // compile: clang++ -std=c++20 -lpng 04-granular.cpp -o 04-granular
2  #define _USE_MATH_DEFINES
3  #include <cmath>
4  #include <vector>
5  #include <random>
6  #include <filesystem>
7  #include "indicators.hpp"
8  #include <png++/png.hpp>
9  #include "AudioFile/AudioFile.h"
10 #include "Envelope.hpp"
11
12 namespace Render
13 {
14     void fillbackground(png::image<png::rgb_pixel>& image);
15     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
16     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
17     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
18     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
19 :vector<uint32_t>& v2);
20     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
21     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
22     void saveimagefile(std::vector<float>& v2, std::string filename);
23     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
24 v1, std::vector<uint32_t>& v2);
25     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
26     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
27 }
28
29 namespace SignalGenerators
30 {
31     void gain(std::vector<float>& v, double gain);
32     void normalize(std::vector<float>& v);

```



```

33     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
34     void generatesample(std::vector<float>& v, int duration);
35     void extendsample(std::vector<float>& v1, std::vector<float>& v2, int n);
36     void saveaudiofile(std::vector<float>& v, std::string filename, int duration);
37     void loadaudiofile(std::vector<float>& v, std::string filename);
38 }
39
40 int main()
41 {
42     namespace fs = std::filesystem;
43     fs::create_directory("granular");
44
45     using namespace indicators;
46     // Hide cursor
47     show_console_cursor(false);
48
49     // Setup ProgressBar
50     ProgressBar bar{
51         option::BarWidth{50},
52         option::Start{"["},
53         option::Fill{"█"},
54         option::Lead{"█"},
55         option::Remainder{"-"},
56         option::End{" ]"},
57         option::PostfixText{"Generate Sample 1/5"},
58         option::ForegroundColor{Color::cyan},
59         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
60     };
61
62     // Update progress
63     bar.set_progress(0);
64
65     const int duration=1;
66     std::vector<float> sample;
67     std::vector<float> sample_extended;
68     SignalGenerators::generatesample(sample, duration);
69     SignalGenerators::saveaudiofile(sample, "01-generatedsample.aiff", 1);
70     Render::saveimagefile(sample, "02-generatedsample.png");
71
72     // Update progress
73     bar.set_progress(20);
74     bar.set_option(option::PostfixText{"Extend Sample 2/5"});
75

```

```

76     SignalGenerators::extendsample(sample,sample_extended,4);
77     SignalGenerators::saveaudiofile(sample_extended,"03-extendedsample.aiff",4);
78
79     // Update progress
80     bar.set_progress(40);
81     bar.set_option(option::PostfixText{"Get grains from sample 3/5"});
82
83     // Get grains from sample.
84     // Add grains to sample_extended.
85     float grains[100][44];
86     uint32_t ptr1=0,ptr2=0;
87     std::random_device rd;
88     std::mt19937 gen(rd());
89     std::uniform_int_distribution<> distria(1, 990);
90     std::uniform_int_distribution<> distrib(1, 4000);
91     std::uniform_int_distribution<> distric(0, 99);
92
93     for (uint32_t i=0;i<100;++i) {
94         ptr1=distria(gen)*44;
95         for (uint32_t j=0;j<44;++j) {
96             grains[i][j]=sample[ptr1++];
97         }
98     }
99
100    // Update progress
101    bar.set_progress(60);
102    bar.set_option(option::PostfixText{"Add grains to sample_extended 4/5"});
103
104    for (uint32_t i=0;i<800;++i) {
105        ptr1=distric(gen);
106        ptr2=distrib(gen)*44;
107        for (uint32_t j=0;j<44;++j) {
108            sample_extended[ptr2++]=grains[ptr1][j];
109        }
110    }
111
112    // Update progress
113    bar.set_progress(80);
114    bar.set_option(option::PostfixText{"Final Granular 5/5"});
115
116    SignalGenerators::normalize(sample_extended);
117    SignalGenerators::saveaudiofile(sample_extended,"04-finalgranular.aiff",4);
118

```

```

119     // Update progress
120     bar.set_progress(100);
121     bar.set_option(option::PostfixText{"Done 5/5"});
122
123     // Show cursor
124     show_console_cursor(true);
125     return 0;
126 }
127
128 namespace SignalGenerators
129 {
130     void gain(std::vector<float>& v, double gain)
131     {
132         for (uint32_t i=0; i<v.size(); ++i) {
133             v[i]=v[i]*gain;
134         }
135     }
136
137     void normalize(std::vector<float>& v)
138     {
139         float max=0.0, value=0.0;
140         for (uint32_t i=0; i<v.size(); ++i) {
141             value=v[i];
142             if (value > max) {max=value;}
143         }
144         // max=std::ceil(max);
145         for (uint32_t i=0; i<v.size(); ++i) {
146             // v[i]=v[i]/max;
147             v[i]=(v[i]/max)*0.707;
148         }
149     }
150
151     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3)
152     {
153         for (uint32_t i=0; i<v1.size(); ++i)
154         {
155             v3[i]=v1[i]+v2[i];
156         }
157     }
158
159     void generatesample(std::vector<float>& v, int duration)
160     {
161         const double sampleRate=44100.0;

```

```

162         std::vector<float> envelope1;
163         std::vector<float> envelope2;
164         std::vector<float> sample1;
165         std::vector<float> sample2;
166
167         ADSR::Envelope env(sampleRate,duration);
168         env.generateenvelope2(envelope1);
169         env.generateenvelope3(envelope2);
170
171         loadaudiofile(sample1,"additive/67-mixed.aiff");
172         loadaudiofile(sample2,"subtractive/08-postfilter.aiff");
173
174         env.applyenvelope(sample1,envelope2);
175         env.applyenvelope(sample2,envelope1);
176
177         v.resize(duration*sampleRate);
178         addwaves(sample1,sample2,v);
179         normalize(v);
180     }
181
182     void extendsample(std::vector<float>& v1, std::vector<float>& v2, int n)
183     {
184         for (uint32_t i=0;i<v1.size();++i) {
185             for (uint32_t j=0;j<n;++j) {
186                 v2.push_back(v1[i]);
187             }
188         }
189     }
190
191     void saveaudiofile(std::vector<float>& v, std::string filename, int duration)
192     {
193         const std::string path="granular/";
194         // Setup the audio file
195         AudioFile<float> a;
196         a.setNumChannels(1);
197         a.setBitDepth(24);
198         a.setNumSamplesPerChannel(44100*duration);
199
200         for (int i=0;i<a.getNumSamplesPerChannel();++i)
201         {
202             for (int channel=0;channel<a.getNumChannels();++channel)
203             {
204                 a.samples[channel][i]=v[i];

```

```

205         }
206     }
207     a.save(path+filename,AudioFileFormat::Aiff);
208 }
209
210 void loadaudiofile(std::vector<float>& v, std::string filename)
211 {
212     AudioFile<float> a;
213     bool loadedOK = a.load(filename);
214     if (loadedOK) {
215         for (int i=0;i<a.getNumSamplesPerChannel();++i)
216             {
217                 for (int channel=0;channel<1;++channel)
218                     {
219                         v.push_back(a.samples[channel][i]);
220                     }
221             }
222     }
223 }
224 }
225
226 namespace Render
227 {
228     void fillbackground(png::image<png::rgb_pixel>& image)
229     {
230         png::rgb_pixel px(0x04,0x13,0x31);
231         for (uint32_t y=0;y<image.get_height();y++) {
232             for (uint32_t x=0;x<image.get_width();++x) {
233                 image.set_pixel(x,y,px);
234             }
235         }
236     }
237
238     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
239     {
240         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
241     ))
242         {
243             png::rgb_pixel px(0x7a,0xb1,0xe3);
244             image.set_pixel(x,y,px);
245         }
246     }
247

```

```
248 void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
249 {
250     int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
251     dx = x2 - x1; dy = y2 - y1;
252     if (dx == 0)
253     {
254         if (y2 < y1) std::swap(y1, y2);
255         for (y = y1; y <= y2; y++)
256             drawpx(image, x1, y);
257         return;
258     }
259     if (dy == 0)
260     {
261         if (x2 < x1) std::swap(x1, x2);
262         for (x = x1; x <= x2; x++)
263             drawpx(image, x, y1);
264         return;
265     }
266     dx1 = abs(dx); dy1 = abs(dy);
267     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
268     if (dy1 <= dx1)
269     {
270         if (dx >= 0)
271         {
272             x = x1; y = y1; xe = x2;
273         }
274         else
275         {
276             x = x2; y = y2; xe = x1;
277         }
278         drawpx(image, x, y);
279         for (i = 0; x<xe; i++)
280         {
281             x = x + 1;
282             if (px<0)
283                 px = px + 2 * dy1;
284             else
285             {
286                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
287                 px = px + 2 * (dy1 - dx1);
288             }
289             drawpx(image, x, y);
290         }

```

```
291     }
292     else
293     {
294         if (dy >= 0)
295         {
296             x = x1; y = y1; ye = y2;
297         }
298         else
299         {
300             x = x2; y = y2; ye = y1;
301         }
302         drawpx(image, x, y);
303         for (i = 0; y<ye; i++)
304         {
305             y = y + 1;
306             if (py <= 0)
307                 py = py + 2 * dx1;
308             else
309             {
310                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x - 1;
311                 py = py + 2 * (dx1 - dy1);
312             }
313             drawpx(image, x, y);
314         }
315     }
316 }
317
318 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
319 {
320     uint32_t y=0, ox=0, oy=0;
321     for (uint32_t x=0; x<image.get_width(); ++x)
322     {
323         y=signalY[x];
324         if (x == 0) {ox=x;oy=y;}
325         drawline(image, x, y, ox, oy);
326         ox=x;oy=y;
327     }
328 }
329
330 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std::\
331 :vector<uint32_t>& v2)
332 {
333     uint32_t halfHeight=image.get_height()/2;
```

```
334         double value=0.0;
335         if (v2.size() == 0 || v2.size() > v1.size()) {
336             v2.resize(v1.size());
337         }
338
339         for (uint32_t i=0;i<v1.size();++i)
340         {
341             value=v1[i];
342             if (value >= 0.0) {
343                 v2[i]=halfHeight-(halfHeight*value);
344             } else if (value < 0.0)
345             {
346                 v2[i]=halfHeight+(halfHeight*fabs(value));
347             }
348         }
349     }
350
351     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
352     {
353         fillbackground(image);
354         drawwave(image,v);
355     }
356
357     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
358     {
359         const std::string path="granular/";
360         png::image<png::rgb_pixel> image(44100,600);
361         renderimage(image,v);
362         image.write(path+filename);
363     }
364
365     void saveimagefile(std::vector<float>& v2, std::string filename)
366     {
367         const std::string path="granular/";
368         png::image<png::rgb_pixel> image(44100,600);
369         std::vector<uint32_t> v;
370         normalizedtoimg(image,v2,v);
371         renderimage(image,v);
372         image.write(path+filename);
373     }
374
375     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
376     v1, std::vector<uint32_t>& v2)
```



```

377     {
378         uint32_t height=image.get_height();
379         if (v2.size() == 0 || v2.size() > v1.size()) {
380             v2.resize(v1.size());
381         }
382         for (uint32_t i=0;i<v1.size();++i)
383         {
384             v2[i]=height-(height*v1[i]);
385         }
386     }
387
388     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
389     {
390         const std::string path="granular/";
391         png::image<png::rgb_pixel> image(44100,600);
392         renderimage(image,v);
393         image.write(path+filename);
394     }
395
396     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
397     {
398         const std::string path="granular/";
399         png::image<png::rgb_pixel> image(44100,600);
400         std::vector<uint32_t> v;
401         normalizedenvelopetoimg(image,v2,v);
402         renderimage(image,v);
403         image.write(path+filename);
404     }
405
406 }

```

---

**05-fm.cpp - 8869 bytes.**

```

1  // compile: clang++ -std=c++20 -lpng 05-fm.cpp -o 05-fm
2  #define _USE_MATH_DEFINES
3  #include <cmath>
4  #include <vector>
5  #include <random>
6  #include <filesystem>
7  #include "indicators.hpp"
8  #include <png++/png.hpp>
9  #include "AudioFile/AudioFile.h"
10 #include "Envelope.hpp"

```

```

11
12 namespace Render
13 {
14     void fillbackground(png::image<png::rgb_pixel>& image);
15     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
16     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
17     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
18     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
19 :vector<uint32_t>& v2);
20     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
21     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
22     void saveimagefile(std::vector<float>& v2, std::string filename);
23     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
24 v1, std::vector<uint32_t>& v2);
25     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
26     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
27 }
28
29 namespace SignalGenerators
30 {
31     void gain(std::vector<float>& v, double gain);
32     void normalize(std::vector<float>& v);
33     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
34     void generatefrequencymodulation(std::vector<float>& v, int duration, float frequen\
35 cyInHz);
36     void saveaudiofile(std::vector<float>& v, std::string filename);
37 }
38
39 int main()
40 {
41     namespace fs = std::filesystem;
42     fs::create_directory("fm");
43
44     const int duration=1;
45     const double sampleRate=44100.0;
46     std::vector<float> fm;
47     std::vector<float> envelope;
48
49     using namespace indicators;
50     // Hide cursor
51     show_console_cursor(false);
52
53     // Setup ProgressBar

```

```

54     ProgressBar bar{
55         option::BarWidth{50},
56         option::Start{"["},
57         option::Fill{"█"},
58         option::Lead{"█"},
59         option::Remainder{"-"},
60         option::End{" ]"},
61         option::PostfixText{"Generate Frequency Modulation (FM) 1/2"},
62         option::ForegroundColor{Color::cyan},
63         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
64     };
65
66     // Update progress
67     bar.set_progress(0);
68
69     SignalGenerators::generatefrequencymodulation(fm,duration,440);
70     SignalGenerators::normalize(fm);
71     SignalGenerators::saveaudiofile(fm,"01-fm.aiff");
72     Render::saveimagefile(fm,"02-fm.png");
73
74     // Update progress
75     bar.set_progress(50);
76     bar.set_option(option::PostfixText{"Apply Envelope to FM 2/2"});
77
78     ADSR::Envelope env(sampleRate,duration);
79     env.generateenvelope(envelope);
80     Render::saveenvelopeimage(envelope,"03-envelope.png");
81
82     env.applyenvelope(fm,envelope);
83     SignalGenerators::normalize(fm);
84     SignalGenerators::saveaudiofile(fm,"04-final_fm.aiff");
85     Render::saveimagefile(fm,"05-final_fm.png");
86
87     // Update progress
88     bar.set_progress(100);
89     bar.set_option(option::PostfixText{"Done 1/2"});
90
91     // Show cursor
92     show_console_cursor(true);
93     return 0;
94 }
95
96 namespace SignalGenerators

```

```
97 {
98     void gain(std::vector<float>& v, double gain)
99     {
100         for (uint32_t i=0;i<v.size();++i) {
101             v[i]=v[i]*gain;
102         }
103     }
104
105     void normalize(std::vector<float>& v)
106     {
107         float max=0.0,value=0.0;
108         for (uint32_t i=0;i<v.size();++i) {
109             value=v[i];
110             if (value > max) {max=value;}
111         }
112         // max=std::ceil(max);
113         for (uint32_t i=0;i<v.size();++i) {
114             // v[i]=v[i]/max;
115             v[i]=(v[i]/max)*0.707;
116         }
117     }
118
119     void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
120     {
121         for (uint32_t i=0;i<v1.size();++i)
122         {
123             v3[i]=v1[i]+v2[i];
124         }
125     }
126
127     void generatefrequencymodulation(std::vector<float>& v, int duration, float frequen\
128 cyInHz)
129     {
130         const double twoPI=2*M_PI;
131         const double sampleRate=44100.0;
132         const double frequencyRadian = twoPI / sampleRate;
133         double modulatorFrequency = frequencyInHz * 3;
134         double modulatorIncrement = frequencyRadian * modulatorFrequency;
135         double modulatorPhase = 0;
136         double carrierIncrement = 0;
137         double carrierPhase = 0;
138         double modulatoramplitude = 2 * modulatorFrequency;
139         double modulatorValue = 0;
```

```
140
141     for (uint32_t i=0;i<sampleRate*duration;++i) {
142         v.push_back(sinf(carrierPhase));
143         modulatorValue = modulatoramplitude * sinf(modulatorPhase);
144         carrierIncrement = frequencyRadian * (frequencyInHz + modulatorValue);
145         carrierPhase = carrierPhase + carrierIncrement;
146         modulatorPhase = modulatorPhase + modulatorIncrement;
147         if (carrierPhase >= twoPI) {
148             carrierPhase -= twoPI;
149         }
150         else if (carrierPhase < 0) {
151             carrierPhase += twoPI;
152         }
153         if (modulatorPhase >= twoPI) {
154             modulatorPhase -= twoPI;
155         }
156     }
157 }
158
159 void saveaudiofile(std::vector<float>& v, std::string filename)
160 {
161     const std::string path="fm/";
162     // Setup the audio file
163     AudioFile<float> a;
164     a.setNumChannels(1);
165     a.setBitDepth(24);
166     a.setNumSamplesPerChannel(44100);
167
168     for (int i=0;i<a.getNumSamplesPerChannel();++i)
169     {
170         for (int channel=0;channel<a.getNumChannels();++channel)
171         {
172             a.samples[channel][i]=v[i];
173         }
174     }
175     a.save(path+filename,AudioFileFormat::Aiff);
176 }
177 }
178
179 namespace Render
180 {
181     void fillbackground(png::image<png::rgb_pixel>& image)
182     {
```

```

183         png::rgb_pixel px(0x04,0x13,0x31);
184         for (uint32_t y=0;y<image.get_height();y++) {
185             for (uint32_t x=0;x<image.get_width();++x) {
186                 image.set_pixel(x,y,px);
187             }
188         }
189     }
190
191     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
192     {
193         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
194     ))
195         {
196             png::rgb_pixel px(0x7a,0xb1,0xe3);
197             image.set_pixel(x,y,px);
198         }
199     }
200
201     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
202     {
203         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
204         dx = x2 - x1; dy = y2 - y1;
205         if (dx == 0)
206         {
207             if (y2 < y1) std::swap(y1, y2);
208             for (y = y1; y <= y2; y++)
209                 drawpx(image, x1, y);
210             return;
211         }
212         if (dy == 0)
213         {
214             if (x2 < x1) std::swap(x1, x2);
215             for (x = x1; x <= x2; x++)
216                 drawpx(image, x, y1);
217             return;
218         }
219         dx1 = abs(dx); dy1 = abs(dy);
220         px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
221         if (dy1 <= dx1)
222         {
223             if (dx >= 0)
224             {
225                 x = x1; y = y1; xe = x2;

```

```
226     }
227     else
228     {
229         x = x2; y = y2; xe = x1;
230     }
231     drawpx(image, x, y);
232     for (i = 0; x < xe; i++)
233     {
234         x = x + 1;
235         if (px < 0)
236             px = px + 2 * dy1;
237         else
238         {
239             if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) y = y + 1; else y = y - 1;
240             px = px + 2 * (dy1 - dx1);
241         }
242         drawpx(image, x, y);
243     }
244 }
245 else
246 {
247     if (dy >= 0)
248     {
249         x = x1; y = y1; ye = y2;
250     }
251     else
252     {
253         x = x2; y = y2; ye = y1;
254     }
255     drawpx(image, x, y);
256     for (i = 0; y < ye; i++)
257     {
258         y = y + 1;
259         if (py <= 0)
260             py = py + 2 * dx1;
261         else
262         {
263             if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) x = x + 1; else x = x - 1;
264             py = py + 2 * (dx1 - dy1);
265         }
266         drawpx(image, x, y);
267     }
268 }
```

```
269     }
270
271     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
272     {
273         uint32_t y=0, ox=0, oy=0;
274         for (uint32_t x=0; x<image.get_width(); ++x)
275         {
276             y=signalY[x];
277             if (x == 0) {ox=x; oy=y;}
278             drawline(image, x, y, ox, oy);
279             ox=x; oy=y;
280         }
281     }
282
283     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
284 :vector<uint32_t>& v2)
285     {
286         uint32_t halfHeight=image.get_height()/2;
287         double value=0.0;
288         if (v2.size() == 0 || v2.size() > v1.size()) {
289             v2.resize(v1.size());
290         }
291
292         for (uint32_t i=0; i<v1.size(); ++i)
293         {
294             value=v1[i];
295             if (value >= 0.0) {
296                 v2[i]=halfHeight-(halfHeight*value);
297             } else if (value < 0.0)
298             {
299                 v2[i]=halfHeight+(halfHeight*fabs(value));
300             }
301         }
302     }
303
304     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
305     {
306         fillbackground(image);
307         drawwave(image, v);
308     }
309
310     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
311     {
```



```
312         const std::string path="fm/";
313         png::image<png::rgb_pixel> image(44100,600);
314         renderimage(image,v);
315         image.write(path+filename);
316     }
317
318     void saveimagefile(std::vector<float>& v2, std::string filename)
319     {
320         const std::string path="fm/";
321         png::image<png::rgb_pixel> image(44100,600);
322         std::vector<uint32_t> v;
323         normalizedtoimg(image,v2,v);
324         renderimage(image,v);
325         image.write(path+filename);
326     }
327
328     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
329 v1,std::vector<uint32_t>& v2)
330     {
331         uint32_t height=image.get_height();
332         if (v2.size() == 0 || v2.size() > v1.size()) {
333             v2.resize(v1.size());
334         }
335         for (uint32_t i=0;i<v1.size();++i)
336         {
337             v2[i]=height-(height*v1[i]);
338         }
339     }
340
341     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
342     {
343         const std::string path="fm/";
344         png::image<png::rgb_pixel> image(44100,600);
345         renderimage(image,v);
346         image.write(path+filename);
347     }
348
349     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
350     {
351         const std::string path="fm/";
352         png::image<png::rgb_pixel> image(44100,600);
353         std::vector<uint32_t> v;
354         normalizedenvelopetoimg(image,v2,v);
```

```

355         renderimage(image,v);
356         image.write(path+filename);
357     }
358
359 }

```

---

#### 06-la.cpp - 9077 bytes.

---

```

1  // compile: clang++ -std=c++20 -lpng 06-la.cpp -o 06-la
2  #define _USE_MATH_DEFINES
3  #include <cmath>
4  #include <vector>
5  #include <random>
6  #include <filesystem>
7  #include "indicators.hpp"
8  #include <png++/png.hpp>
9  #include "AudioFile/AudioFile.h"
10 #include "Envelope.hpp"
11
12 namespace Render
13 {
14     void fillbackground(png::image<png::rgb_pixel>& image);
15     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
16     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
17     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
18     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std::\
19 :vector<uint32_t>& v2);
20     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
21     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
22     void saveimagefile(std::vector<float>& v2, std::string filename);
23     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
24 v1, std::vector<uint32_t>& v2);
25     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
26     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
27 }
28
29 namespace SignalGenerators
30 {
31     void gain(std::vector<float>& v, double gain);
32     void normalize(std::vector<float>& v);
33     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
34     void generatesubtractive(std::vector<float>& v);
35     void generatesample(std::vector<float>& v);

```

```
36     void saveaudiofile(std::vector<float>& v, std::string filename);
37 }
38
39 int main()
40 {
41     namespace fs = std::filesystem;
42     fs::create_directory("la");
43
44     const int duration=1;
45     const double sampleRate=44100.0;
46     std::vector<float> sample;
47     std::vector<float> subtractive;
48     std::vector<float> envelope1;
49     std::vector<float> envelope2;
50     std::vector<float> envelope;
51     std::vector<float> lineararithmetic;
52
53     using namespace indicators;
54     // Hide cursor
55     show_console_cursor(false);
56
57     // Setup ProgressBar
58     ProgressBar bar{
59         option::BarWidth{50},
60         option::Start{"["},
61         option::Fill{"█"},
62         option::Lead{"█"},
63         option::Remainder{"-"},
64         option::End{" ]"},
65         option::PostfixText{"Setup Linear Arithmetic 1/2"},
66         option::ForegroundColor{Color::cyan},
67         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
68     };
69
70     // Update progress
71     bar.set_progress(0);
72
73     SignalGenerators::generatesample(sample);
74     SignalGenerators::normalize(sample);
75
76     SignalGenerators::generatesubtractive(subtractive);
77     SignalGenerators::normalize(subtractive);
78
```

```

79     ADSR::Envelope env(sampleRate,duration);
80     env.generateenvelope(envelope);
81     env.generateenvelope3(envelope1);
82     env.generateenvelope2(envelope2);
83
84     env.applyenvelope(sample,envelope1);
85     SignalGenerators::normalize(sample);
86     env.applyenvelope(subtractive,envelope2);
87     SignalGenerators::normalize(sample);
88     lineararithmetic.resize(sample.size());
89     SignalGenerators::addwaves(sample,subtractive,lineararithmetic);
90     SignalGenerators::normalize(lineararithmetic);
91     SignalGenerators::saveaudiofile(lineararithmetic,"01-lineararithmetic.aiff");
92     Render::saveimagefile(lineararithmetic,"02-lineararithmetic.png");
93
94     // Update progress
95     bar.set_progress(50);
96     bar.set_option(option::PostfixText{"Generating: Linear Arithmetic 2/2"});
97
98     env.applyenvelope(lineararithmetic,envelope);
99     SignalGenerators::normalize(lineararithmetic);
100    SignalGenerators::saveaudiofile(lineararithmetic,"03-final_lineararithmetic.aiff");
101    Render::saveimagefile(lineararithmetic,"04-final_lineararithmetic.png");
102
103    // Update progress
104    bar.set_progress(100);
105    bar.set_option(option::PostfixText{"Done 2/2"});
106
107    // Show cursor
108    show_console_cursor(true);
109    return 0;
110 }
111
112 namespace SignalGenerators
113 {
114     void gain(std::vector<float>& v, double gain)
115     {
116         for (uint32_t i=0;i<v.size();++i) {
117             v[i]=v[i]*gain;
118         }
119     }
120
121     void normalize(std::vector<float>& v)

```

```
122     {
123         float max=0.0,value=0.0;
124         for (uint32_t i=0;i<v.size();++i) {
125             value=v[i];
126             if (value > max) {max=value;}
127         }
128         // max=std::ceil(max);
129         for (uint32_t i=0;i<v.size();++i) {
130             // v[i]=v[i]/max;
131             v[i]=(v[i]/max)*0.707;
132         }
133     }
134
135 void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
136 {
137     for (uint32_t i=0;i<v1.size();++i)
138     {
139         v3[i]=v1[i]+v2[i];
140     }
141 }
142
143 void generatesubtractive(std::vector<float>& v)
144 {
145     AudioFile<float> a;
146     a.load("subtractive/08-postfilter.aiff");
147
148     for (int i=0;i<a.getNumSamplesPerChannel();++i)
149     {
150         for (int channel=0;channel<a.getNumChannels();++channel)
151         {
152             v.push_back(a.samples[channel][i]);
153         }
154     }
155 }
156
157 void generatesample(std::vector<float>& v)
158 {
159     AudioFile<float> a;
160     a.load("fm/01-fm.aiff");
161
162     for (int i=0;i<a.getNumSamplesPerChannel();++i)
163     {
164         for (int channel=0;channel<a.getNumChannels();++channel)
```

```

165         {
166             v.push_back(a.samples[channel][i]);
167         }
168     }
169 }
170
171 void saveaudiofile(std::vector<float>& v, std::string filename)
172 {
173     const std::string path="la/";
174     // Setup the audio file
175     AudioFile<float> a;
176     a.setNumChannels(1);
177     a.setBitDepth(24);
178     a.setNumSamplesPerChannel(44100);
179
180     for (int i=0;i<a.getNumSamplesPerChannel();++i)
181     {
182         for (int channel=0;channel<a.getNumChannels();++channel)
183         {
184             a.samples[channel][i]=v[i];
185         }
186     }
187     a.save(path+filename,AudioFileFormat::Aiff);
188 }
189 }
190
191 namespace Render
192 {
193     void fillbackground(png::image<png::rgb_pixel>& image)
194     {
195         png::rgb_pixel px(0x04,0x13,0x31);
196         for (uint32_t y=0;y<image.get_height();y++) {
197             for (uint32_t x=0;x<image.get_width();++x) {
198                 image.set_pixel(x,y,px);
199             }
200         }
201     }
202
203     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
204     {
205         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
206     ))
207     {

```

```

208         png::rgb_pixel px(0x7a,0xb1,0xe3);
209         image.set_pixel(x,y,px);
210     }
211 }
212
213 void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
214 {
215     int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
216     dx = x2 - x1; dy = y2 - y1;
217     if (dx == 0)
218     {
219         if (y2 < y1) std::swap(y1, y2);
220         for (y = y1; y <= y2; y++)
221             drawpx(image, x1, y);
222         return;
223     }
224     if (dy == 0)
225     {
226         if (x2 < x1) std::swap(x1, x2);
227         for (x = x1; x <= x2; x++)
228             drawpx(image, x, y1);
229         return;
230     }
231     dx1 = abs(dx); dy1 = abs(dy);
232     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
233     if (dy1 <= dx1)
234     {
235         if (dx >= 0)
236         {
237             x = x1; y = y1; xe = x2;
238         }
239         else
240         {
241             x = x2; y = y2; xe = x1;
242         }
243         drawpx(image, x, y);
244         for (i = 0; x<xe; i++)
245         {
246             x = x + 1;
247             if (px<0)
248                 px = px + 2 * dy1;
249             else
250                 {

```

```

251         if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
252         px = px + 2 * (dy1 - dx1);
253     }
254     drawpx(image, x, y);
255 }
256 }
257 else
258 {
259     if (dy >= 0)
260     {
261         x = x1; y = y1; ye = y2;
262     }
263     else
264     {
265         x = x2; y = y2; ye = y1;
266     }
267     drawpx(image, x, y);
268     for (i = 0; y<ye; i++)
269     {
270         y = y + 1;
271         if (py <= 0)
272             py = py + 2 * dx1;
273         else
274         {
275             if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
276             py = py + 2 * (dx1 - dy1);
277         }
278         drawpx(image, x, y);
279     }
280 }
281 }
282
283 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
284 {
285     uint32_t y=0, ox=0, oy=0;
286     for (uint32_t x=0; x<image.get_width(); ++x)
287     {
288         y=signalY[x];
289         if (x == 0) {ox=x;oy=y;}
290         drawline(image, x, y, ox, oy);
291         ox=x;oy=y;
292     }
293 }

```



```

294
295     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
296 :vector<uint32_t>& v2)
297     {
298         uint32_t halfHeight=image.get_height()/2;
299         double value=0.0;
300         if (v2.size() == 0 || v2.size() > v1.size()) {
301             v2.resize(v1.size());
302         }
303
304         for (uint32_t i=0;i<v1.size();++i)
305         {
306             value=v1[i];
307             if (value >= 0.0) {
308                 v2[i]=halfHeight-(halfHeight*value);
309             } else if (value < 0.0)
310             {
311                 v2[i]=halfHeight+(halfHeight*fabs(value));
312             }
313         }
314     }
315
316     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
317     {
318         fillbackground(image);
319         drawwave(image,v);
320     }
321
322     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
323     {
324         const std::string path="la/";
325         png::image<png::rgb_pixel> image(44100,600);
326         renderimage(image,v);
327         image.write(path+filename);
328     }
329
330     void saveimagefile(std::vector<float>& v2, std::string filename)
331     {
332         const std::string path="la/";
333         png::image<png::rgb_pixel> image(44100,600);
334         std::vector<uint32_t> v;
335         normalizedtoimg(image,v2,v);
336         renderimage(image,v);

```

```
337         image.write(path+filename);
338     }
339
340     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
341 v1, std::vector<uint32_t>& v2)
342     {
343         uint32_t height=image.get_height();
344         if (v2.size() == 0 || v2.size() > v1.size()) {
345             v2.resize(v1.size());
346         }
347         for (uint32_t i=0; i<v1.size(); ++i)
348         {
349             v2[i]=height-(height*v1[i]);
350         }
351     }
352
353     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
354     {
355         const std::string path="la/";
356         png::image<png::rgb_pixel> image(44100,600);
357         renderimage(image,v);
358         image.write(path+filename);
359     }
360
361     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
362     {
363         const std::string path="la/";
364         png::image<png::rgb_pixel> image(44100,600);
365         std::vector<uint32_t> v;
366         normalizedenvelopetoimg(image,v2,v);
367         renderimage(image,v);
368         image.write(path+filename);
369     }
370
371 }
```

---

**07-pd.cpp - 8581 bytes.**


---

```

1 // compile: clang++ -std=c++20 -lpng 07-pd.cpp -o 07-pd
2
3 #define _USE_MATH_DEFINES
4 #include <cmath>
5 #include <vector>
6 #include <random>
7 #include <filesystem>
8 #include "indicators.hpp"
9 #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "Envelope.hpp"
12
13 namespace Render
14 {
15     void fillbackground(png::image<png::rgb_pixel>& image);
16     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
17     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
18     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
19     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
20 :vector<uint32_t>& v2);
21     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
22     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
23     void saveimagefile(std::vector<float>& v2, std::string filename);
24     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
25 v1, std::vector<uint32_t>& v2);
26     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
27     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
28 }
29
30 namespace SignalGenerators
31 {
32     void gain(std::vector<float>& v, double gain);
33     void normalize(std::vector<float>& v);
34     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
35     void generatephasedistortionwave(std::vector<float>& v, int duration, float frequ\
36 encyInHz, float x1, float y1, float x2, bool isCosine);
37     void saveaudiofile(std::vector<float>& v, std::string filename);
38 }
39
40 int main()
41 {
42     namespace fs = std::filesystem;

```

```

43     fs::create_directory("pd");
44
45     const int duration=1;
46     const double sampleRate=44100.0;
47     std::vector<float> pd;
48     std::vector<float> envelope;
49
50     using namespace indicators;
51     // Hide cursor
52     show_console_cursor(false);
53
54     // Setup ProgressBar
55     ProgressBar bar{
56         option::BarWidth{50},
57         option::Start{"["},
58         option::Fill{"█"},
59         option::Lead{"█"},
60         option::Remainder{"-"},
61         option::End{" ]"},
62         option::PostfixText{"Generate Phase Distortion 1/2"},
63         option::ForegroundColor{Color::cyan},
64         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
65     };
66
67     // Update progress
68     bar.set_progress(0);
69
70     SignalGenerators::generatephasedistortionwave(pd,duration,440,0.2,0.5,0.7,true);
71     SignalGenerators::saveaudiofile(pd,"01-pd.aiff");
72     Render::saveimagefile(pd,"02-pd.png");
73
74     // Update progress
75     bar.set_progress(50);
76     bar.set_option(option::PostfixText{"Final Phase Distortion 2/2"});
77
78     ADSR::Envelope env(sampleRate,duration);
79     env.generateenvelope(envelope);
80     Render::saveenvelopeimage(envelope,"03-envelope.png");
81
82     env.applyenvelope(pd,envelope);
83     SignalGenerators::normalize(pd);
84     SignalGenerators::saveaudiofile(pd,"04-final-pd.aiff");
85     Render::saveimagefile(pd,"05-final-pd.png");

```

```

86
87     // Update progress
88     bar.set_progress(100);
89     bar.set_option(option::PostfixText{"Done 2/2"});
90
91     // Show cursor
92     show_console_cursor(true);
93     return 0;
94 }
95
96 namespace SignalGenerators
97 {
98     void gain(std::vector<float>& v, double gain)
99     {
100         for (uint32_t i=0;i<v.size();++i) {
101             v[i]=v[i]*gain;
102         }
103     }
104
105     void normalize(std::vector<float>& v)
106     {
107         float max=0.0,value=0.0;
108         for (uint32_t i=0;i<v.size();++i) {
109             value=v[i];
110             if (value > max) {max=value;}
111         }
112         // max=std::ceil(max);
113         for (uint32_t i=0;i<v.size();++i) {
114             // v[i]=v[i]/max;
115             v[i]=(v[i]/max)*0.707;
116         }
117     }
118
119     void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
120     {
121         for (uint32_t i=0;i<v1.size();++i)
122         {
123             v3[i]=v1[i]+v2[i];
124         }
125     }
126
127     void generatephasedistortionwave(std::vector<float>& v, int duration, float frequ\
128     encyInHz, float x1, float y1, float x2, bool isCosine)

```

```

129     {
130         const double sampleRate=44100.0;
131         float delta=frequencyInHz/(sampleRate*duration);
132         float pos=0;
133         float xDelta=(x2-x1)/(sampleRate*duration);
134         float warpedPos;
135         float m;
136         float b;
137         float x=x1;
138
139         for (uint32_t i=0;i<sampleRate*duration;++i)
140         {
141             if (pos < x)
142             {
143                 m=y1/x;
144                 warpedPos=m*pos;
145             } else {
146                 m=(1.0-y1)/(1.0-x);
147                 b=1.0-m;
148                 warpedPos=m*pos+b;
149             }
150             if (isCosine) {
151                 v.push_back(cos(2.0*M_PI*warpedPos));
152             } else {
153                 v.push_back(sin(2.0*M_PI*warpedPos));
154             }
155             pos += delta;
156             while (pos >= 1.0) {pos-=1.0;}
157             x+=xDelta;
158         }
159     }
160
161 void saveaudiofile(std::vector<float>& v, std::string filename)
162 {
163     const std::string path="pd/";
164     // Setup the audio file
165     AudioFile<float> a;
166     a.setNumChannels(1);
167     a.setBitDepth(24);
168     a.setNumSamplesPerChannel(44100);
169
170     for (int i=0;i<a.getNumSamplesPerChannel();++i)
171     {

```

```

172         for (int channel=0;channel<a.getNumChannels();++channel)
173         {
174             a.samples[channel][i]=v[i];
175         }
176     }
177     a.save(path+filename,AudioFileFormat::Aiff);
178 }
179 }
180
181 namespace Render
182 {
183     void fillbackground(png::image<png::rgb_pixel>& image)
184     {
185         png::rgb_pixel px(0x04,0x13,0x31);
186         for (uint32_t y=0;y<image.get_height();y++) {
187             for (uint32_t x=0;x<image.get_width();++x) {
188                 image.set_pixel(x,y,px);
189             }
190         }
191     }
192
193     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
194     {
195         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
196     ))
197         {
198             png::rgb_pixel px(0x7a,0xb1,0xe3);
199             image.set_pixel(x,y,px);
200         }
201     }
202
203     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
204     {
205         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
206         dx = x2 - x1; dy = y2 - y1;
207         if (dx == 0)
208         {
209             if (y2 < y1) std::swap(y1, y2);
210             for (y = y1; y <= y2; y++)
211                 drawpx(image, x1, y);
212             return;
213         }
214         if (dy == 0)

```

```
215         {
216             if (x2 < x1) std::swap(x1, x2);
217             for (x = x1; x <= x2; x++)
218                 drawpx(image, x, y1);
219             return;
220         }
221     dx1 = abs(dx); dy1 = abs(dy);
222     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
223     if (dy1 <= dx1)
224     {
225         if (dx >= 0)
226         {
227             x = x1; y = y1; xe = x2;
228         }
229         else
230         {
231             x = x2; y = y2; xe = x1;
232         }
233         drawpx(image, x, y);
234         for (i = 0; x<xe; i++)
235         {
236             x = x + 1;
237             if (px<0)
238                 px = px + 2 * dy1;
239             else
240             {
241                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
242                 px = px + 2 * (dy1 - dx1);
243             }
244             drawpx(image, x, y);
245         }
246     }
247     else
248     {
249         if (dy >= 0)
250         {
251             x = x1; y = y1; ye = y2;
252         }
253         else
254         {
255             x = x2; y = y2; ye = y1;
256         }
257         drawpx(image, x, y);
```



```

258         for (i = 0; y<ye; i++)
259         {
260             y = y + 1;
261             if (py <= 0)
262                 py = py + 2 * dx1;
263             else
264             {
265                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
266                 py = py + 2 * (dx1 - dy1);
267             }
268             drawpx(image, x, y);
269         }
270     }
271 }
272
273 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
274 {
275     uint32_t y=0, ox=0, oy=0;
276     for (uint32_t x=0; x<image.get_width(); ++x)
277     {
278         y=signalY[x];
279         if (x == 0) {ox=x; oy=y;}
280         drawline(image, x, y, ox, oy);
281         ox=x; oy=y;
282     }
283 }
284
285 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
286 :vector<uint32_t>& v2)
287 {
288     uint32_t halfHeight=image.get_height()/2;
289     double value=0.0;
290     if (v2.size() == 0 || v2.size() > v1.size()) {
291         v2.resize(v1.size());
292     }
293
294     for (uint32_t i=0; i<v1.size(); ++i)
295     {
296         value=v1[i];
297         if (value >= 0.0) {
298             v2[i]=halfHeight-(halfHeight*value);
299         } else if (value < 0.0)
300         {

```

```

301             v2[i]=halfHeight+(halfHeight*fabs(value));
302         }
303     }
304 }
305
306 void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
307 {
308     fillbackground(image);
309     drawwave(image,v);
310 }
311
312 void saveimagefile(std::vector<uint32_t>& v, std::string filename)
313 {
314     const std::string path="pd/";
315     png::image<png::rgb_pixel> image(44100,600);
316     renderimage(image,v);
317     image.write(path+filename);
318 }
319
320 void saveimagefile(std::vector<float>& v2, std::string filename)
321 {
322     const std::string path="pd/";
323     png::image<png::rgb_pixel> image(44100,600);
324     std::vector<uint32_t> v;
325     normalizedtoimg(image,v2,v);
326     renderimage(image,v);
327     image.write(path+filename);
328 }
329
330 void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
331 v1, std::vector<uint32_t>& v2)
332 {
333     uint32_t height=image.get_height();
334     if (v2.size() == 0 || v2.size() > v1.size()) {
335         v2.resize(v1.size());
336     }
337     for (uint32_t i=0; i<v1.size(); ++i)
338     {
339         v2[i]=height-(height*v1[i]);
340     }
341 }
342
343 void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)

```

```

344     {
345         const std::string path="pd/";
346         png::image<png::rgb_pixel> image(44100,600);
347         renderimage(image,v);
348         image.write(path+filename);
349     }
350
351     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
352     {
353         const std::string path="pd/";
354         png::image<png::rgb_pixel> image(44100,600);
355         std::vector<uint32_t> v;
356         normalizedenvelopetoimg(image,v2,v);
357         renderimage(image,v);
358         image.write(path+filename);
359     }
360
361 }

```

---

#### 08-scanned.cpp - 9965 bytes.

```

1  // compile: clang++ -std=c++20 -lpng 08-scanned.cpp -o 08-scanned
2
3  #define _USE_MATH_DEFINES
4  #include <cmath>
5  #include <vector>
6  #include <random>
7  #include <filesystem>
8  #include "indicators.hpp"
9  #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "Envelope.hpp"
12
13 #include <iostream>
14
15 namespace Render
16 {
17     void fillbackground(png::image<png::rgb_pixel>& image);
18     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
19     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
20     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
21     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
22 :vector<uint32_t>& v2);

```

```

23     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
24     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
25     void saveimagefile(std::vector<float>& v2, std::string filename);
26     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&& \
27 v1, std::vector<uint32_t>& v2);
28     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
29     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
30 }
31
32 namespace SignalGenerators
33 {
34     void gain(std::vector<float>& v, double gain);
35     void normalize(std::vector<float>& v);
36     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
37     void generatescannedwave(std::vector<float>& v, int duration, float frequencyInHz \
38 );
39     void saveaudiofile(std::vector<float>& v, std::string filename);
40 }
41
42 int main()
43 {
44     namespace fs = std::filesystem;
45     fs::create_directory("scanned");
46
47     const int duration=1;
48     const double sampleRate=44100.0;
49     std::vector<float> scanned;
50     std::vector<float> envelope;
51
52     using namespace indicators;
53     // Hide cursor
54     show_console_cursor(false);
55
56     // Setup ProgressBar
57     ProgressBar bar{
58         option::BarWidth{50},
59         option::Start{"["},
60         option::Fill{"█"},
61         option::Lead{"█"},
62         option::Remainder{"-"},
63         option::End{" ]"},
64         option::PostfixText{"Generate Scanned 1/2"},
65         option::ForegroundColor{Color::cyan},

```

```
66         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
67     };
68
69     // Update progress
70     bar.set_progress(0);
71
72     SignalGenerators::generatescannedwave(scanned,duration,440);
73     SignalGenerators::saveaudiofile(scanned,"01-scanned.aiff");
74     Render::saveimagefile(scanned,"02-scanned.png");
75
76     // Update progress
77     bar.set_progress(50);
78     bar.set_option(option::PostfixText{"Final Scanned 2/2"});
79
80     ADSR::Envelope env(sampleRate,duration);
81     env.generateenvelope(envelope);
82
83     env.applyenvelope(scanned,envelope);
84     SignalGenerators::normalize(scanned);
85     SignalGenerators::saveaudiofile(scanned,"03-final-scanned.aiff");
86     Render::saveimagefile(scanned,"04-final-scanned.png");
87
88     // Update progress
89     bar.set_progress(100);
90     bar.set_option(option::PostfixText{"Done 2/2"});
91
92     // Show cursor
93     show_console_cursor(true);
94     return 0;
95 }
96
97 namespace SignalGenerators
98 {
99     void gain(std::vector<float>& v, double gain)
100     {
101         for (uint32_t i=0;i<v.size();++i) {
102             v[i]=v[i]*gain;
103         }
104     }
105
106     void normalize(std::vector<float>& v)
107     {
108         float max=0.0,value=0.0;
```

```

109         for (uint32_t i=0;i<v.size();++i) {
110             value=v[i];
111             if (value > max) {max=value;}
112         }
113         // max=std::ceil(max);
114         for (uint32_t i=0;i<v.size();++i) {
115             // v[i]=v[i]/max;
116             v[i]=(v[i]/max)*0.707;
117         }
118     }
119
120 void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
121 {
122     for (uint32_t i=0;i<v1.size();++i)
123     {
124         v3[i]=v1[i]+v2[i];
125     }
126 }
127
128 typedef struct {
129     float k, b;
130     float updateRate, updateSize;
131 } SystemDesc;
132
133 typedef struct {
134     int size;
135     float *position;
136     float *velocity;
137 } ScanState;
138
139 void scan(std::vector<float>& v,ScanState *state, SystemDesc *system, float frequen\
140 cyInHz, float duration)
141 {
142     const double sampleRate=44100.0;
143     float *previousPosition=new float[128];
144     for (int i=0;i<sampleRate*duration;++i) {
145         auto scanPos = state->size * frequencyInHz * (float(i) / sampleRate);
146         int index = ((int) scanPos) % state->size;
147         v.push_back(state->position[index]);
148
149         float elapsed = 0;
150         while(elapsed < system->updateRate) {
151             previousPosition[state->size - 1] = state->position[state->size - 1];

```

```

152         for (int j = 0; j < state->size; j++) {
153             auto prevIndex = j > 0 ? j - 1 : state->size - 1;
154             auto nextIndex = j < state->size - 1 ? j + 1 : 0;
155             previousPosition[j] = state->position[j];
156             previousPosition[nextIndex] = state->position[nextIndex];
157
158             float prev = previousPosition[prevIndex];
159             float next = previousPosition[nextIndex];
160             float deltaX = (prev + next + 0) / 3 - state->position[j];
161             float force = system->k * deltaX - system->b * state->velocity[j];
162
163             state->velocity[j] += force * system->updateSize;
164             state->position[j] += state->velocity[j] * system->updateSize;
165         }
166         elapsed += system->updateSize;
167     }
168 }
169
170 delete [] previousPosition;
171 }
172
173 void generaterescannedwave(std::vector<float>& v, int duration, float frequencyInHz)
174 {
175     const double sampleRate=44100.0;
176     SystemDesc system = {.8, .01, .016, .01};
177     ScanState state;
178     state.size = 128;
179     state.position = new float[state.size];
180     state.velocity = new float[state.size];
181     std::random_device rd;
182     std::mt19937 gen(rd());
183     std::uniform_real_distribution<> dis(0.0, 1.0);
184
185     for (int i=0;i<state.size;++i) {
186         float randOffset=dis(gen);
187         float amplitude=0.1 + 0.5 * ((1+i) %2);
188         state.position[i]=sin(2*M_PI*i/state.size/4) * amplitude + randOffset;
189         state.velocity[i]=dis(gen);
190     }
191
192     scan(v,&state,&system,frequencyInHz,duration);
193
194     delete [] state.position;

```

```

195         delete [] state.velocity;
196     }
197
198     void saveaudiofile(std::vector<float>& v, std::string filename)
199     {
200         const std::string path="scanned/";
201         // Setup the audio file
202         AudioFile<float> a;
203         a.setNumChannels(1);
204         a.setBitDepth(24);
205         a.setNumSamplesPerChannel(44100);
206
207         for (int i=0;i<a.getNumSamplesPerChannel();++i)
208         {
209             for (int channel=0;channel<a.getNumChannels();++channel)
210             {
211                 a.samples[channel][i]=v[i];
212             }
213         }
214         a.save(path+filename,AudioFileFormat::Aiff);
215     }
216 }
217
218 namespace Render
219 {
220     void fillbackground(png::image<png::rgb_pixel>& image)
221     {
222         png::rgb_pixel px(0x04,0x13,0x31);
223         for (uint32_t y=0;y<image.get_height();y++) {
224             for (uint32_t x=0;x<image.get_width();++x) {
225                 image.set_pixel(x,y,px);
226             }
227         }
228     }
229
230     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
231     {
232         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
233     ))
234     {
235         png::rgb_pixel px(0x7a,0xb1,0xe3);
236         image.set_pixel(x,y,px);
237     }

```



```
238     }
239
240     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
241     {
242         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
243         dx = x2 - x1; dy = y2 - y1;
244         if (dx == 0)
245         {
246             if (y2 < y1) std::swap(y1, y2);
247             for (y = y1; y <= y2; y++)
248                 drawpx(image, x1, y);
249             return;
250         }
251         if (dy == 0)
252         {
253             if (x2 < x1) std::swap(x1, x2);
254             for (x = x1; x <= x2; x++)
255                 drawpx(image, x, y1);
256             return;
257         }
258         dx1 = abs(dx); dy1 = abs(dy);
259         px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
260         if (dy1 <= dx1)
261         {
262             if (dx >= 0)
263             {
264                 x = x1; y = y1; xe = x2;
265             }
266             else
267             {
268                 x = x2; y = y2; xe = x1;
269             }
270             drawpx(image, x, y);
271             for (i = 0; x<xe; i++)
272             {
273                 x = x + 1;
274                 if (px<0)
275                     px = px + 2 * dy1;
276                 else
277                 {
278                     if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
279                     px = px + 2 * (dy1 - dx1);
280                 }

```

```

281         drawpx(image, x, y);
282     }
283 }
284 else
285 {
286     if (dy >= 0)
287     {
288         x = x1; y = y1; ye = y2;
289     }
290     else
291     {
292         x = x2; y = y2; ye = y1;
293     }
294     drawpx(image, x, y);
295     for (i = 0; y<ye; i++)
296     {
297         y = y + 1;
298         if (py <= 0)
299             py = py + 2 * dx1;
300         else
301         {
302             if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
303             py = py + 2 * (dx1 - dy1);
304         }
305         drawpx(image, x, y);
306     }
307 }
308 }
309
310 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
311 {
312     uint32_t y=0, ox=0, oy=0;
313     for (uint32_t x=0; x<image.get_width(); ++x)
314     {
315         y=signalY[x];
316         if (x == 0) {ox=x;oy=y;}
317         drawline(image, x, y, ox, oy);
318         ox=x;oy=y;
319     }
320 }
321
322 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
323 :vector<uint32_t>& v2)

```

```
324     {
325         uint32_t halfHeight=image.get_height()/2;
326         double value=0.0;
327         if (v2.size() == 0 || v2.size() > v1.size()) {
328             v2.resize(v1.size());
329         }
330
331         for (uint32_t i=0;i<v1.size();++i)
332         {
333             value=v1[i];
334             if (value >= 0.0) {
335                 v2[i]=halfHeight-(halfHeight*value);
336             } else if (value < 0.0)
337             {
338                 v2[i]=halfHeight+(halfHeight*fabs(value));
339             }
340         }
341     }
342
343     void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
344     {
345         fillbackground(image);
346         drawwave(image,v);
347     }
348
349     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
350     {
351         const std::string path="scanned/";
352         png::image<png::rgb_pixel> image(44100,600);
353         renderimage(image,v);
354         image.write(path+filename);
355     }
356
357     void saveimagefile(std::vector<float>& v2, std::string filename)
358     {
359         const std::string path="scanned/";
360         png::image<png::rgb_pixel> image(44100,600);
361         std::vector<uint32_t> v;
362         normalizedtoimg(image,v2,v);
363         renderimage(image,v);
364         image.write(path+filename);
365     }
366
```

```

367     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
368     v1, std::vector<uint32_t>& v2)
369     {
370         uint32_t height=image.get_height();
371         if (v2.size() == 0 || v2.size() > v1.size()) {
372             v2.resize(v1.size());
373         }
374         for (uint32_t i=0; i<v1.size(); ++i)
375         {
376             v2[i]=height-(height*v1[i]);
377         }
378     }
379
380     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
381     {
382         const std::string path="scanned/";
383         png::image<png::rgb_pixel> image(44100,600);
384         renderimage(image,v);
385         image.write(path+filename);
386     }
387
388     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
389     {
390         const std::string path="scanned/";
391         png::image<png::rgb_pixel> image(44100,600);
392         std::vector<uint32_t> v;
393         normalizedenvelopetoimg(image,v2,v);
394         renderimage(image,v);
395         image.write(path+filename);
396     }
397
398 }

```

---

**09-vectorsynth.cpp - 10237 bytes.**


---

```

1 // compile: clang++ -std=c++20 -lpng 09-vectorsynth.cpp -o 09-vectorsynth
2 #define _USE_MATH_DEFINES
3 #include <cmath>
4 #include <vector>
5 #include <random>
6 #include <filesystem>
7 #include "indicators.hpp"
8 #include <png++/png.hpp>
9 #include "AudioFile/AudioFile.h"
10 #include "Envelope.hpp"
11
12 namespace Render
13 {
14     void fillbackground(png::image<png::rgb_pixel>& image);
15     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
16     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
17     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
18     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
19 :vector<uint32_t>& v2);
20     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
21     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
22     void saveimagefile(std::vector<float>& v2, std::string filename);
23     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
24 v1, std::vector<uint32_t>& v2);
25     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
26     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
27 }
28
29 namespace SignalGenerators
30 {
31     void gain(std::vector<float>& v, double gain);
32     void normalize(std::vector<float>& v);
33     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
34     void generatevectorsynth(std::vector<float>& v, int duration, double frequencyInHz);
35     void saveaudiofile(std::vector<float>& v, std::string filename);
36 }
37
38 int main()
39 {
40     namespace fs = std::filesystem;
41     fs::create_directory("vectorsynth");
42

```

```
43     const int duration=1;
44     const double sampleRate=44100.0;
45     std::vector<float> vectorsynth;
46     std::vector<float> envelope;
47
48     using namespace indicators;
49     // Hide cursor
50     show_console_cursor(false);
51
52     // Setup ProgressBar
53     ProgressBar bar{
54         option::BarWidth{50},
55         option::Start{"["},
56         option::Fill{"█"},
57         option::Lead{"█"},
58         option::Remainder{"-"},
59         option::End{" ]"},
60         option::PostfixText{"Setup Vector Synth 1/2"},
61         option::ForegroundColor{Color::cyan},
62         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
63     };
64
65     // Update progress
66     bar.set_progress(0);
67
68     SignalGenerators::generatevectorsynth(vectorsynth,duration,440);
69     SignalGenerators::saveaudiofile(vectorsynth,"01-vectorsynth.aiff");
70     Render::saveimagefile(vectorsynth,"02-vectorsynth.png");
71
72     // Update progress
73     bar.set_progress(50);
74     bar.set_option(option::PostfixText{"Generating: Vector Synth 2/2"});
75
76     ADSR::Envelope env(sampleRate,duration);
77     env.generateenvelope(envelope);
78     env.applyenvelope(vectorsynth,envelope);
79     SignalGenerators::saveaudiofile(vectorsynth,"03-final_vectorsynth.aiff");
80     Render::saveimagefile(vectorsynth,"04-final_vectorsynth.png");
81
82     // Update progress
83     bar.set_progress(100);
84     bar.set_option(option::PostfixText{"Done 2/2"});
85
```

```
86     // Show cursor
87     show_console_cursor(true);
88     return 0;
89 }
90
91 namespace SignalGenerators
92 {
93     void gain(std::vector<float>& v, double gain)
94     {
95         for (uint32_t i=0;i<v.size();++i) {
96             v[i]=v[i]*gain;
97         }
98     }
99
100    void normalize(std::vector<float>& v)
101    {
102        float max=0.0,value=0.0;
103        for (uint32_t i=0;i<v.size();++i) {
104            value=v[i];
105            if (value > max) {max=value;}
106        }
107        // max=std::ceil(max);
108        for (uint32_t i=0;i<v.size();++i) {
109            // v[i]=v[i]/max;
110            v[i]=(v[i]/max)*0.707;
111        }
112    }
113
114    void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
115    {
116        for (uint32_t i=0;i<v1.size();++i)
117        {
118            v3[i]=v1[i]+v2[i];
119        }
120    }
121
122    void generatetrianglwave(std::vector<float>& v, int duration, double frequencyInHz)
123    {
124        const double sampleRate=44100.0;
125
126        for (uint32_t i=0;i<sampleRate*duration;++i) {
127            v.push_back(M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate)));
128        }
129    }
130 }
```

```

129     }
130
131     void generateinversesawtoothwave(std::vector<float>& v, int duration, double frequen\
132 ncyInHz)
133     {
134         const double sampleRate=44100.0;
135         double period=sampleRate/frequencyInHz;
136         std::vector<float> temp;
137
138         for (uint32_t i=0;i<period;++i) {
139             temp.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
140         }
141         for (uint32_t i=0;i<sampleRate*duration;++i) {
142             v.push_back(temp[temp.size()-(i%temp.size())]);
143         }
144     }
145
146     void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz)
147     {
148         const double sampleRate=44100.0;
149
150         for (uint32_t i=0;i<sampleRate*duration;++i) {
151             v.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
152         }
153     }
154
155     void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz)
156     {
157         const double sampleRate=44100.0;
158         double period=sampleRate/frequencyInHz;
159         double dutyCycle=period*0.5;
160         double ss=0.0;
161
162         for (uint32_t i=0;i<sampleRate*duration;++i) {
163             if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
164             {
165                 ss=0.7;
166             } else {
167                 ss=-0.7;
168             }
169             v.push_back(ss);
170         }
171         v[0]=0.0;

```



```

172         v[v.size()-1]=0.0;
173     }
174
175     void generatevectorsynth(std::vector<float>& v, int duration, double frequencyInHz)
176     {
177         float vectortransition[4][4]={{1.0,0.5,0.25,0.0},{0.5,1.0,0.0,0.25},{0.25,0.0,1.0,\
178 0.5},{0.0,0.25,0.5,1.0}};
179         const double sampleRate=44100.0;
180         std::vector<float> sample1;
181         std::vector<float> sample2;
182         std::vector<float> sample3;
183         std::vector<float> sample4;
184
185         generatetrianglwave(sample1,duration,frequencyInHz);
186         generateinversesawtoothwave(sample2,duration,frequencyInHz);
187         generatesawtoothwave(sample3,duration,frequencyInHz);
188         generatesquarewave(sample4,duration,frequencyInHz);
189
190         int ptr=0;
191         for (int i=0;i<4;++i) {
192             for (int j=0;j<11025;++j) {
193                 sample1[ptr]=sample1[ptr]*vectortransition[i][0];
194                 sample2[ptr]=sample2[ptr]*vectortransition[i][1];
195                 sample3[ptr]=sample3[ptr]*vectortransition[i][2];
196                 sample4[ptr]=sample4[ptr]*vectortransition[i][3];
197                 ptr++;
198             }
199         }
200
201         v.resize(sample1.size());
202         addwaves(sample1,sample2,v);
203         addwaves(v,sample3,v);
204         addwaves(v,sample4,v);
205         normalize(v);
206     }
207
208     void saveaudiofile(std::vector<float>& v, std::string filename)
209     {
210         const std::string path="vectorsynth/";
211         // Setup the audio file
212         AudioFile<float> a;
213         a.setNumChannels(1);
214         a.setBitDepth(24);

```

```

215         a.setNumSamplesPerChannel(44100);
216
217         for (int i=0;i<a.getNumSamplesPerChannel();++i)
218         {
219             for (int channel=0;channel<a.getNumChannels();++channel)
220             {
221                 a.samples[channel][i]=v[i];
222             }
223         }
224         a.save(path+filename,AudioFileFormat::Aiff);
225     }
226 }
227
228 namespace Render
229 {
230     void fillbackground(png::image<png::rgb_pixel>& image)
231     {
232         png::rgb_pixel px(0x04,0x13,0x31);
233         for (uint32_t y=0;y<image.get_height();y++) {
234             for (uint32_t x=0;x<image.get_width();++x) {
235                 image.set_pixel(x,y,px);
236             }
237         }
238     }
239
240     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
241     {
242         if ((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
243     ))
244     {
245         png::rgb_pixel px(0x7a,0xb1,0xe3);
246         image.set_pixel(x,y,px);
247     }
248 }
249
250     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
251     {
252         int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
253         dx = x2 - x1; dy = y2 - y1;
254         if (dx == 0)
255         {
256             if (y2 < y1) std::swap(y1, y2);
257             for (y = y1; y <= y2; y++)

```

```
258             drawpx(image, x1, y);
259         return;
260     }
261     if (dy == 0)
262     {
263         if (x2 < x1) std::swap(x1, x2);
264         for (x = x1; x <= x2; x++)
265             drawpx(image, x, y1);
266         return;
267     }
268     dx1 = abs(dx); dy1 = abs(dy);
269     px = 2 * dy1 - dx1;          py = 2 * dx1 - dy1;
270     if (dy1 <= dx1)
271     {
272         if (dx >= 0)
273         {
274             x = x1; y = y1; xe = x2;
275         }
276         else
277         {
278             x = x2; y = y2; xe = x1;
279         }
280         drawpx(image, x, y);
281         for (i = 0; x<xe; i++)
282         {
283             x = x + 1;
284             if (px<0)
285                 px = px + 2 * dy1;
286             else
287             {
288                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y - 1;
289                 px = px + 2 * (dy1 - dx1);
290             }
291             drawpx(image, x, y);
292         }
293     }
294     else
295     {
296         if (dy >= 0)
297         {
298             x = x1; y = y1; ye = y2;
299         }
300         else
```

```

301         {
302             x = x2; y = y2; ye = y1;
303         }
304         drawpx(image, x, y);
305         for (i = 0; y<ye; i++)
306         {
307             y = y + 1;
308             if (py <= 0)
309                 py = py + 2 * dx1;
310             else
311             {
312                 if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
313                 py = py + 2 * (dx1 - dy1);
314             }
315             drawpx(image, x, y);
316         }
317     }
318 }
319
320 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
321 {
322     uint32_t y=0, ox=0, oy=0;
323     for (uint32_t x=0; x<image.get_width(); ++x)
324     {
325         y=signalY[x];
326         if (x == 0) {ox=x; oy=y;}
327         drawline(image, x, y, ox, oy);
328         ox=x; oy=y;
329     }
330 }
331
332 void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
333 :vector<uint32_t>& v2)
334 {
335     uint32_t halfHeight=image.get_height()/2;
336     double value=0.0;
337     if (v2.size() == 0 || v2.size() > v1.size()) {
338         v2.resize(v1.size());
339     }
340
341     for (uint32_t i=0; i<v1.size(); ++i)
342     {
343         value=v1[i];

```

```
344         if (value >= 0.0) {
345             v2[i]=halfHeight-(halfHeight*value);
346         } else if (value < 0.0)
347         {
348             v2[i]=halfHeight+(halfHeight*fabs(value));
349         }
350     }
351 }
352
353 void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
354 {
355     fillbackground(image);
356     drawwave(image,v);
357 }
358
359 void saveimagefile(std::vector<uint32_t>& v, std::string filename)
360 {
361     const std::string path="vectorsynth/";
362     png::image<png::rgb_pixel> image(44100,600);
363     renderimage(image,v);
364     image.write(path+filename);
365 }
366
367 void saveimagefile(std::vector<float>& v2, std::string filename)
368 {
369     const std::string path="vectorsynth/";
370     png::image<png::rgb_pixel> image(44100,600);
371     std::vector<uint32_t> v;
372     normalizedtoimg(image,v2,v);
373     renderimage(image,v);
374     image.write(path+filename);
375 }
376
377 void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
378 v1, std::vector<uint32_t>& v2)
379 {
380     uint32_t height=image.get_height();
381     if (v2.size() == 0 || v2.size() > v1.size()) {
382         v2.resize(v1.size());
383     }
384     for (uint32_t i=0; i<v1.size(); ++i)
385     {
386         v2[i]=height-(height*v1[i]);
```

```

387         }
388     }
389
390     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
391     {
392         const std::string path="vectorsynth/";
393         png::image<png::rgb_pixel> image(44100,600);
394         renderimage(image,v);
395         image.write(path+filename);
396     }
397
398     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
399     {
400         const std::string path="vectorsynth/";
401         png::image<png::rgb_pixel> image(44100,600);
402         std::vector<uint32_t> v;
403         normalizedenvelopetoimg(image,v2,v);
404         renderimage(image,v);
405         image.write(path+filename);
406     }
407
408 }

```

---

#### 10-virtualanalog.cpp - 1351 bytes.

```

1 // compile: clang++ -std=c++20 10-virtualanalog.cpp -o 10-virtualanalog
2 #include <iostream>
3
4 int main()
5 {
6     std::cout << "An analog modeling synthesizer is a synthesizer that generates the so\
7 unds of traditional analog synthesizers using DSP components and software algorithms\
8 . Analog modeling synthesizers simulate the behavior of the original electric and el\
9 ectronic circuitry in order to digitally replicate their tone.\n\n";
10    std::cout << "This method of synthesis is also referred to as Virtual Analog or VA.\
11 Analog modeling synthesizers can be more reliable than their true analog counterpar\
12 ts since the oscillator pitch is ultimately maintained by a digital clock, and the d\
13 igital hardware is typically less susceptible to temperature changes.\n\n";
14    std::cout << "While analog synthesizers need an oscillator circuit for each voice o\
15 f polyphony, analog modeling synthesizers don't face this problem. This means that m\
16 any of them, especially the more modern models, can produce as many polyphonic voice\
17 s as the CPU on which they run can handle.\n\n";
18    std::cout << "Modeling synths also provide patch storage capabilities and MIDI supp\

```

```

19 ort not found on most true analog instruments. Analog modeling synthesizers that run\
20 entirely within a host computer operating system are typically referred to as analo\
21 g software synthesizers.\n\n";
22     return 0;
23 }

```

---

#### 11-wavetable.cpp - 10677 bytes.

---

```

1 // compile: clang++ -std=c++20 -lpng 11-wavetable.cpp -o 11-wavetable
2 #define _USE_MATH_DEFINES
3 #include <cmath>
4 #include <vector>
5 #include <random>
6 #include <filesystem>
7 #include "indicators.hpp"
8 #include <png++/png.hpp>
9 #include "AudioFile/AudioFile.h"
10 #include "Envelope.hpp"
11
12 namespace Render
13 {
14     void fillbackground(png::image<png::rgb_pixel>& image);
15     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
16     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
17     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
18     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
19 :vector<uint32_t>& v2);
20     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
21     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
22     void saveimagefile(std::vector<float>& v2, std::string filename);
23     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
24 v1, std::vector<uint32_t>& v2);
25     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
26     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
27 }
28
29 namespace SignalGenerators
30 {
31     void gain(std::vector<float>& v, double gain);
32     void normalize(std::vector<float>& v);
33     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
34     void generatewavetable(std::vector<float>& v, int duration);
35     void saveaudiofile(std::vector<float>& v, std::string filename);

```

```
36 }
37
38 int main()
39 {
40     namespace fs = std::filesystem;
41     fs::create_directory("wavetable");
42
43     const double sampleRate=44100.0;
44     const int duration=1;
45     std::vector<float> wavetablewave;
46     std::vector<float> envelope;
47
48     using namespace indicators;
49     // Hide cursor
50     show_console_cursor(false);
51
52     // Setup ProgressBar
53     ProgressBar bar{
54         option::BarWidth{50},
55         option::Start{"["},
56         option::Fill{"█"},
57         option::Lead{"█"},
58         option::Remainder{"-"},
59         option::End{" ]"},
60         option::PostfixText{"Setting up: wavetable 1/2"},
61         option::ForegroundColor{Color::cyan},
62         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
63     };
64
65     // Update progress
66     bar.set_progress(0);
67
68     SignalGenerators::generatewavetable(wavetablewave,duration);
69     SignalGenerators::normalize(wavetablewave);
70     SignalGenerators::saveaudiofile(wavetablewave,"01-wavetablewave.aiff");
71     Render::saveimagefile(wavetablewave,"02-wavetablewave.png");
72
73     // Update progress
74     bar.set_progress(50);
75     bar.set_option(option::PostfixText{"Generating wavetable 2/2"});
76
77     ADSR::Envelope env(sampleRate,duration);
78     env.generateenvelope(envelope);
```



```

79     env.applyenvelope(wavetablewave, envelope);
80     SignalGenerators::saveaudiofile(wavetablewave, "03-final_wavetablewave.aiff");
81     Render::saveimagefile(wavetablewave, "04-final_wavetablewave.png");
82
83     // Update progress
84     bar.set_progress(100);
85     bar.set_option(option::PostfixText{"Done 2/2"});
86
87     // Show cursor
88     show_console_cursor(true);
89     return 0;
90 }
91
92 namespace SignalGenerators
93 {
94     void gain(std::vector<float>& v, double gain)
95     {
96         for (uint32_t i=0; i<v.size(); ++i) {
97             v[i]=v[i]*gain;
98         }
99     }
100
101     void normalize(std::vector<float>& v)
102     {
103         float max=0.0, value=0.0;
104         for (uint32_t i=0; i<v.size(); ++i) {
105             value=v[i];
106             if (value > max) {max=value;}
107         }
108         // max=std::ceil(max);
109         for (uint32_t i=0; i<v.size(); ++i) {
110             // v[i]=v[i]/max;
111             v[i]=(v[i]/max)*0.707;
112         }
113     }
114
115     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3)
116     {
117         for (uint32_t i=0; i<v1.size(); ++i)
118         {
119             v3[i]=v1[i]+v2[i];
120         }
121     }

```

```

122
123     int lfo[2048];
124     float wavetable[2048][2048];
125
126     void setuplfo()
127     {
128         for (int i=0;i<2048;++i)
129         {
130             lfo[i]=int(2047-(1023-1023*sin((static_cast<double> (i) / 2048) * 21.5 * 2.0
131 PI)));
132         }
133     }
134
135     void setupwavetable()
136     {
137         for (int i=0;i<2048;++i) {
138             wavetable[0][i]=M_2_PI*asin(sin(440*2*M_PI*i/2048));
139         }
140
141         for (int i=0;i<2048;++i) {
142             wavetable[511][i]=-2/M_PI*atan(1/tan(440*M_PI*i/2048));
143         }
144
145         for (int i=0;i<2048;++i) {
146             wavetable[1023][i]=sin((static_cast<double> (i) / 2048) * 440 * 2.0 * M_PI)
147         }
148
149         for (int i=0;i<2048;++i) {
150             wavetable[1535][i]=M_2_PI*asin(sin(440*2*M_PI*i/2048));
151         }
152
153         for (int i=0;i<2048;++i) {
154             wavetable[2047][i]=-2/M_PI*atan(1/tan(440*M_PI*i/2048));
155         }
156
157         float diff=0.0;
158         float step=0.0;
159         for (int i=0;i<2048;++i) {
160             diff=std::fabs(wavetable[0][i] - wavetable[511][i]);
161             step=diff/511;
162             for (int j=0;j<512;++j) {
163                 if (wavetable[0][i] < wavetable[511][i]) {
164                     wavetable[j][i]=wavetable[0][i]+(step*j);

```

```

165         } else if (wavetable[0][i] > wavetable[511][i]) {
166             wavetable[j][i]=wavetable[0][i]-(step*j);
167         } else if (wavetable[0][i] == wavetable[511][i]) {
168             wavetable[j][i]=wavetable[0][i];
169         }
170     }
171 }
172
173 for (int i=0;i<2048;++i) {
174     diff=std::fabs(wavetable[511][i] - wavetable[1023][i]);
175     step=diff/511;
176     for (int j=511;j<1024;++j) {
177         if (wavetable[511][i] < wavetable[1023][i]) {
178             wavetable[j][i]=wavetable[511][i]+(step*j);
179         } else if (wavetable[511][i] > wavetable[1023][i]) {
180             wavetable[j][i]=wavetable[511][i]-(step*j);
181         } else if (wavetable[511][i] == wavetable[1023][i]) {
182             wavetable[j][i]=wavetable[511][i];
183         }
184     }
185 }
186
187 for (int i=0;i<2048;++i) {
188     diff=std::fabs(wavetable[1023][i] - wavetable[1535][i]);
189     step=diff/511;
190     for (int j=1023;j<1536;++j) {
191         if (wavetable[1023][i] < wavetable[1535][i]) {
192             wavetable[j][i]=wavetable[1023][i]+(step*j);
193         } else if (wavetable[1023][i] > wavetable[1535][i]) {
194             wavetable[j][i]=wavetable[1023][i]-(step*j);
195         } else if (wavetable[1023][i] == wavetable[1535][i]) {
196             wavetable[j][i]=wavetable[1023][i];
197         }
198     }
199 }
200
201 for (int i=0;i<2048;++i) {
202     diff=std::fabs(wavetable[1535][i] - wavetable[2047][i]);
203     step=diff/511;
204     for (int j=1535;j<2048;++j) {
205         if (wavetable[1535][i] < wavetable[2047][i]) {
206             wavetable[j][i]=wavetable[1535][i]+(step*j);
207         } else if (wavetable[1535][i] > wavetable[2047][i]) {

```

```

208         wavetable[j][i]=wavetable[1535][i]-(step*j);
209     } else if (wavetable[1535][i] == wavetable[2047][i]) {
210         wavetable[j][i]=wavetable[1535][i];
211     }
212     }
213 }
214 }
215
216 void generatwavetable(std::vector<float>& v, int duration)
217 {
218     const double sampleRate=44100.0;
219     setuplfo();
220     setupwavetable();
221
222     for (int i=0;i<sampleRate*duration;++i) {
223         v.push_back(wavetable[lfo[i%2048]][i%2048]);
224     }
225 }
226
227 void saveaudiofile(std::vector<float>& v, std::string filename)
228 {
229     const std::string path="wavetable/";
230     // Setup the audio file
231     AudioFile<float> a;
232     a.setNumChannels(1);
233     a.setBitDepth(24);
234     a.setNumSamplesPerChannel(44100);
235
236     for (int i=0;i<a.getNumSamplesPerChannel();++i)
237     {
238         for (int channel=0;channel<a.getNumChannels();++channel)
239         {
240             a.samples[channel][i]=v[i];
241         }
242     }
243     a.save(path+filename,AudioFileFormat::Aiff);
244 }
245 }
246
247 namespace Render
248 {
249     void fillbackground(png::image<png::rgb_pixel>& image)
250     {

```



```
294     }
295     else
296     {
297         x = x2; y = y2; xe = x1;
298     }
299     drawpx(image, x, y);
300     for (i = 0; x < xe; i++)
301     {
302         x = x + 1;
303         if (px < 0)
304             px = px + 2 * dy1;
305         else
306         {
307             if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) y = y + 1; else y = y - 1;
308             px = px + 2 * (dy1 - dx1);
309         }
310         drawpx(image, x, y);
311     }
312 }
313 else
314 {
315     if (dy >= 0)
316     {
317         x = x1; y = y1; ye = y2;
318     }
319     else
320     {
321         x = x2; y = y2; ye = y1;
322     }
323     drawpx(image, x, y);
324     for (i = 0; y < ye; i++)
325     {
326         y = y + 1;
327         if (py <= 0)
328             py = py + 2 * dx1;
329         else
330         {
331             if ((dx < 0 && dy < 0) || (dx > 0 && dy > 0)) x = x + 1; else x = x - 1;
332             py = py + 2 * (dx1 - dy1);
333         }
334         drawpx(image, x, y);
335     }
336 }
```

```
337     }
338
339     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
340     {
341         uint32_t y=0, ox=0, oy=0;
342         for (uint32_t x=0; x<image.get_width(); ++x)
343         {
344             y=signalY[x];
345             if (x == 0) {ox=x; oy=y;}
346             drawline(image, x, y, ox, oy);
347             ox=x; oy=y;
348         }
349     }
350
351     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
352 :vector<uint32_t>& v2)
353     {
354         uint32_t halfHeight=image.get_height()/2;
355         double value=0.0;
356         if (v2.size() == 0 || v2.size() > v1.size()) {
357             v2.resize(v1.size());
358         }
359
360         for (uint32_t i=0; i<v1.size(); ++i)
361         {
362             value=v1[i];
363             if (value >= 0.0) {
364                 v2[i]=halfHeight-(halfHeight*value);
365             } else if (value < 0.0)
366             {
367                 v2[i]=halfHeight+(halfHeight*fabs(value));
368             }
369         }
370     }
371
372     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
373     {
374         fillbackground(image);
375         drawwave(image, v);
376     }
377
378     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
379     {
```

```
380         const std::string path="wavetable/";
381         png::image<png::rgb_pixel> image(44100,600);
382         renderimage(image,v);
383         image.write(path+filename);
384     }
385
386     void saveimagefile(std::vector<float>& v2, std::string filename)
387     {
388         const std::string path="wavetable/";
389         png::image<png::rgb_pixel> image(44100,600);
390         std::vector<uint32_t> v;
391         normalizedtoimg(image,v2,v);
392         renderimage(image,v);
393         image.write(path+filename);
394     }
395
396     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
397 v1,std::vector<uint32_t>& v2)
398     {
399         uint32_t height=image.get_height();
400         if (v2.size() == 0 || v2.size() > v1.size()) {
401             v2.resize(v1.size());
402         }
403         for (uint32_t i=0;i<v1.size();++i)
404         {
405             v2[i]=height-(height*v1[i]);
406         }
407     }
408
409     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
410     {
411         const std::string path="wavetable/";
412         png::image<png::rgb_pixel> image(44100,600);
413         renderimage(image,v);
414         image.write(path+filename);
415     }
416
417     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
418     {
419         const std::string path="wavetable/";
420         png::image<png::rgb_pixel> image(44100,600);
421         std::vector<uint32_t> v;
422         normalizedenvelopetoimg(image,v2,v);
```



```

423         renderimage(image,v);
424         image.write(path+filename);
425     }
426
427 }

```

---

### 12-physicalmodelling\_karplus\_strong.cpp - 9153 bytes.

---

```

1 // compile: clang++ -std=c++20 -lpng 12-physicalmodelling_karplus_strong.cpp -o 12-p\
2 hysicalmodelling_karplus_strong
3 #define _USE_MATH_DEFINES
4 #include <cmath>
5 #include <vector>
6 #include <random>
7 #include <filesystem>
8 #include "indicators.hpp"
9 #include <png++/png.hpp>
10 #include "AudioFile/AudioFile.h"
11 #include "Envelope.hpp"
12
13 namespace Render
14 {
15     void fillbackground(png::image<png::rgb_pixel>& image);
16     void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
17     void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
18     void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY);
19     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
20 :vector<uint32_t>& v2);
21     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v);
22     void saveimagefile(std::vector<uint32_t>& v, std::string filename);
23     void saveimagefile(std::vector<float>& v2, std::string filename);
24     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
25 v1, std::vector<uint32_t>& v2);
26     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
27     void saveenvelopeimage(std::vector<float>& v2, std::string filename);
28 }
29
30 namespace SignalGenerators
31 {
32     void gain(std::vector<float>& v, double gain);
33     void normalize(std::vector<float>& v);
34     void addwaves(std::vector<float>& v1, std::vector<float>& v2, std::vector<float>& v3);
35     void generatekarplusstrong(std::vector<float>& v, int duration, double frequencyInH\

```

```

36 z, double feedback);
37     void saveaudiofile(std::vector<float>& v, std::string filename);
38 }
39
40 int main()
41 {
42     namespace fs = std::filesystem;
43     fs::create_directory("physicalmodelling");
44
45     const int duration=1;
46     const double sampleRate=44100.0;
47     const double feedback=0.2;
48     std::vector<float> karplusstrong;
49     int octave=3;
50     int note=0;
51     double frequencyInHz=float(440 * pow(2.0, ((double)((octave - 4) * 12 + note)) / 12\
52 .0));
53
54     using namespace indicators;
55     // Hide cursor
56     show_console_cursor(false);
57
58     // Setup ProgressBar
59     ProgressBar bar{
60         option::BarWidth{50},
61         option::Start{"["},
62         option::Fill{"█"},
63         option::Lead{"█"},
64         option::Remainder{"-"},
65         option::End{" ]"},
66         option::PostfixText{"Generate Physical Modelling 1/1"},
67         option::ForegroundColor{Color::cyan},
68         option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
69     };
70
71     // Update progress
72     bar.set_progress(0);
73
74     SignalGenerators::generatekarplusstrong(karplusstrong,duration,frequencyInHz,0.996);
75     SignalGenerators::normalize(karplusstrong);
76     SignalGenerators::saveaudiofile(karplusstrong,"01-karplusstrong.aiff");
77     Render::saveimagefile(karplusstrong,"02-karplusstrong.png");
78

```

```

79     // Update progress
80     bar.set_progress(100);
81     bar.set_option(option::PostfixText{"Done 1/1"});
82
83     // Show cursor
84     show_console_cursor(true);
85     return 0;
86 }
87
88 namespace SignalGenerators
89 {
90
91     class KSString
92     {
93     public:
94         KSString(float frequencyInHz, float sampleRate, float feedback)
95         {
96             std::random_device rd;
97             std::mt19937 gen(rd());
98             std::uniform_real_distribution<> dis(-1.0, 1.0);
99
100            m_buffer.resize(uint32_t(float(sampleRate) / frequencyInHz));
101            for (size_t i = 0, c = m_buffer.size(); i < c; ++i) {
102                m_buffer[i] = dis(gen);
103            }
104            m_index = 0;
105            m_feedback = feedback;
106        }
107
108        float GenerateSample()
109        {
110            // get our sample to return
111            float ret = m_buffer[m_index];
112
113            // low pass filter (average) some samples
114            float value = (m_buffer[m_index] + m_buffer[(m_index + 1) % m_buffer.size()]
115 ] * 0.5f * m_feedback;
116            m_buffer[m_index] = value;
117
118            // move to the next sample
119            m_index = (m_index + 1) % m_buffer.size();
120
121            // return the sample from the buffer

```

```
122         return ret;
123     }
124
125     private:
126         std::vector<float> m_buffer;
127         size_t m_index;
128         float m_feedback;
129     };
130
131     void gain(std::vector<float>& v, double gain)
132     {
133         for (uint32_t i=0;i<v.size();++i) {
134             v[i]=v[i]*gain;
135         }
136     }
137
138     void normalize(std::vector<float>& v)
139     {
140         float max=0.0,value=0.0;
141         for (uint32_t i=0;i<v.size();++i) {
142             value=v[i];
143             if (value > max) {max=value;}
144         }
145         // max=std::ceil(max);
146         for (uint32_t i=0;i<v.size();++i) {
147             // v[i]=v[i]/max;
148             v[i]=(v[i]/max)*0.707;
149         }
150     }
151
152     void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
153     {
154         for (uint32_t i=0;i<v1.size();++i)
155         {
156             v3[i]=v1[i]+v2[i];
157         }
158     }
159
160     void generatekarplusstrong(std::vector<float>& v, int duration, double frequencyInH\
161 z, double feedback)
162     {
163         const double sampleRate=44100.0;
164         KSString pluck(frequencyInHz,sampleRate,0.996f);
```

```
165         v.resize(sampleRate);
166         for (uint32_t i=0;i<v.size();++i) {
167             v[i]=0;
168             v[i]+=pluck.GenerateSample();
169         }
170     }
171 }
172
173 void saveaudiofile(std::vector<float>& v, std::string filename)
174 {
175     const std::string path="physicalmodelling/";
176     // Setup the audio file
177     AudioFile<float> a;
178     a.setNumChannels(1);
179     a.setBitDepth(24);
180     a.setNumSamplesPerChannel(44100);
181
182     for (int i=0;i<a.getNumSamplesPerChannel();++i)
183     {
184         for (int channel=0;channel<a.getNumChannels();++channel)
185         {
186             a.samples[channel][i]=v[i];
187         }
188     }
189     a.save(path+filename,AudioFileFormat::Aiff);
190 }
191 }
192
193
194 namespace Render
195 {
196     void fillbackground(png::image<png::rgb_pixel>& image)
197     {
198         png::rgb_pixel px(0x04,0x13,0x31);
199         for (uint32_t y=0;y<image.get_height();y++) {
200             for (uint32_t x=0;x<image.get_width();++x) {
201                 image.set_pixel(x,y,px);
202             }
203         }
204     }
205
206     void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
207     {
```

```

208         if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height()))\
209     ))
210     {
211         png::rgb_pixel px(0x7a,0xb1,0xe3);
212         image.set_pixel(x,y,px);
213     }
214 }
215
216 void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
217 {
218     int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
219     dx = x2 - x1; dy = y2 - y1;
220     if (dx == 0)
221     {
222         if (y2 < y1) std::swap(y1, y2);
223         for (y = y1; y <= y2; y++)
224             drawpx(image, x1, y);
225         return;
226     }
227     if (dy == 0)
228     {
229         if (x2 < x1) std::swap(x1, x2);
230         for (x = x1; x <= x2; x++)
231             drawpx(image, x, y1);
232         return;
233     }
234     dx1 = abs(dx); dy1 = abs(dy);
235     px = 2 * dy1 - dx1;           py = 2 * dx1 - dy1;
236     if (dy1 <= dx1)
237     {
238         if (dx >= 0)
239         {
240             x = x1; y = y1; xe = x2;
241         }
242         else
243         {
244             x = x2; y = y2; xe = x1;
245         }
246         drawpx(image, x, y);
247         for (i = 0; x<xe; i++)
248         {
249             x = x + 1;
250             if (px<0)

```

```

251         px = px + 2 * dy1;
252     else
253     {
254         if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y - 1;
255         px = px + 2 * (dy1 - dx1);
256     }
257     drawpx(image, x, y);
258 }
259 }
260 else
261 {
262     if (dy >= 0)
263     {
264         x = x1; y = y1; ye = y2;
265     }
266     else
267     {
268         x = x2; y = y2; ye = y1;
269     }
270     drawpx(image, x, y);
271     for (i = 0; y<ye; i++)
272     {
273         y = y + 1;
274         if (py <= 0)
275             py = py + 2 * dx1;
276         else
277         {
278             if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x - 1;
279             py = py + 2 * (dx1 - dy1);
280         }
281         drawpx(image, x, y);
282     }
283 }
284 }
285
286 void drawwave(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& signalY)
287 {
288     uint32_t y=0, ox=0, oy=0;
289     for (uint32_t x=0; x<image.get_width(); ++x)
290     {
291         y=signalY[x];
292         if (x == 0) {ox=x;oy=y;}
293         drawline(image, x, y, ox, oy);

```

```

294             ox=x;oy=y;
295         }
296     }
297
298     void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1, std:\
299 :vector<uint32_t>& v2)
300     {
301         uint32_t halfHeight=image.get_height()/2;
302         double value=0.0;
303         if (v2.size() == 0 || v2.size() > v1.size()) {
304             v2.resize(v1.size());
305         }
306
307         for (uint32_t i=0;i<v1.size();++i)
308         {
309             value=v1[i];
310             if (value >= 0.0) {
311                 v2[i]=halfHeight-(halfHeight*value);
312             } else if (value < 0.0)
313             {
314                 v2[i]=halfHeight+(halfHeight*fabs(value));
315             }
316         }
317     }
318
319     void renderimage(png::image<png::rgb_pixel>& image, std::vector<uint32_t>& v)
320     {
321         fillbackground(image);
322         drawwave(image,v);
323     }
324
325     void saveimagefile(std::vector<uint32_t>& v, std::string filename)
326     {
327         const std::string path="physicalmodelling/";
328         png::image<png::rgb_pixel> image(44100,600);
329         renderimage(image,v);
330         image.write(path+filename);
331     }
332
333     void saveimagefile(std::vector<float>& v2, std::string filename)
334     {
335         const std::string path="physicalmodelling/";
336         png::image<png::rgb_pixel> image(44100,600);

```



```
337         std::vector<uint32_t> v;
338         normalizedtoimg(image,v2,v);
339         renderimage(image,v);
340         image.write(path+filename);
341     }
342
343     void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
344 v1,std::vector<uint32_t>& v2)
345     {
346         uint32_t height=image.get_height();
347         if (v2.size() == 0 || v2.size() > v1.size()) {
348             v2.resize(v1.size());
349         }
350         for (uint32_t i=0;i<v1.size();++i)
351         {
352             v2[i]=height-(height*v1[i]);
353         }
354     }
355
356     void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
357     {
358         const std::string path="physicalmodelling/";
359         png::image<png::rgb_pixel> image(44100,600);
360         renderimage(image,v);
361         image.write(path+filename);
362     }
363
364     void saveenvelopeimage(std::vector<float>& v2, std::string filename)
365     {
366         const std::string path="physicalmodelling/";
367         png::image<png::rgb_pixel> image(44100,600);
368         std::vector<uint32_t> v;
369         normalizedenvelopetoimg(image,v2,v);
370         renderimage(image,v);
371         image.write(path+filename);
372     }
373
374 }
```

---

**quiz.cpp - 338679 bytes.**


---

```

1 // compile: clang++ -std=c++20 quiz.cpp -o quiz
2 #include <iostream>
3 #include <fstream>
4 #include <vector>
5 #include <algorithm>
6 #include <random>
7
8 namespace quiz
9 {
10     class Quiz
11     {
12     public:
13         Quiz(const std::string &q, const std::string &a) {_a=a;_q=q;}
14         virtual ~Quiz() {}
15         std::string getQ() {return _q;}
16         std::string getA() {return _a;}
17     private:
18         std::string _q;
19         std::string _a;
20     };
21 }
22
23 std::vector<quiz::Quiz> game{
24     quiz::Quiz("0-5v", "Denotes a range of 0 to 5 volts, which is common for gates, trigg\
25     ers, and modulation control voltages in modular synthesizers. Gates and triggers - w\
26     high initiate events such as new notes - typically rise from 0v to 5v (0 to 10v is a\
27     lso common), with roughly the middle of that onset starting the event. Gates are con\
28     sidered high when held at 5v (or 10v), and then low when they return to 0v."),
29     quiz::Quiz("1 pole", "This format of numbers and abbreviations (dB/oct = decibels per\
30     octave) is often used to refer to the frequency response behavior of a filter. A fi\
31     lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
32     ters) the frequency spectrum of a signal going through it so that its loudness is mu\
33     ltiples of 6 decibels weaker for each octave further away you get from the cutoff fr\
34     equency. A 6dB/octave filter is often referred to as a "one pole" filter (as each po\
35     le of a filter's design results in 6dB of attenuation), and has a relatively weak ef\
36     fect on the signal going through it. Low Pass Gates (LPGs) typically - but not alway\
37     s - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 deci\
38     bels for every octave above its cutoff frequency."),
39     quiz::Quiz("1 ppqn", "The most common sequencer clock division forwards it one step (\
40     pulse) per quarter note. This is often the core sync pulse that is distributed in a \
41     modular system, and is either multiplied or divided to create other musical division\
42     s."),

```

```
43 quiz::Quiz("1 v/oct", "The most common standard for controlling pitch in a modular sy\
44 nthesizer. Under the system, increasing the voltage going into a VCO (Voltage Contro\
45 lled Oscillator) 1 volt – say, from 0.5v to 1.5v – would raise its pitch by one octa\
46 ve."),
47 quiz::Quiz("1.2 v/oct", "Buchla compatible synths have standardized on the 1.2 volt p\
48 er octave system, instead of the more common 1 v/oct. With 12 semitones to an octave\
49 in Western music, an equally tempered scale would work out to precisely 0.1 volts f\
50 or a change in pitch of 1 semitone."),
51 quiz::Quiz("1/4\"", "The most common connector size used for 5U (Moog format) modular \
52 synthesizers. These are TS (tip/sleeve) jacks and plugs, similar to guitar and other\
53 instrument cables."),
54 quiz::Quiz("1/8\"", "Often used to incorrectly describe the connector size commonly us\
55 ed in Eurorack format modules, as well as Buchla audio signals. In fact, Eurorack mo\
56 dules use 3.5mm jacks and plugs (slightly larger than 1/8"); Buchla uses Switchcraft\
57 Tini-Jax connectors. Tini-Jax are 3.5mm in diameter, but are slightly different phy\
58 sically from a common 3.5 mm jack. 1/8" plugs would be loose in both of these jacks,\
59 so make sure you get 3.5mm connectors ordering parts or cables for these formats.")\
60 ,
61 quiz::Quiz("10 vpp", "An abbreviation for \"10 volts peak to peak\" with peak to peak\
62 being the difference between the lowest and highest voltage reached during a signal\
63 's travels. This is a common voltage range for both audio and modulation signals in \
64 a modular synthesizer. The actual range is between -5 and +5 volts. The precise rang\
65 e may be varied to change the depth of their effect, so don't get too hung up on spe\
66 cific voltage ranges. Pay more attention to whether they vary between 0v and some va\
67 lue, or swing in roughly equal amounts both above and below 0v (as 10vpp does)."),
68 quiz::Quiz("12 dB/oct", "This format of numbers and abbreviations (dB/oct = decibels \
69 per octave) is often used to refer to the frequency response behavior of a filter. A\
70 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\
71 filters) the frequency spectrum of a signal going through it so that its loudness is\
72 multiples of 12 decibels weaker for each octave further away you get from the cutof\
73 f frequency. A 12dB/octave filter is often referred to as a "two pole" filter (as ea\
74 ch pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and \
75 Oberheim instruments often featured 2-pole filters, often resulting in brighter sou\
76 nds when compared to those with 4-pole instruments."),
77 quiz::Quiz("16'", "Sometimes seen on octave selector switches on oscillators. It refe\
78 rs to the length of an organ pipe. Longer pipes = lower pitches; 16' is in the mid-b\
79 ass range. A pipe or setting half as long (8') is one octave higher; a pipe half as \
80 long again (4') is two octaves higher; etc."),
81 quiz::Quiz("18 dB/oct", "This format of numbers and abbreviations (dB/oct = decibels \
82 per octave) is often used to refer to the frequency response behavior of a filter. A\
83 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\
84 filters) the frequency spectrum of a signal going through it so that its loudness is\
85 multiples of 18 decibels weaker for each octave further away you get from the cutof\
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86 f frequency. It is often used a coded shorthand for when someone wants to refer to a\  
87 cid-type bass lines from a TB-303 without mentioning the instrument by name."),  
88 quiz::Quiz("2 Pole","This format of numbers and abbreviations (dB/oct = decibels per\  
89 octave) is often used to refer to the frequency response behavior of a filter. A fi\  
90 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\  
91 ters) the frequency spectrum of a signal going through it so that its loudness is mu\  
92 ltiples of 12 decibels weaker for each octave further away you get from the cutoff f\  
93 requency. A 12dB/octave filter is often referred to as a "two pole" filter (as each \  
94 pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and Obe\  
95 rheim instruments often featured 2-pole filters, often resulting in brighter sounds \  
96 when compared to those with 4-pole instruments."),  
97 quiz::Quiz("2.5 mm","A common screw thread size used to mount Eurorack modules. This\  
98 size is most common when using a system of loose nuts that slide along the rails th\  
99 at the modules are attached to."),  
100 quiz::Quiz("24 dB/oct","This format of numbers and abbreviations (dB/oct = decibels \  
101 per octave) is often used to refer to the frequency response behavior of a filter. A\  
102 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\  
103 filters) the frequency spectrum of a signal going through it so that its loudness is\  
104 multiples of 24 decibels weaker for each octave further away you get from the cutof\  
105 f frequency. This design is often used in vintage Moog and Roland synths. 4-pole fil\  
106 ters are often associated with subjectively fatter, more "round" sounds than 2-pole \  
107 filters - but generalizations are always dangerous."),  
108 quiz::Quiz("24 ppqn","A common master clock division used in MIDI, DIN sync, and oth\  
109 er systems common to electronic music and synthesizers. It means internally, 24 subd\  
110 ivisions of time are counted for every quarter note at the current tempo. This fast \  
111 internal clock can then be divided down to create sixteenth notes ( $\div 6$ ), eighth notes\  
112 ( $\div 12$ ), eight note triplets ( $\div 8$ ), etc."),  
113 quiz::Quiz("2'", "Sometimes seen on octave selector switches for oscillators. It refe\  
114 rs to the length of an organ pipe. Shorter pipes = higher pitches; 2' is rarely seen\  
115 on modular oscillators as it's rather high in pitch - two octaves above middle C as\  
116 a starting point. A pipe or setting twice as long (4') is one octave lower; a pipe \  
117 twice as long again (8') is two octaves lower; etc."),  
118 quiz::Quiz("3 mm","A common screw thread size used to mount Eurorack modules. This s\  
119 ize is most common when using module mounting rails that have been pre-drilled."),  
120 quiz::Quiz("3 Pole","This format of numbers and abbreviations (dB/oct = decibels per\  
121 octave) is often used to refer to the frequency response behavior of a filter. A fi\  
122 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\  
123 ters) the frequency spectrum of a signal going through it so that its loudness is mu\  
124 ltiples of 18 decibels weaker for each octave further away you get from the cutoff f\  
125 requency. It is often used a coded shorthand for when someone wants to refer to acid\  
126 -type bass lines from a TB-303 without mentioning the instrument by name."),  
127 quiz::Quiz("3.5 mm","The standard connector size used for jacks and cables in Eurora\  
128 ck format modular synthesizers. Note that this is slightly larger than 1/8")."),
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129 quiz::Quiz("303","The TB-303 Bass Line by Roland became a cult favorite in Acid Hous\
130 e and other flavors of EDM (Electronic Dance Music) for its rubbery, slithery synth \
131 bass sound. Many attribute the sound of the 303 to its filter design;"),
132 quiz::Quiz("32'","Sometimes seen on octave selector switches on oscillators. It refe\
133 rs to the length of an organ pipe. Longer pipes = lower pitches; 32' is the lowest s\
134 etting you will see and is getting into earthquake territory. A pipe or setting half\
135 as long (16') is one octave higher; a pipe half as long again (8') is two octaves h\
136 igher; etc."),
137 quiz::Quiz("3U","Refers to modules that are 3 rack units (U) high – the Eurorack sta\
138 ndard, which is by far the most common modular format today, even though it's one of\
139 the youngest formats."),
140 quiz::Quiz("4 Pole","This format of numbers and abbreviations (dB/oct = decibels per\
141 octave) is often used to refer to the frequency response behavior of a filter. A fi\
142 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
143 ters) the frequency spectrum of a signal going through it so that its loudness is mu\
144 ltiples of 24 decibels weaker for each octave further away you get from the cutoff f\
145 requency. This design is often used in vintage Moog and Roland synths. 4-pole filter\
146 s are often associated with subjectively fatter, more “round” sounds than 2-pole fil\
147 ters – but generalizations are always dangerous."),
148 quiz::Quiz("4-40","A screw thread size occasionally used to mount Eurorack modules. \
149 This size is used by Pittsburgh Modular for their cases, for example."),
150 quiz::Quiz("4U","Refers to modules that are 4U (rack units) high – namely, Buchla an\
151 d Serge systems, as well as do-it-yourself clones of these modules. Both Buchla and \
152 Serge lean toward a more experimental approach to synthesis and music, so some users\
153 wear “4U” as a badge of honor that they’re non-conformist and cool. (And they are.)\
154 "),
155 quiz::Quiz("4'","Sometimes seen on octave selector switches on oscillators. It refer\
156 s to the length of an organ pipe. Shorter pipes = higher pitches; 4' is the highest \
157 octave setting you will see on most oscillators. A pipe or setting twice as long (8'\
158 ) is one octave lower; a pipe twice as long again (16') is two octaves lower; etc.")\
159 ,
160 quiz::Quiz("5U","Refers to modules that are 5U (rack units) or 8.75” (22.2 cm) high,\
161 which is most often associated with the vintage Moog standard and those who have fo\
162 llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
163 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
164 fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
165 d is both historical and physically large, some users “5U” as a badge of honor that \
166 they’re traditional and cool. (And the are.) There was also a briefly popular 5U for\
167 mat from MOTM that used a different width and power connection. It has since been di\
168 scontinued, but there are still diehard MOTM format users today."),
169 quiz::Quiz("6 dB/oct","This format of numbers and abbreviations (dB/oct = decibels p\
170 er octave) is often used to refer to the frequency response behavior of a filter. A \
171 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (f\
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172 filters) the frequency spectrum of a signal going through it so that its loudness is \
173 multiples of 6 decibels weaker for each octave further away you get from the cutoff \
174 frequency. A 6dB/octave filter is often referred to as a "one pole" filter (as each \
175 pole of a filter's design results in 6dB of attenuation), and has a relatively weak \
176 effect on the signal going through it. Low Pass Gates (LPGs) typically - but not alw\
177 ays - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 de\
178 cibels for every octave above its cutoff frequency."),
179 quiz::Quiz("808","The TR-808 Rhythm Composer by Roland created all of its sounds usi\
180 ng analog circuitry. When it first came out, it was not well loved, as the analog so\
181 unds weren't realistic enough. But later, music styles such as House and Hip-Hop ado\
182 pted its big, booming synthetic sounds. When a module says it recreates "808" drums,\
183 this is the instrument they are trying to emulate. Most copied is the 808 kick drum\
184 sound, which tends to be a low-pitched, long-decaying sine-like wave often with a s\
185 nappy attack."),
186 quiz::Quiz("8'", "Sometimes seen on octave selector switches on oscillators. It refer\
187 s to the length of an organ pipe. Shorter pipes = higher pitches; 8' is typically as\
188 sociated with middle C. A pipe or setting half as long (4') is one octave higher; a \
189 pipe or setting twice as long (16') is one octave lower."),
190 quiz::Quiz("909","The TR-909 Rhythm Composer was the follow-up to Roland's now-rever\
191 ed TR-808. It combined digital samples for the hi-hat and cymbal along with the 808'\
192 s analog sounds, and has also become popular. When a module says it produces 909-lik\
193 e sounds, this is the instrument it is referencing."),
194 quiz::Quiz("A-440","This is the frequency in hertz (cycles per second) of the A abov\
195 e Middle C. It is often used as a tuning reference."),
196 quiz::Quiz("A/B Technique","A stereo microphone placement technique in which two car\
197 dioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart fr\
198 om each other (depending on the size of the sound source) to create a left/right ste\
199 reo image. Also known as Spaced Pair."),
200 quiz::Quiz("A/D","Abbreviation of Analog-to-Digital Conversion, the conversion of a \
201 quantity that has continuous changes (like electrical signals) into numbers that app\
202 roximate those changes (i.e., computer data)."),
203 quiz::Quiz("Absolute Phase","This term describes a perfect polarity between an origi\
204 nal signal (into the microphone) and the reproduced signal (through the speaker). Wh\
205 en positive pressure exerted upon the microphone is translated as positive pressure \
206 to the loudspeaker, the two are in "absolute phase."."),
207 quiz::Quiz("Absorption","In acoustics, absorption is what happens when sound waves a\
208 re absorbed by a surface, as opposed to bouncing off the surface (reflection). Absor\
209 ptive materials in a control room, for example, tend to "deaden" the sound of the ro\
210 om because the sound energy is absorbed rather than reflected. (See also "Reflection\
211 .")"),
212 quiz::Quiz("AC Coupled","An AC coupled input attempts to remove any constant DC volt\
213 age going through it. This is useful if have an audio signal (such as the output of \
214 an oscillator) which is AC in nature, and you want to remove any accidental DC offse\
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215 t that might have crept into it. These offsets can cause one half of the AC waveform\  
216 to clip prematurely, or can cause clicks at the start and end of envelopes or mutes\  
217 . However, this coupling can mildly distort a wave going through it, as in essence A\  
218 C coupling is a high pass filter that is attempting to remove very low frequency com\  
219 ponents."),  
220 quiz::Quiz("AC", "Alternating Current - The type of electrical current found in stand\  
221 ard electrical outlets and studio signals running through audio lines. In AC, the cu\  
222 rrent “alternates” directions, flowing back and forth through the circuit. In modula\  
223 r terms, AC refers to a voltage that alternates between positive and negative values\  
224 - such as the output of an oscillator."),  
225 quiz::Quiz("Accelerometer", "A device that measures the acceleration to which it is s\  
226 ubjected and creates an electric signal to match it. In music and audio, acceleromet\  
227 ers are found in such things as microphones and guitar pickups."),  
228 quiz::Quiz("Acorn Tube", "Named for its acorn-like shape, an acorn tube is a small va\  
229 cuum tube used in ultra high frequency (UHF) electronics such as tube amplifiers."),  
230 quiz::Quiz("Acoustic Amplifier", "The part of a musical instrument that vibrates in r\  
231 esponse to the initial vibration of the instrument, causing the surrounding air to m\  
232 ove more efficiently and making the sound louder. For example: the body of an acoust\  
233 ic guitar, the bell of a horn, a drum’s shell, and the wooden soundboard of a piano.\  
234 "),  
235 quiz::Quiz("Acoustic Echo Chamber", "A room designed with hard, non-parallel surfaces\  
236 to create reverberation. In recording studios, they are used to add natural reverb \  
237 to a dry signal."),  
238 quiz::Quiz("Acoustics", "The science of the sound—more specifically, the science of t\  
239 he properties and behavior of sound waves. A good understanding of acoustics is esse\  
240 ntial to audio engineering and studio design."),  
241 quiz::Quiz("Active Device", "A component that is designed with the ability to control\  
242 electrical current (as opposed to a “Passive Device”). In the recording studio, act\  
243 ive devices are generally components that include an amplifier. (See also “Passive D\  
244 evices.”)"),  
245 quiz::Quiz("Active Multiple", "Quite often you need to split or copy a signal to send\  
246 to more than one destination. This is commonly done with a multiple, where you plug\  
247 one source in, and then plug in additional patch cables to go off to multiple desti\  
248 nations. An active or buffered multiple is one that includes a buffer circuit betwee\  
249 n the input and output, making sure the signal does not lose its strength or integri\  
250 ty by being split too many times, and that no funny business happening on one of the\  
251 outputs affects any of the other connections. Some modules have good buffering buil\  
252 t into their outputs, and can drive multiple modules without issue. But if you try t\  
253 o use a passive mult to connect to, say, three oscillators, and you realize the trac\  
254 king isn't very good (they quickly go out of tune as you go up and down the scale), \  
255 then you need a buffered mult instead."),  
256 quiz::Quiz("Actuator", "The part of a switch that causes change of the contact connec\  
257 tions (e.g., toggle, pushbutton, or rocker)."),
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258 quiz::Quiz("AD","Shorthand for a two-stage Attack/Decay envelope. This simple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 volts) at a speed defined by its Attack parameter, and then immediately falls back to 0 volts at a rate defined by its Decay parameter. A variation on this is the AHD envelope: After finishing the Attack stage, it holds at the maximum level for a specified amount of time (in contrast to an AR envelope, which holds at the maximum level for as long as the note on gate is high), and then decays back to zero. I have heard there are some envelopes that are a hybrid of AHD and AR in that they hold the maximum level for either the defined Hold time or the as long as the incoming gate is high;"),
261
262 quiz::Quiz("Additive Synthesis","One of the main properties that make a sound unique is the mixture of harmonics - pure component frequencies - that it is built from. Additive synthesis is a technique that gives you direct control over each of those component harmonics, allowing you to directly dial in the mix you want. As immediate and intuitive as that sounds on paper (or on screen), in reality it takes a lot of work to craft the correct mixture to recreate another sound, especially since the strength of each harmonic usually varies over time. Additive synthesis oscillators are relatively rare in modular synths; two examples are the Verbo Harmonic Oscillator and the Make Noise TELHARMONIC."),
266
267 quiz::Quiz("ADSR","An envelope generator with four stages: Attack, Decay, Sustain, and Release. When this envelope generator receives a gate input, it typically starts at 0 volts (which is the equivalent of silence when connected to a Voltage Controlled Amplifier, or the lowest frequency when connected to a voltage controlled filter or oscillator) and raises to the maximum voltage it can output (typically 5 to 10 volts depending on system; it can often be set with an output level control) over a time set by the Attack control. Once it reaches that level, the output voltage immediately starts dropping to speed set by the Decay control until it reaches the voltage set by the Sustain control. If the input gate is still active, this level is maintained until the gate goes back to 0 volts (usually because you released the key on a controlling keyboard, etc.). At that time, the output voltage then starts dropping back to 0 volts at the rate set by the Release control."),
276
277 quiz::Quiz("AES","Audio Engineering Society."),
278
279 quiz::Quiz("AES3","(sometimes called AES/EBU) A digital audio transfer standard developed by the Audio Engineering Society and the European Broadcasting Union for carrying dual-channel digital audio data between devices. AES3 is the protocol behind XLR cables, as well as RCA and S/PDIF cables."),
283
284 quiz::Quiz("AFG","The AFG (Audio Frequency Generator) is a very full-featured analog oscillator released by Livewire Electronics. It has since been discontinued, but refurbished B-stock units come up for sale every now and then. The expansion modules were, to the best of my knowledge, never released (at least not widely)."),
288
289 quiz::Quiz("Aftertouch","(Also called "Pressure Sensitivity") some keyboards measure how hard you press down on the keys, and convert this to a voltage (or other control signal such as MIDI, which can then be converted into a control voltage) that you can use to add expression to a note, such as adding vibrato or opening the filter with
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301 der. Monophonic aftertouch measures one pressure value for the entire keyboard, rega\
302 rdless of which key(s) you are pressing; polyphonic aftertouch produces a signal for\
303 each individual key. Important trivia: Touch plate keyboards actually measure the s\
304 urface area of the skin touching them rather than pressure or force – so you can inc\
305 rease or decrease the aftertouch amount by rolling between the tip and length of you\
306 r finger."),
307 quiz::Quiz("AHDSR","Attack, Hold, Decay, Sustain, and Release. This is a slightly fa\
308 ncier ADSR envelope that holds the voltage typically at its maximum value for a spec\
309 ified time after the attack is done rising and before the decay starts falling."),
310 quiz::Quiz("Aliasing","A type of digital signal distortion that occurs in a sampler \
311 when the incoming signal frequency exceeds the Nyquist frequency for that unit. The \
312 sampler reproduces it at an incorrect frequency, or an “alias,” causing a distortion\
313 or artifact in the sound. If you play back a digital audio file where half of the s\
314 ample rate is an audible pitch, you will also hear a mirror image of the sound’s har\
315 monic content reproduced started at that half-sample-rate pivot (unless some excelle\
316 nt filtering has taken place). (See also “Nyquist Frequency.”)."),
317 quiz::Quiz("Alternating Current (or AC)","The type of electrical current found in st\
318 andard electrical outlets and studio signals running through audio lines. In AC, the\
319 current “alternates” directions, flowing back and forth through the circuit."),
320 quiz::Quiz("AM","Amplitude Modulation (AM) is the name given the to the technique of\
321 varying the amplitude or loudness of one signal known as the carrier (typically an \
322 audio signal, swinging both above and below 0 volts) with a second signal called the\
323 modulator. In the typical amplitude modulation (AM) scenario, a low frequency oscil\
324 lator with a positive voltage (say, between 0v and 5v, or maybe something smaller su\
325 ch as between 1v and 2v) is fed into the control input of a voltage controlled ampli\
326 fier to add vibrato to an audio signal passing through it. Technically, this is know\
327 n as a two-quadrant multiplier or modulator, as any negative swings in the modulatio\
328 n signal are ignored; when patching tremolo, you may need to make sure an offset vol\
329 tage is being added to your LFO to make sure the sound doesn’t cut out on the lower \
330 excursions of the LFO’s waveform."),
331 quiz::Quiz("Ambience","In most cases, this refers to the “atmosphere” of a certain p\
332 lace, like a restaurant. But in recording, it refers to the part of the sound that c\
333 omes from the surrounding environment rather than directly from the sound source. Fo\
334 r example, the sound waves coming into your ears from a cello being played are comin\
335 g directly from the source, but the sound of the same cello coming to you after boun\
336 cing off the back wall is ambient sound."),
337 quiz::Quiz("Ambient Field","The area away from the sound source where the reverberat\
338 ion is louder than the direct sound."),
339 quiz::Quiz("Ambient Miking","This refers to placing a microphone in the ambient fiel\
340 d of a room to record the ambient reverberations of the sound. The recording enginee\
341 r often does this in addition to direct micing of the instrument(s) to create a blen\
342 d or mix of direct and reverberant sound in the recording."),
343 quiz::Quiz("Amp","An abbreviation for “Amplifier,” “Amplitude” or “Ampere,” dependin\
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344 g on context."),
345 quiz::Quiz("Ampere","The unit of measure for electrical current, abbreviated Amp."),
346 quiz::Quiz("Amplifier","A device that increases the level or amplitude of an electri\
347 cal signal, making the resulting sound louder."),
348 quiz::Quiz("Amplitude Modulation","Amplitude Modulation (AM) is the name given the t\
349 o the technique of varying the amplitude or loudness of one signal known as the carr\
350 ier (typically an audio signal, swinging both above and below 0 volts) with a second\
351 signal called the modulator. In the typical amplitude modulation (AM) scenario, a l\
352 ow frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe so\
353 mething smaller such as between 1v and 2v) is fed into the control input of a voltag\
354 e controlled amplifier to add vibrato to an audio signal passing through it. Technic\
355 ally, this is known as a two-quadrant multiplier or modulator, as any negative swing\
356 s in the modulation signal are ignored; when patching tremolo, you may need to make \
357 sure an offset voltage is being added to your LFO to make sure the sound doesn't cut\
358 out on the lower excursions of the LFO's waveform."),
359 quiz::Quiz("Amplitude","The height of a waveform above or below the zero line. In au\
360 dio, this usually translates to the signal strength or the volume of the sound."),
361 quiz::Quiz("Analog Recording","A recording of the continuous changes of an audio wav\
362 eform. The most common example of analog recording in a recording studio is recordin\
363 g on reel-to-reel magnetic tape."),
364 quiz::Quiz("Analog Shift Register","An Analog Shift Register (ASR) is a cross betwee\
365 n a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how t\
366 hose work). When initially triggered, it samples the incoming voltage, and presents \
367 that at its first output. On the second trigger, the incoming voltage is sampled aga\
368 in with this new voltage presented at the first output, while the original voltage i\
369 s now moved to a second output. This game of \"telephone\" is passed along for as ma\
370 ny stages as the ASR has - traditionally three or four."),
371 quiz::Quiz("Analog To Digital Converter (A/D; or ADC)","A device that translates a c\
372 ontinuously changing signal (analog) into numeric values that approximate those chan\
373 ges (digital). In audio recording, this refers to converting recorded sound from ele\
374 ctrical voltages to computerized data."),
375 quiz::Quiz("Analog","The term analog implies a signal is continuously variable, comp\
376 ared to digital where a signal has been converted into discrete numbers. In the land\
377 of modular synthesizers, analog refers to a circuit design that has no digital (or \
378 at least, computer-based) components - instead, it does all of its processing using \
379 transistors, diodes, capacitors, and the such rather than CPUs and DSPs."),
380 quiz::Quiz("AND function","One of the most common Boolean or binary logic functions,\
381 AND says only output a gate on signal if all of the inputs see \"high\" gate signals \
382 (i.e. input 1 and input 2 etc. all have gate ons). A NAND function has an inverted o\
383 utput: The output would be low if both inputs were high, but otherwise would be high\
384 ."),
385 quiz::Quiz("AR","The two-stage Attack/Release envelope raises from 0 volts to its ma\
386 ximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack param\
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387 eter, and then stays at that value for as long as the gate signal fed into the envel\
388 ope generator stays high. Then when the gate signal goes back to zero, the envelope'\
389 s output also falls back to zero at a rate set by its Release parameter. (There is a\
390 separate type of envelope known as an AHD – Attack/Hold/Decay – where you specify a\
391 fixed time for the level to stay at its maximum, rather than pay attention to the g\
392 ate signal.)"),
393 quiz::Quiz("Arpeggiator","Putting on our music theory hat for a second, an arpeggio \
394 is a type of “broken chord” where the notes are played individually rather than all \
395 at once. An arpeggiator – usually built into a keyboard, or a device inserted betwee\
396 n your keyboard and sound module – makes it easier for you to play arpeggios: You ju\
397 st hold down the notes of the chord, and it automatically plays the notes one at a t\
398 ime, over and over again, like a step sequencer you can program on the fly just by h\
399 olding down a chord. Good arpeggiators have options for different patterns (up, down\
400 , back and forth, random, etc.), and even a latch or hold where it will keep doing t\
401 his even after you’ve released the keys."),
402 quiz::Quiz("ASR","An Analog Shift Register (ASR) is a cross between a Sample & Hold \
403 module and a Bucket Brigade Delay (assuming you already know how those work). When i\
404 nitially triggered, it samples the incoming voltage, and presents that at its first \
405 output. On the second trigger, the incoming voltage is sampled again with this new v\
406 oltage presented at the first output, while the original voltage is now moved to a s\
407 econd output. This game of \"telephone\" is passed along for as many stages as the A\
408 SR has – traditionally three or four."),
409 quiz::Quiz("Attack/Decay/Sustain/Release","An envelope generator with four stages: A\
410 ttack, Decay, Sustain, and Release. When this envelope generator receives a gate inp\
411 ut, it typically starts at 0 volts (which is the equivalent of silence when connecte\
412 d to a Voltage Controlled Amplifier, or the lowest frequency when connected to a vol\
413 tage controlled filter or oscillator) and raises to the maximum voltage it can outpu\
414 t (typically 5 to 10 volts depending on system; it can often be set with an output l\
415 evel control) over a time set by the Attack control. Once it reaches that level, the\
416 output voltage immediately starts dropping to speed set by the Decay control it unt\
417 il it reaches the voltage set by the Sustain control. If the input gate is still act\
418 ive, this level is maintained until the gate goes back to 0 volts (usually because y\
419 ou released the key on a controlling keyboard, etc.). At that time, the output volta\
420 ge then starts dropping back to 0 volts at the rate set by the Release control."),
421 quiz::Quiz("Attack/Decay","Shorthand for a two-stage Attack/Decay envelope. This sim\
422 ple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perh\
423 aps 10 volts) at a speed defined by its Attack parameter, and then immediately falls\
424 back to 0 volts at a rate defined by its Decay parameter. A variation on this is th\
425 e AHD envelope: After finishing the Attack stage, it holds at the maximum level for \
426 a specified amount of time (in contrast to an AR envelope, which holds at the maximu\
427 m level for as long as the note on gate is high), and then decays back to zero. I ha\
428 ve heard there are some envelopes that a hybrid of AHD and AR in that they hold the \
429 maximum level for either the defined Hold time or the as long as the incoming gate i\
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430 s high;"),
431 quiz::Quiz("Attack/Hold/Decay/Sustain/Release", "This is a slightly fancier ADSR envelope that holds the voltage typically at its maximum value for a specified time after the attack is done rising and before the decay starts falling."),
432
433 r the attack is done rising and before the decay starts falling."),
434 quiz::Quiz("Attack/Release", "The two-stage Attack/Release envelope raises from 0 volts to its maximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack parameter, and then stays at that value for as long as the gate signal fed into the envelope generator stays high. Then when the gate signal goes back to zero, the envelope's output also falls back to zero at a rate set by its Release parameter. (There is a separate type of envelope known as an AHD - Attack/Hold/Decay - where you specify a fixed time for the level to stay at its maximum, rather than pay attention to the gate signal.)"),
440
441 ion to the gate signal.)"),
442 quiz::Quiz("Attack", "This usually refers to the first stage of an envelope that occurs at the onset of a note, as it rises from 0 volts (silence when controlling an amplifier module) to typically the value of maximum loudness. Percussive and plucked sounds have very fast attacks; slow, languid wind or string instrument phrases may have long attacks."),
446
447 he reduction of electrical or acoustic signal strength. In audio, attenuation is measured in decibels (dB) and is typically heard as a reduction in volume. Sound waves traveling through the air naturally attenuate as they travel away from the source of the sound. Engineers also purposefully attenuate signals in the studio through gain controls or pads to prevent overload."),
451
452 ge going through it."),
453
454 he polarity of the signal or voltage going through it. Most attenuverters use pass through no signal at their center position; as you turn them clockwise, you turn up the normal version of the signal; as you turn them counterclockwise, they turn up an inverted version of the signal. Some attenuverters are a normal attenuator with a polarity switch added on."),
459
460 y full-featured analog oscillator released by Livewire Electronics. It has since been discontinued, but refurbished B-stock units come up for sale every now and then. The expansion modules were, to the best of my knowledge, never released (at least not widely)."),
464
465 ns can hear with our ears. In the technical sense, audio refers to the transmission, recording or reproduction of sound, whether digitally, electrically or acoustically."),
468
469 "The process of re-recording dialogue for film in a controlled environment after the film is shot, for the purpose of replacing poorly recorded dialogue."),
471
472 "A compressor with a long release time, which is
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473     used to keep the volume of the audio at a consistent level."),
474     quiz::Quiz("Automation","Programming certain changes to occur automatically during r\
475     ecording and/or playback. In the studio, engineers use automation on their consoles \
476     or computers so various parameters will change automatically at different times duri\
477     ng multitrack recording and playback. This pre-programming feature makes it easier t\
478     o create those changes than attempting to perform them all manually in real time."),
479     quiz::Quiz("Auxiliary Equipment","External signal processing devices that work along\
480     side the mixing console to modify the signal."),
481     quiz::Quiz("Auxiliary Return","(Abbreviated Aux Return or Return) The input on a con\
482     sole or DAW that returns the effected signal sent through the auxiliary send back in\
483     to the channel mix."),
484     quiz::Quiz("Auxiliary Send","(Abbreviated Aux Send or Send) A control to adjust the \
485     signal level being sent from the input channel on a console or DAW to auxiliary equi\
486     pment or plug-ins through the auxiliary bus. This is typically used for creating an \
487     effects loop that processes a portion of the signal, then returns it into the mix th\
488     rough the auxiliary return."),
489     quiz::Quiz("Axis","An imaginary line around which a device operates. For example: in\
490     microphone use, the axis is an imaginary line coming out from the front of the micr\
491     ophone in the direction of motion of the diaphragm, delineating the optimum locati\
492     on for the mic to pick up the sound. Sounds that occur "off-axis" from the microphone \
493     will not be picked up as clearly."),
494     quiz::Quiz("Background Noise","Refers to either 1) The ambient noise in a room unrel\
495     ated to the instrument(s) or vocal(s) being recorded; or 2) The system noise unrelat\
496     ed to the recorded signal. (All electronics emit a level of noise.)"),
497     quiz::Quiz("Baffles","Sound absorbing panels that are used to prevent sound waves fr\
498     om entering or leaving a space."),
499     quiz::Quiz("Balance","1) The relative level of two or more instruments in a mix, or \
500     the relative level of audio signals in the channels of a stereo recording. 2) To eve\
501     n out the relative levels of audio signals in the channels of stereo recording."),
502     quiz::Quiz("Balanced Audio","This refers to a system where three wires are used to c\
503     arry an audio signal: one is the ground (the 0 volt reference), the second carries t\
504     he audio signal as it varies above and below 0v, and the third carries an inverted c\
505     opy of the audio signal that goes negative while the original is going positive. Bal\
506     anced audio usually implies a reference signal level of +4dB (higher than line level\
507     ; still lower than most modular synths), although microphone signals - much weaker b\
508     y comparison, and therefore more susceptible to outside noise - are almost always ba\
509     lanced as well. Modular synths tend to use unbalanced audio for their internal signa\
510     ls. If you require a balanced output (or input), you need a special module that conv\
511     erts between balanced and unbalanced audio, plus does any necessary level matching.")\
512     ),
513     quiz::Quiz("Balanced Cable","A cable consisting of three wires (two signal wires and\
514     a ground wire) and two connectors. The two signal wires carry the same signal in op\
515     posite polarities, providing protection against interference and noise in a balanced\
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516 system. Examples of balanced cables include tip-ring-sleeve (TRS) stereo cables and\  
517 XLR cables."),  
518 quiz::Quiz("Balanced Mixer","A circuit or device that generates the sum and differen\  
519 ce frequencies of two input signals."),  
520 quiz::Quiz("Balanced Modulator","Balanced or ring modulation is a special type of am\  
521 plitude modulation, where one bipolar (swinging both above and below 0 volts) signal\  
522 - the modulator - is used to vary the amplitude of a second bipolar signal, known a\  
523 s the carrier. The modulator's frequency is both added to and subtracted from the ca\  
524 rrier's frequency; the resulting harmonics replace the original carrier and modulato\  
525 r."),  
526 quiz::Quiz("Banana","An alternate type of connector (https://en.wikipedia.org/wiki/B\  
527 anana\_connector) used by 4U systems such as Buchla (control voltages) and Serge (bot\  
528 h control and audio). These cables have only one wire, so they carry only the signal\  
529 , relying on the module panels and chassis of the system to provide the ground refer\  
530 ence. Banana connectors have an advantage in that they are usually "stackable" meani\  
531 ng you can plug a one jack into the back of another, providing a passive multiple.")\  
532 ,  
533 quiz::Quiz("Band Pass Filter","A device, circuit or plug-in that allows a narrow ban\  
534 d of frequencies to pass through the circuit, rejecting or attenuating frequencies t\  
535 hat are either higher or lower than the specified range."),  
536 quiz::Quiz("Band Stop Filter","A device, circuit or plug-in that attenuates a narrow\  
537 band of frequencies in the signal, allowing frequencies outside the band to pass. T\  
538 he exact opposite of a band pass filter."),  
539 quiz::Quiz("Band Track","(Sometimes abbreviated "Track") A mixdown of a song minus t\  
540 he lead vocal and/or background vocals. In other words, a mixed track containing onl\  
541 y the instrumental parts of the song."),  
542 quiz::Quiz("Band","1) A range of frequencies, often identified by the center frequen\  
543 cy of the range. 2) A group of musicians playing together."),  
544 quiz::Quiz("Bandpass Filter","A bandpass filter (BPF) leaves the harmonics around th\  
545 e center, corner or cutoff frequency untouched, and attenuates those above and below\  
546 the center frequency. The further away you get from the center, the more they are a\  
547 ttenuated, based on the number of poles in the filter, with each pole equalling 6 de\  
548 cibels of attenuation for each octave you get away from that center."),  
549 quiz::Quiz("Bandwidth","In signal processing, bandwidth refers to the usable frequen\  
550 cy range of a communication channel, measured by the difference between the device's\  
551 highest and lowest usable frequencies."),  
552 quiz::Quiz("Bank","1) A collection of sound patches, sequencer data and/or operating\  
553 parameters of a synthesizer's generators and modifiers in memory. 2) A group of sou\  
554 nd modules as a unit."),  
555 quiz::Quiz("Bar","In music notation, bar is another term for measure a specified per\  
556 iod of time containing a certain number of beats, and marked by bar lines on each si\  
557 de of the written measure."),  
558 quiz::Quiz("Bark Scale","The human auditory (hearing) system can be thought of as co\  

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559 nsisting of a series of bandpass filters. Interestingly, the spacing of these filter\  
560 s do not strictly follow either a linear frequency scale or a logarithmic musical sc\  
561 ale. The Bark Scale is an attempt to determine what the center frequency and bandwid\  
562 th of those \"hearing filters\" are (known as critical bands).\"),  
563 quiz::Quiz(\"Barrier Miking\", \"A microphone placement technique in which a microphone \  
564 is placed close to a reflective surface. When done correctly, barrier miking ensures\  
565 that both the direct and reflected sounds reach the microphone simultaneously, prev\  
566 enting phase cancellation between the two.\"),  
567 quiz::Quiz(\"Basic Session\", \"The first audio recording session for recording the basi\  
568 c tracks that serve as the song’s foundation (for example, the drums and bass).\"),  
569 quiz::Quiz(\"Bass Reflex\", \"A type of loudspeaker cabinet design in which a port (open\  
570 ing) in the speaker cabinet enhances bass frequencies. The principle is that the sou\  
571 nd pressure generated by the back of the speaker cone inside the cabinet is routed o\  
572 ut the port at the front of the cabinet, mixed with the sound coming from the front \  
573 of the woofer. Changing the port size and position will greatly change the character\  
574 of the low frequencies.\"),  
575 quiz::Quiz(\"Bass\", \"The lower range of audio frequencies up to approximately 250 Hz. \  
576 A reference value.\"),  
577 quiz::Quiz(\"BBD\", \"An early design for an echo or delay effect where the input audio \  
578 would be sampled as an analog voltage, and held for a brief moment. Then at the next\  
579 above-audio sample rate clock pulse, this voltage would get passed to the next samp\  
580 le and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade d\  
581 elays (BBDs) usually have numbers of stages or buckets that are powers of two (256, \  
582 512, 1024, 2048, etc.); the delay length is determined by the number of stages multi\  
583 plied by the time interval between samples.\"),  
584 quiz::Quiz(\"Beaming\", \"A phenomenon found in loudspeakers in which higher frequencies\  
585 are projected straight out of the loudspeaker, rather than dispersing along with th\  
586 e lower frequencies. When you stand on-axis in front of the speaker, it sounds as th\  
587 ough it is only reproducing the high frequencies, rather than the mids or lows. This\  
588 phenomenon is alleviated by routing the high frequencies through horns in the loudsp\  
589 eaker.\"),  
590 quiz::Quiz(\"Beat Mapping\", \"The process of adjusting the tempo variations in a record\  
591 ed piece of music to fit the set tempo of the project. In a DAW, this is done using \  
592 time stretching tools and cuts to synchronize the transients to the appropriate temp\  
593 o markers. This technique is often used, for example, to reconcile a drum or bass pe\  
594 rformance that was recorded without a click track.\"),  
595 quiz::Quiz(\"Beat\", \"1) The steady, even pulse in music. 2) The action of two sounds o\  
596 r audio signals of slightly different frequency interfering with one another and cau\  
597 sing periodic increases and decreases in volume, heard to the ear as “beats.”\"),  
598 quiz::Quiz(\"Beating\", \"When two oscillators are tuned to very nearly – but not quite \  
599 – the same frequency, the difference between them causes an interference pattern kno\  
600 wn as beating. When the difference in frequency is below the audio rate, this can so\  
601 und like a tremolo applied to the loudness of the combined sound.\"),
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602 quiz::Quiz("Beatmatching","A technique predominantly used by DJs to synchronize the \
603 tempos of two recorded tracks, generally through the use of time stretching and pitch \
604 h shifting tools, to create a seamless transition from one song into another."),
605 quiz::Quiz("Beats Per Minute (B.P.M.)","BPM (beats per minute) is the most common way \
606 y of stating tempo: How many beats (typically, quarter notes) should be counted every \
607 y minute. A tempo of 120 beats per minute means there would be two beats every second \
608 d (120 beats/minute x 1 minute/60 seconds = 2). The number of steady even pulses in \
609 music occurring in one minute, defining the tempo of the song."),
610 quiz::Quiz("Berlin School","A particular style of electronic music popularized by the \
611 e likes of Tangerine Dream and Klaus Schulze based on analog synthesizers, heavy on \
612 repetitive sequences and floating chords or drones with solos played on top. More recent \
613 versions of Berlin School music can be heard from Node and Red Shift."),
614 quiz::Quiz("Bi-amplification","A technique in which high and low frequencies in a speaker \
615 eaker or speaker system are driven by two separate amplifiers."),
616 quiz::Quiz("Bi-Directional Pattern","A microphone pickup pattern which is most sensitive \
617 tive to picking up sounds directly in front and back of the mic, effectively rejecting \
618 ng sounds coming from the sides. Also called a "figure-8 pattern.""),
619 quiz::Quiz("Binary","A cornerstone of digital systems is the binary counting method, \
620 where each digit can have only two different values: 0 or 1; off or on; low or high \
621 . A binary signal can only have one of these two states. Therefore, a gate or trigger \
622 r signal in a modular synth – even if generated by analog circuitry – could be referred \
623 red to as a binary type signal. See the entry for Boolean for things you can do with \
624 binary signals like gates and divided clocks."),
625 quiz::Quiz("Bipolar","A voltage that can range both above and below zero is referred \
626 to as bipolar. Some modulation signals inside a modular synth – such as vibrato (varying \
627 ying the pitch of an oscillator both above and below the note it is supposed to be \
628 playing) – are bipolar in nature."),
629 quiz::Quiz("Bit","The smallest unit of digital information representing a single "0" \
630 or "1.""),
631 quiz::Quiz("Bitrate (or Bit Depth)","In digital recording, the number of computer bits \
632 ts used to describe each sample. The greater the bitrate, the greater the dynamic range \
633 nge of the sampled sound. The quality and resolution of an audio sample are described \
634 d as a combination of sample rate and bitrate. (See also "Sample Rate.""),
635 quiz::Quiz("Blending","The mixing of multiple sounds or channels together to form one \
636 e sound, or mixing the left and right signals together."),
637 quiz::Quiz("Blue Noise","Technically, a type of noise whose power density (spectral \
638 loudness) increases 3 dB per octave with increasing frequency. It has a very "hissy" \
639 characteristic, lacking in bass."),
640 quiz::Quiz("Boolean","Boolean logic only can have two states: high or low; 1 or 0; on \
641 n or off."),
642 quiz::Quiz("Boom Stand","A microphone stand equipped with a telescoping support arm \
643 to hold the microphone."),
644 quiz::Quiz("Boom","A telescoping support arm attached to a microphone stand holding \
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645 the microphone."),
646 quiz::Quiz("Boost","To increase gain at specific frequencies with an equalizer."),
647 quiz::Quiz("Bouncing","(also called “Ping-Ponging” or “Ponging“) The technique of co\
648 mbining and mixing multiple tracks onto one or two tracks (mono or stereo). This can\
649 be done in real-time or analog by playing the tracks through the console and record\
650 ing them onto separate tracks, or digitally through a digital audio workstation. Bou\
651 ncing was once used frequently by engineers to free up additional tracks for recordi\
652 ng, but in digital workstations where tracks are virtually unlimited, this practice \
653 is basically obsolete. Today, engineers typically bounce tracks for the purpose of c\
654 reating a preliminary or final mix of a song."),
655 quiz::Quiz("Boundary Microphone","An omnidirectional microphone designed to be place\
656 d flush against a flat surface (or boundary), effectively creating a “half-Omni” pic\
657 kup pattern while eliminating the danger of phase issues from reflected sounds. A po\
658 pular type of boundary microphone is Crown Audio’s trademark Pressure Zone Microphon\
659 e (PZM)."),
660 quiz::Quiz("BPF","A bandpass filter (BPF) leaves the harmonics around the center, co\
661 rner or cutoff frequency untouched, and attenuates those above and below the center \
662 frequency. The further away you get from the center, the more they are attenuated, b\
663 ased on the number of poles in the filter, with each pole equalling 6 decibels of at\
664 tenuation for each octave you get away from that center."),
665 quiz::Quiz("BPM","BPM (beats per minute) is the most common way of stating tempo: Ho\
666 w many beats (typically, quarter notes) should be counted every minute. A tempo of 1\
667 20 beats per minute means there would be two beats every second (120 beats/minute x \
668 1 minute/60 seconds = 2)."),
669 quiz::Quiz("Breathing","Pumping and Breathing – In studio jargon, an effect created \
670 when a compressor is rapidly compressing and releasing the sound, creating audible c\
671 hanges in the signal level. “Pumping” generally refers to the audible increase of so\
672 und levels after compression has taken place; “breathing” refers to a similar effect\
673 with vocals, raising the signal volume just as the vocalist is inhaling. Pumping an\
674 d breathing is a sign of cheap compression or over-compression, and is usually undes\
675 irable, although some engineers and musicians use it on purpose occasionally to crea\
676 te a particular effect."),
677 quiz::Quiz("Brickwall Filter","A certain type of low-pass filter exhibiting a steep \
678 cutoff slope which resembles a “brick wall.” While these filters are often found in \
679 A/D converters to prevent aliasing, their steep cutoff can introduce unwanted side-e\
680 ffects to the audio signal, such as phase shift."),
681 quiz::Quiz("Bridging","A technique of feeding a single input to both channels of an \
682 amplifier, then summing them into one, thereby effectively doubling the amplifier po\
683 wer supplied to the signal."),
684 quiz::Quiz("Brownian Noise","Also referred to as brown noise, technically it’s a typ\
685 e of noise whose power density (spectral loudness) decreases 6 dB per octave with in\
686 creasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a di\
687 stance. It can also be used a slowly changing random control voltage or modulation s\
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688 ignal, instead of as an audio source."),
689 quiz::Quiz("Buchla Bongos","This is a classic patch where a complex sound source - s\
690 uch as one oscillator frequency modulating another - is sent through a Low Pass Gate\
691 with either just a trigger to "strike" the vactrol inside or otherwise an instant a\
692 ttack/fast decay envelope to create a nice percussive sound. The fact that the low p\
693 ass gate reduces the higher harmonics as its volume dies away helps tame the harmoni\
694 cs coming from the complex source, and give it a decay similar to a struck percussiv\
695 e instrument."),
696 quiz::Quiz("Bucket Brigade Delay","An early design for an echo or delay effect where\
697 the input audio would be sampled as an analog voltage, and held for a brief moment.\
698 Then at the next above-audio sample rate clock pulse, this voltage would get passed\
699 to the next sample and hold (bucket) in the circuit, while a new level was sampled.\
700 Bucket brigade delays (BBDs) usually have numbers of stages or buckets that are pow\
701 ers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the numbe\
702 r of stages multiplied by the time interval between samples."),
703 quiz::Quiz("Bucking","A type of phase cancellation in which two identical signals or\
704 frequencies, having the same amplitude but opposite polarity, cancel one another ou\
705 t. Most commonly used in the context of musical instrument frequencies. Example: a "\
706 Humbucker" guitar pickup is designed to remove or "buck" hum frequencies from the si\
707 gnal using this principle."),
708 quiz::Quiz("Buffered Multiple","Quite often you need to split or copy a signal to se\
709 nd to more than one destination. This is commonly done with a multiple, where you pl\
710 ug one source in, and then plug in additional patch cables to go off to multiple des\
711 tinations. An active or buffered multiple is one that includes a buffer circuit betw\
712 een the input and output, making sure the signal does not lose its strength or integ\
713 rity by being split too many times, and that no funny business happening on one of t\
714 he outputs affects any of the other connections. Some modules have good buffering bu\
715 ilt into their outputs, and can drive multiple modules without issue. But if you try\
716 to use a passive mult to connect to, say, three oscillators, and you realize the tr\
717 acking isn't very good (they quickly go out of tune as you go up and down the scale)\
718 , then you need a buffered mult instead."),
719 quiz::Quiz("Bulk Dump","Short for System Exclusive Bulk Dump, a method of transmitti\
720 ng data such as the internal parameters between MIDI devices."),
721 quiz::Quiz("Burst Generator","When you send this module a trigger, it outputs a stre\
722 am or "burst" of triggers in response. You usually have control over the number of t\
723 riggers, the spacing between them, and often the probability that individual trigger\
724 output will be sent or skipped (for random patterns). At its most tame, it can be u\
725 se to create "double pluck" triggers in response to a normal note on; and its most e\
726 xtreme, it is used to trigger a high-energy, chaotic stream of drum hits that may or\
727 may not be in time with the music."),
728 quiz::Quiz("Bus Board","This simple circuit board takes the output of your modular s\
729 ystem's power supply and creates multiple copies of it, routed to connectors that go\
730 to your individual modules."),
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731 quiz::Quiz("Bus","An audio pathway by which one or more signals, usually from differ\
732 ent sources, are routed to a designated place. Because busses are highly connected t\
733 o signal flow, they serve a broad range of purposes in audio applications. 2) A shor\
734 thand term for the signals themselves that are routed through the bus (see also "Sub\
735 group")."),
736 quiz::Quiz("Byte","Information (data) bits in a grouping of eight. One byte = eight \
737 bits."),
738 quiz::Quiz("Cable Assembly","Cable that is ready for installation in specific applic\
739 ations and usually terminated with connectors."),
740 quiz::Quiz("Cable Harness","A grouping of cables or wires used to interconnect elect\
741 ronic systems."),
742 quiz::Quiz("Cable Sheath","Conductive protective cover that is applied to cables."),
743 quiz::Quiz("Cable","A group of one or more insulated conductors, optical fibers, or \
744 a combination of both within an enveloping jacket, typically for transmitting electr\
745 ical signals of different types."),
746 quiz::Quiz("Capacitor","An electronic device made of two plates separated by an insu\
747 lator, designed to store electrostatic energy. The capacitor is a key component in c\
748 ondenser microphones, for example."),
749 quiz::Quiz("Capstan","A mechanical part of a magnetic tape recorder that controls th\
750 e speed of the tape as it passes across the tape heads."),
751 quiz::Quiz("Capsule","Space-travel definitions aside, this is the name given to the \
752 part of a microphone that contains the diaphragm and active element, the mechanical \
753 structure that converts acoustic sound waves into electrical current."),
754 quiz::Quiz("Carbon Microphone","A microphone that uses carbon granules to convert so\
755 und waves to electrical impulses. The carbon element sits between two plates; as sou\
756 nd waves hit the carbon granules, it generates changes in resistance between the pla\
757 tes, affecting the electrical signal."),
758 quiz::Quiz("Cardioid Pattern","A microphone pickup pattern which is most sensitive t\
759 o sound coming from the front, less from the sides, and least from the back of the d\
760 iaphragm. So named because the pickup pattern is in the shape of a heart (cardio).")\
761 ,
762 quiz::Quiz("Carrier","There are a few different synthesis techniques where one usual\
763 ly audio-rate signal varies another audio signal. For example, in frequency modulati\
764 on, a second signal (called the modulator) varies the frequency (pitch) of the main \
765 signal, called the carrier. More specifics are described in the entries on frequency\
766 modulation and amplitude modulation."),
767 quiz::Quiz("Cascade","To connect or "daisy chain" two mixers so that the stereo mixi\
768 ng busses of the first mixer feed into the stereo busses of the second."),
769 quiz::Quiz("CCW","Counter-clockwise, usually in the context of rotating a control th\
770 e left (in the opposite direction of how a clock's hands move)."),
771 quiz::Quiz("CD","An abbreviation for Compact Disc, or a small optical disk with digi\
772 tal audio recorded on it."),
773 quiz::Quiz("Cent","When tuning instruments, a semitone is divided into 100 units cal\
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774 led cents; there are 1200 cents per octave (100 x 12 semitones). When one oscillator\  
775 is detuned compared to another, the difference in their frequencies is sometimes me\  
776 asured in cents."),  
777 quiz::Quiz("Center Frequency","The frequency of an audio signal that is most affecte\  
778 d by an equalizer, either boosting or attenuating the frequency. Drawn graphically, \  
779 this is the very top or bottom (the "peak") of the frequency bell-shaped curve."),  
780 quiz::Quiz("Channel Path","The complete signal path from the sound source to the mul\  
781 titrack recorder (or DAW). For example, an audio signal that travels from the microp\  
782 hone to the preamplifier, then into a channel strip on the mixing console, then is s\  
783 ent through the outputs into the recorder. This is different from the monitor path, \  
784 which feeds a mix of signals into monitor speakers or headphones without affecting t\  
785 he recorded signals. (See also "Monitor Path.")"),  
786 quiz::Quiz("Channel","1) An audio recording made on a portion of the width of a mult\  
787 itrack tape, or isolated within a digital audio workstation, usually for the purpose\  
788 of combining with other channels. 2) A single path that an audio signal travels or \  
789 can travel through a device from an input to an output."),  
790 quiz::Quiz("Chaotic","Believe it or not, chaotic does not mean completely random to \  
791 mathematicians. Chaos theory deals with systems that are random within certain bound\  
792 aries - such as the path of a wobbling wheel or the frequency of a dripping faucet. \  
793 Although they are not out of control, neither are they completely predictable. In sy\  
794 nthesis, a chaotic system usually refers to a modulation generator that is similar t\  
795 o a low frequency oscillator, but which has unpredictable wobbles or glitches in an \  
796 otherwise loosely or occasionally repetitive pattern. It can also refer to bursts of\  
797 triggers that do not follow musical divisions."),  
798 quiz::Quiz("Chase","The automatic adjusting of the speed of a recorder (or sequencer\  
799 ) to keep time with another recorder."),  
800 quiz::Quiz("Chord Chart","A shorthand form of musical notation that provides the bas\  
801 ic chord changes and essential rhythmic information of a song. Most commonly used by\  
802 studio session players, rhythm sections or jazz bands to provide the skeletal struc\  
803 ture of the song while allowing players room to create their own parts and improvise\  
804 . While lead sheets typically focus on melody line and chord structure, chord charts\  
805 display mainly chord changes and rhythm. (See also "Lead Sheet.")"),  
806 quiz::Quiz("Chord","Three or more musical pitches sung or played together."),  
807 quiz::Quiz("Chorus","1) The part of a song that is repeated with the same music and \  
808 lyrics each time, often containing the main point or hook of the song. 2) A musical \  
809 singing group with many singers. 3) A delay effect that simulates a vocal chorus by \  
810 adding several delays with a mild amount of feedback and a medium amount of depth.")\  
811 ,  
812 quiz::Quiz("Circuit","1) One complete path of electric current. 2) Similar to defini\  
813 tion 1, but including all audio signal paths and components to accomplish a particul\  
814 ar audio function."),  
815 quiz::Quiz("Class Compliant","This refers to a device that is \"plug and play\" - it\  
816 can be plugged directly into a computer or other host and immediately be recognized\
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817   without additional drivers needing to be installed. This comes up in the modular wo\
818   rld with MIDI to CV/Gate interfaces that use USB: If your converter is a USB Host, a\
819   nd you plug a class compliant USB Device such as a controller keyboard or fader pane\
820   l into it, the converter will recognize it."),
821   quiz::Quiz("Click Track","A metronome “click” fed into headphone monitors for the pu\
822   rpose of helping the musicians play in time with the song."),
823   quiz::Quiz("Clip","All active electronic circuits have a limit on how strong of a si\
824   gnal can pass through them. These limits are often associated with the positive and \
825   negative power supply levels. If the signal attempts to go beyond these limits, they\
826   instead get chopped or clipped off at that limit. For example, an input voltage of \
827   +12 volts may get through without alteration, but +13 volts at the input would come \
828   out as 12 volts. This clipping causes distortion in the waveform, usually adding hig\
829   her harmonics (such as a harsh buzz). Different circuits enter clipping in different\
830   ways – some may have a bit of rounding off before they reach that flat threshold; t\
831   his is referred to as soft clipping and is often desirable as it can be less harsh. \
832   Clipping is so named because the resulting graphic waveform looks like the edges of \
833   the waveform have been “clipped.”),
834   quiz::Quiz("Clock Signal","A signal sent by a device within the circuit that generat\
835   es steady pulses or codes to keep other devices in sync with each other. An example \
836   in the music world is sequencing via MIDI. The sequencer sends a clock signal so con\
837   nected devices will play in time."),
838   quiz::Quiz("Clock","Usually refers to the main rhythmic pulse in a system. Often, th\
839   e clock pulse is much faster than anything it might drive, such as a sequencer or LF\
840   O. The most common clock rate is 24 ppqn (pulses per quarter note), as is the case w\
841   ith MIDI clocks and DIN Sync. However, a trigger that drives a sequencer forward one\
842   note at a time may also be called the “clock” in a system. Indeed, there are module\
843   s that create divisions and multiplications of the main clock to generate new clock \
844   signals with a relationship to the main clock."),
845   quiz::Quiz("Clockwise","Clockwise, as in rotating a control the the right – in the s\
846   ame direction as a clock's hands move."),
847   quiz::Quiz("Close Miking","A microphone placement technique that places the mic clos\
848   e to the sound source to pick up the direct sound and reject ambient sound."),
849   quiz::Quiz("Coaxial Cable","(abbreviated “Coax”) A two-conductor cable that consists\
850   of one conductor surrounded by a shield."),
851   quiz::Quiz("Coincident Miking","A stereo miking technique in which two microphones a\
852   re placed with their heads as close to each other as possible. This prevents phase c\
853   ancillation problems in the mix because the distance from the sound to either microp\
854   hone is the same."),
855   quiz::Quiz("Compander","A signal processor serving as a combination compressor and e\
856   xpander, primarily used for noise reduction purposes in analog systems. The audio s\
857   ignal is compressed prior to recording, then expanded at the reproduction stage. Com\
858   panding is the principle behind Dolby noise reduction systems."),
859   quiz::Quiz("Comparator","An electrical device that compares the level of one voltage\
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860 to a second. That second voltage may be a second input on a comparator synth module\  
861 , or may be set with a knob or internal reference voltage. Most often, a comparator \  
862 outputs a gate signal that goes high when the first signal is higher than the second\  
863 (or vice versa), and which goes low when the first signal is lower than the second.\  
864 At audio rates, it converts an input waveform into a square or pulse wave, with the\  
865 second signal setting when the new waveform goes high or low in voltage."),  
866 quiz::Quiz("Comping","1) In digital audio workstations (DAWs), the process of blendi\  
867 ng portions of multiple recorded takes to create a “compilation” track. (See also “T\  
868 ake,” “Playlist.”) 2) In jazz music performance, an abbreviation for “accompanying.”\  
869 "),  
870 quiz::Quiz("Complex Oscillator","This module typically has a pair of oscillators beh\  
871 ind one panel that is prewired where one oscillator modulates the other’s frequency \  
872 (known as Frequency Modulation or FM synthesis); some also allow you to quickly swit\  
873 ch them so that the first modulates the amplitude of the second, or some other varia\  
874 tion. They may also have waveshapers built in. They are based on a popular module cr\  
875 eated by Buchla, which is a standard of the “West Coast” approach to synthesis."),  
876 quiz::Quiz("Compression Driver","A diaphragm that feeds a sound pressure wave into a\  
877 horn loudspeaker."),  
878 quiz::Quiz("Compression Ratio","The rate by which a compressor attenuates an incomin\  
879 g signal, measured in decibels. For example, a compression ratio of 4:1 means the co\  
880 mpressor will only allow a 1 dB increase in the signal for every 4 dB increase in th\  
881 e signal above the threshold."),  
882 quiz::Quiz("Compression","1) In signal processing, the action performed by a compres\  
883 sor (see also “Compressor”). 2) In acoustics, the increased air pressure caused by t\  
884 he peak of a sound pressure wave, used in the context of “compression and rarefactio\  
885 n” (see also “Rarefaction”)."),  
886 quiz::Quiz("Compressor","A signal processor or plug-in that reduces the dynamic rang\  
887 e of an audio signal by amplifying its quieter sections and attenuating its louder o\  
888 nes."),  
889 quiz::Quiz("Condenser Microphone","A microphone in which sound is converted into ele\  
890 ctrical current through changes in a capacitor. The sound pressure waves move the di\  
891 aphragm, producing changes in capacitance which are then changed into electrical vol\  
892 tage."),  
893 quiz::Quiz("Contact Microphone","A microphone designed to pick up vibrations from so\  
894 lid objects (as opposed to vibrations in the air). Also known as a “pickup” or “piez\  
895 o,” this microphone is often used as an acoustic guitar pickup to pick up the vibrat\  
896 ions from the soundboard, or by experimental musicians creating “noise music” from a\  
897 variety of objects."),  
898 quiz::Quiz("Control Voltage Processor","CVP is the abbreviation for a module that al\  
899 lows processing of the voltage going through it – such as amplifying or attenuating \  
900 it, offsetting it in a positive or negative direction, introducing slew (slurring of\  
901 changes in voltage), and possibly other functions such as deriving a gate signal fr\  
902 om an incoming voltage by running it through a comparator. Make Noise’s Maths is per\  

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903 haps the most well known control voltage processor out there; you will also find som\
904 e modules with CVP specifically in their name. Regardless, it's good to have one or \
905 more of this type of module in your system to help massage voltages to get them to d\
906 o what you want (or to teach them new tricks)."),
907 quiz::Quiz("Control Voltage","The concept of control voltage (CV) is at the very roo\
908 t of modular synthesizer. The general idea is that analog voltage levels are used co\
909 ntrol functions and parameters of a module. For example, one control voltage may det\
910 ermine the pitch played by an oscillator; a second control voltage may determine how\
911 loud that signal is after it's passed through a voltage-controlled amplifier. CV is\
912 the most common shorthand to refer to control voltage – for example, when a synthes\
913 izer module says it features “CV over the filter’s resonance,” that means there is a\
914 control voltage input to control the amount of resonance (feedback) – not just the \
915 customary knob on the front panel."),
916 quiz::Quiz("Controller","In the broadest sense, a controller is any device that is u\
917 sed to control another device. Most commonly used in the context of MIDI controllers\
918 , which send out MIDI signals to control other connected MIDI instruments and device\
919 s. Other examples of controllers in the recording studio can include monitor control\
920 lers, DAW controllers and DJ controllers."),
921 quiz::Quiz("Corner Frequency","The cutoff or corner frequency of a filter is the poi\
922 nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
923 uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
924 00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
925 tered” – reduced in loudness – the further above 500Hz you go."),
926 quiz::Quiz("Counter Clockwise","Counter-clockwise, usually in the context of rotatin\
927 g a control the left (in the opposite direction of how a clock's hands move)."),
928 quiz::Quiz("CPU","Abbreviation for Central Processing Unit, the main “brain” chip in\
929 a computer (also known simply as “Processor”)."),
930 quiz::Quiz("Critical Distance","The distance from the sound source at which the dire\
931 ct sound and the reverberant sound are at equal volume. Critical distance varies acc\
932 ording to the space; in a room with absorbent walls, the critical distance will be f\
933 urther from the source, and in a reverberant room, the distance will be closer to th\
934 e source."),
935 quiz::Quiz("Crossfade","An audio editing technique in which one sound is faded out a\
936 s another sound is faded in, to create a seamless transition between the two. Audio \
937 engineers use crossfading, for example, to blend two takes or more “takes” of a reco\
938 rded track into a composite take. Club DJs also use crossfading to transition from o\
939 ne song to the next with no stops."),
940 quiz::Quiz("Crossover Frequency","The frequency at which the crossover stops sending\
941 the signal to one speaker and starts sending it to another."),
942 quiz::Quiz("Crossover","An audio filter component that splits an audio signal into t\
943 wo or more bands or signals, usually to be fed into different components of a loudsp\
944 eaker system according to frequency range. (Also called a “crossover network.”)"),
945 quiz::Quiz("Crosstalk","The unwanted leakage of an audio signal between two audio ch\
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946 annels—for example, overlapping signals between channels on a mixing console, or ove\
947 rlapping audio between two tracks of audiotape."),
948 quiz::Quiz("Cue","In general terms, a cue is the starting point for a piece of music\
949 or section of music. Depending on the context, the word “cue” may describe: 1) The \
950 point at which a musician or vocalist is supposed to start playing or singing; 2) Th\
951 e audio fed to the musicians through headphones so they can determine when to start \
952 playing/singing; 3) A specific location point on the music timeline within a DAW or \
953 on the tape; or 4) To set the tape or disc to a certain starting point in the song (\
954 “cueing” the tape). A cue can even refer to an entire section of music being used fo\
955 r video production."),
956 quiz::Quiz("Cutoff Frequency","The cutoff or corner frequency of a filter is the poi\
957 nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
958 uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
959 00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
960 tered” – reduced in loudness – the further above 500Hz you go."),
961 quiz::Quiz("Cutoff Slope","The rate of reduction of the frequencies beyond the passb\
962 and of a filter. The slope is described as the number of dB the filter reduces the s\
963 ignal for each octave past the cutoff frequency."),
964 quiz::Quiz("CV/Gate","This is the shorthand to say a synthesizer may be controlled b\
965 y voltages – usually for pitch – and gate signals to indicate when a note is “on.” A\
966 n increasing number of controller keyboards are including CV/Gate output in addition\
967 to the customary MIDI (Musical Instrument Digital Interface), making them much easi\
968 er to connect to a modular synthesizer, as no additional MIDI to CV interface is req\
969 uired."),
970 quiz::Quiz("CV","The concept of control voltage (CV) is at the very root of modular \
971 synthesizer. The general idea is that analog voltage levels are used control functio\
972 ns and parameters of a module. For example, one control voltage may determine the pi\
973 tch played by an oscillator; a second control voltage may determine how loud that si\
974 gnal is after it’s passed through a voltage-controlled amplifier. CV is the most com\
975 mon shorthand to refer to control voltage – for example, when a synthesizer module s\
976 ays it features “CV over the filter’s resonance,” that means there is a control volt\
977 age input to control the amount of resonance (feedback) – not just the customary kno\
978 b on the front panel."),
979 quiz::Quiz("CVP","CVP is the abbreviation for a module that allows processing of the\
980 voltage going through it – such as amplifying or attenuating it, offsetting it in a\
981 positive or negative direction, introducing slew (slurring of changes in voltage), \
982 and possibly other functions such as deriving a gate signal from an incoming voltage\
983 by running it through a comparator. Make Noise’s Maths is perhaps the most well kno\
984 wn control voltage processor out there; you will also find some modules with CVP spe\
985 cifically in their name. Regardless, it’s good to have one or more of this type of m\
986 odule in your system to help massage voltages to get them to do what you want (or to\
987 teach them new tricks)."),
988 quiz::Quiz("CW","Clockwise, as in rotating a control the the right – in the same dir\
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989  action as a clock's hands move."),
990  quiz::Quiz("Cycle","One complete expression of a waveform beginning at a certain poi\
991  nt, progressing through the zero line to the wave's highest and lowest points, and r\
992  eturning to the same value as the starting point. One complete vibration or sound wa\
993  ve."),
994  quiz::Quiz("D-Sub Connector","Abbreviation for “D-subminiature connector,” a D-sub i\
995  s a multipin connector that is most often used to connect a computer to a VGA monito\
996  r, but also used occasionally in digital audio applications in the recording studio.\
997  "),
998  quiz::Quiz("D/A","Abbreviation for Digital to Analog conversion, which changes digit\
999  al data numbers (digital audio signal) into discrete voltage level. The reverse proc\
1000  ess of A/D. Also known as DAC."),
1001  quiz::Quiz("DADSR","This is a slightly fancier take on the standard ADSR envelope ge\
1002  nerator that introduces an initial timed delay before the initial attack stage (risi\
1003  ng from 0 to a peak level) begins. One patch idea is to route this type of envelope \
1004  to a low pass filter cutoff, so there's initially a muted, filtered sound when the n\
1005  ote starts, and then after a pause it starts to swell into a brighter, fuller sound.\
1006  "),
1007  quiz::Quiz("Daisy Chain","The connection of three or more devices in a series, where\
1008  the audio signal passes through one device to reach a second, and through the secon\
1009  d to reach the third, etc."),
1010  quiz::Quiz("Damping Factor","Describes an amplifier's ability to restrain the pushba\
1011  ck motion (back-EMF) of the loudspeaker cone when the audio signal stops."),
1012  quiz::Quiz("Damping","The reduction of energy in a vibrating system, through frictio\
1013  n. Can refer to the reduced amplitude in an electrical signal, or the stifled vibrat\
1014  ions of a musical instrument (for example, the damper pedal on an acoustic piano).")\
1015  ,
1016  quiz::Quiz("DAW","An abbreviation for Digital Audio Workstation, a device or softwar\
1017  e program designed for recording and mixing audio digitally."),
1018  quiz::Quiz("dB","An abbreviation for decibel, a measurement ratio that compares sign\
1019  al strengths (usually audio levels)."),
1020  quiz::Quiz("DBX","A series of noise reduction systems, named for the company that de\
1021  veloped them. DBX noise reduction has been less commercially successful than the mor\
1022  e widely known Dolby systems, but is still found on occasion in recording studios.")\
1023  ,
1024  quiz::Quiz("DC Coupled","When a module says its inputs are DC Coupled, that means it\
1025  can accept DC voltages (constant or slowly changing voltages) and pass them through\
1026  unaltered. This is important if, for example, you want to use a VCA to control the \
1027  amplitude of an envelope going through it: You would need one that was DC coupled, a\
1028  s an AC coupled input would try to remove the DC component of the signal (such as it\
1029  s sustain level) and return it to 0v."),
1030  quiz::Quiz("DC","Electrical current that flows in a single direction, as opposed to \
1031  Alternating Current (AC), which flows in alternating directions. Many electronic dev\
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1032 ices run on DC, which is usually provided by battery power, USB power or an AC adapt\
1033 er plugged into the wall. In modular terms, DC refers to a voltage that tends to sta\
1034 y at one steady level for awhile, such as a gate output that switches between 0v whe\
1035 n a note is off and 5 or 10v when a note is on. It can also refer to a slowly changi\
1036 ng voltage, such as an envelope."),
1037 quiz::Quiz("DCO","A DCO (Digitally Controlled Oscillator) is a hybrid design for an \
1038 analog oscillator that – instead of using a voltage level to determine the pitch of \
1039 the oscillator – uses a digital device such as a counter to determine the length of \
1040 each waveform cycle and therefore the pitch. On the plus side, tuning is very stable\
1041 , unlike some all-analog designs. On the minus side, there are no imperfections in p\
1042 itch that cause subtle detuning (and therefore the perception of “fatness”) when usi\
1043 ng more than oscillator per voice."),
1044 quiz::Quiz("De-esser","An audio compressor designed to reduce the volume of sibilant\
1045 sounds and frequencies, especially those produced by pronouncing the letter “s.”),
1046 quiz::Quiz("Decay","In general, decay refers to a voltage or overall level dropping \
1047 down from some high point, such as the decay stage of an envelope generator. A real-\
1048 world analogy is that after you initially strike a drum or pluck a string, it decays\
1049 in volume from its initial loudness eventually all the way to silence. It can also \
1050 refer to the tail of a reverb or echo effect where the sound dies away over time."),
1051 quiz::Quiz("Decca Tree","A stereo microphone placement technique involving three mic\
1052 rophones (usually omnidirectional) placed in a “T” pattern. Commonly used in miking \
1053 choirs, orchestras and other large ensembles, but variations of the Decca tree techn\
1054 ique are also being used today in surround sound situations."),
1055 quiz::Quiz("Decibel","(abbreviated “dB“) The ratio measurement of two levels accordi\
1056 ng to a scale where a certain percentage change comprises one unit. Most often used \
1057 to describe audio levels."),
1058 quiz::Quiz("Degaussing","The process of demagnetizing an object. In the context of a\
1059 udio, degaussing essentially erases the recording on magnetic tape."),
1060 quiz::Quiz("Delay/Attack/Decay/Sustain/Release","This is a slightly fancier take on \
1061 the standard ADSR envelope generator that introduces an initial timed delay before t\
1062 he initial attack stage (rising from 0 to a peak level) begins. One patch idea is to\
1063 route this type of envelope to a low pass filter cutoff, so there’s initially a mut\
1064 ed, filtered sound when the note starts, and then after a pause it starts to swell i\
1065 nto a brighter, fuller sound."),
1066 quiz::Quiz("Delay","You all know what the word delay means in the normal world; it c\
1067 an appear in different forms inside a modular synth. For example, it can refer to th\
1068 e spacing between repeats in an echo; that’s why an echo device is often known as a \
1069 “delay” effect. It can also refer to a programmable amount of time you delay a signa\
1070 l, such as a gate, trigger, or initial stage of an envelope so a note would start la\
1071 ter than it was actually played. Also, 1) An process by which an audio signal is rec\
1072 orded to a medium or device, reproduced at a time delay, then mixed with the origina\
1073 l, non-delayed signal to create a variety of effects such as a fuller sound, echo, c\
1074 horusing, flanging, etc. 2) A signal processor that creates delay effects."),
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1075 quiz::Quiz("Demo","A preliminary recording that is intended to give the listener an \
1076 idea of how a song could sound in a final production. A demo usually involves minima\
1077 l tracking or production, almost like a “rough draft” of a recording."),
1078 quiz::Quiz("Detune","If you have two oscillators tuned to exactly the same frequency\
1079 - and I mean, exactly the same frequency - there’s not much point in having more th\
1080 an one oscillator. However, when you change the tuning of one ever so slightly - in \
1081 other words, detune it - you will start to hear interesting interactions between the\
1082 two, often referred to as chorusing or beating. The result tends to be more interes\
1083 ting and “full” - and a bit more natural, as two singers or instruments can rarely h\
1084 it exactly the same note. To purposely cause an instrument or signal to play out of \
1085 tune (usually slightly). This effect can be used for a number of purposes in the stu\
1086 dio, but is often used in “double-tracking,” blending the detuned instrument/track w\
1087 ith the original to create a fuller sound."),
1088 quiz::Quiz("DI","The process of sending an electrical audio signal directly from an \
1089 instrument to the mixing console through the use of electric pickups or direct boxes\
1090 , as opposed to using a microphone."),
1091 quiz::Quiz("Dialogue","The spoken word recorded in film/video sound, commercials and\
1092 instructional recordings."),
1093 quiz::Quiz("Diaphragm","The part of a microphone that moves in response to sound wav\
1094 es, converting them to electrical signals."),
1095 quiz::Quiz("Difference","A fancy way of saying you subtracted on control voltage fro\
1096 m another. It can also be applied to audio or harmonics."),
1097 quiz::Quiz("Digital Audio Workstation","abbreviated DAW) A device or computer softwa\
1098 re that records and mixes audio digitally and creates digital audio files. A DAW can\
1099 be a standalone unit or an integrated set of components, but today they are most co\
1100 mmonly found as “in-the-box” software programs run from a computer. The most common \
1101 DAW program found in recording studios is Pro Tools; other commonly used programs in\
1102 clude Reason, Ableton and Logic."),
1103 quiz::Quiz("Digital Multimeter","A small device that tests electrical voltage, curre\
1104 nt, and resistance. Multimeters are useful in recording studios for calibrating elec\
1105 trical systems and troubleshooting problems."),
1106 quiz::Quiz("Digital Recording","The process of converting audio signals into numbers\
1107 that represent the waveform, then storing these numbers as data."),
1108 quiz::Quiz("Digital Signal Processing","(abbreviated “DSP”) Any signal processing do\
1109 ne after an analog audio signal has been converted into digital audio."),
1110 quiz::Quiz("Digital to Analog Converter","(abbreviated D/A) A device that converts t\
1111 he digital data of digital audio into voltage levels that approximate the original a\
1112 nalog audio."),
1113 quiz::Quiz("Digital","There was a time when digital (referring to circuitry based ar\
1114 ound binary logic, computers, and the such compared to the old-fashioned transistors\
1115 , op amps, capacitors, and other bits that make up analog circuitry) was a dirty wor\
1116 ld among synthesists. The assumption was digital techniques created sounds that were \
1117 more sterile, brittle, and abrasive - and just not as “authentic.” Today, digital ci\
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1118 rcuitry is embraced in synthesizers, including modular systems. Although analog will\  
1119 always hold a special place in our hearts, a well-implemented digital circuit can s\  
1120 ound just as good as an analog one, while digital signal processing and programming \  
1121 can create a wider range of sounds than most analog circuitry."),  
1122 quiz::Quiz("Digitally Controlled Oscillator","A DCO (Digitally Controlled Oscillator\  
1123 ) is a hybrid design for an analog oscillator that – instead of using a voltage leve\  
1124 l to determine the pitch of the oscillator – uses a digital device such as a counter\  
1125 to determine the length of each waveform cycle and therefore the pitch. On the plus\  
1126 side, tuning is very stable, unlike some all-analog designs. On the minus side, the\  
1127 re are no imperfections in pitch that cause subtle detuning (and therefore the perce\  
1128 ption of “fatness”) when using more than oscillator per voice."),  
1129 quiz::Quiz("DIN Stereo","A stereo microphone placement technique that places two car\  
1130 dioid microphones about 20cm apart and set outward from each other at a 90-degree an\  
1131 gle to create a stereo image. Particularly for stereo miking at close ranges. (See \  
1132 also “Near-Coincident Miking.”)"),  
1133 quiz::Quiz("DIN Sync","A clock signal for controlling the tempo of sequencers, arpeg\  
1134 giators, and drum machines, distributed using cables with DIN-style connectors (yes,\  
1135 just like old-fashioned MIDI connectors, but DIN Sync is even older). Roland pionee\  
1136 red this standard, which included sending 24 pulses per quarter note (PPQN), giving \  
1137 rise to the alternate name Sync24. Korg equipment used a variation of this running a\  
1138 t 48 pulses per quarter note, also known as Sync48. DIN Sync is still a popular way \  
1139 of sending a clock signal to a modular synth today, especially when interfacing with\  
1140 other vintage synthesizers, sequencers, and drum machines."),  
1141 quiz::Quiz("Diode Ladder Filter","This is a filter design most often associated with\  
1142 the Roland TB-303 Bass Line, which is known for its rubbery sound with eager resona\  
1143 nce."),  
1144 quiz::Quiz("Diode","An electrical component that enables easy electrical current flo\  
1145 w in one direction but not the other. In the recording studio, these are commonly fo\  
1146 und in the vacuum tubes of tube amplifiers."),  
1147 quiz::Quiz("Direct Box","A small device that to converts an unbalanced, high-impedan\  
1148 ce speaker or instrument-level output to a balanced, low-impedance mic-level output.\  
1149 Frequently used in the signal path connecting electric instruments “directly” to th\  
1150 e mixing console, as opposed to miking them acoustically. Also called “direct inject\  
1151 ion box” or “DI box.”"),  
1152 quiz::Quiz("Direct Current","In modular terms, DC refers to a voltage that tends to \  
1153 stay at one steady level for awhile, such as a gate output that switches between 0v \  
1154 when a note is off and 5 or 10v when a note is on. It can also refer to a slowly cha\  
1155 nging voltage, such as an envelope. (abbreviated “DC”) Electrical current that flows\  
1156 in a single direction, as opposed to Alternating Current (AC), which flows in alter\  
1157 nating directions. Many electronic devices run on DC, which is usually provided by b\  
1158 attery power, USB power or an AC adapter plugged into the wall."),  
1159 quiz::Quiz("Direct Injection","(abbreviated “DI”) The process of sending an electric\  
1160 al audio signal directly from an instrument to the mixing console through the use of\  

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1161 electric pickups or direct boxes, as opposed to using a microphone."),
1162 quiz::Quiz("Direct Out","An output available on some consoles which is fed directly \
1163 from the preamplifier stage of the input, bypassing the channel strips and faders. T\
1164 his feature is often used to send a “dry” signal to a monitor mix or a recording dev\
1165 ice."),
1166 quiz::Quiz("Direct Sound","The sound that reaches a microphone or a listener’s ear w\
1167 ithout hitting or bouncing off any obstacles (as opposed to reflected or ambient sou\
1168 nd)."),
1169 quiz::Quiz("Directional Pattern","1) In microphones, a term meaning the same thing a\
1170 s “Pick Up Pattern,” a description of the area in which a microphone is most sensiti\
1171 ve to sounds. 2) In loudspeakers, it is the pattern of dispersion, the area that the\
1172 sound from a speaker will evenly cover in a listening area."),
1173 quiz::Quiz("Dispersion (also Dispersion Angle)","The area that is effectively cover\
1174 ed by the sound coming from a loudspeaker; specifically, the imaginary boundaries on \
1175 either side of the speaker at which the sound level is 6 dB lower than if you were s\
1176 tanding directly in front of the speaker. Each speaker has both a horizontal and ver\
1177 tical dispersion angle."),
1178 quiz::Quiz("Distant Miking","The technique of placing a microphone far from the soun\
1179 d source in order to pick up a combination of the direct and reflected sounds."),
1180 quiz::Quiz("Distortion","Refers to the deforming of a waveform at the output of a de\
1181 vice as compared with the input, usually due to overload, creating a distorted or “d\
1182 irty” signal. While electrical or audio distortion is typically unwanted and avoided\
1183 , it is frequently used in controlled situations in audio to create certain desirabl\
1184 e effects, particularly with electric guitars and amplifiers."),
1185 quiz::Quiz("Diversity","1) In audio settings: the use of two or more antennas in a w\
1186 ireless receiver system to prevent dropouts in the audio from a wireless microphone.\
1187 2) In other settings: the embracing of the uniqueness of all individuals."),
1188 quiz::Quiz("Dolby","The brand name of a manufacturer of noise reduction systems and \
1189 other audio systems, to improve performance and fidelity of audio recording, playbac\
1190 k, and transmission."),
1191 quiz::Quiz("Doppler Effect","The phenomenon in which the human ear perceives a chang\
1192 e in the frequency (pitch) of a sound while the sound source is in motion. As the so\
1193 und source approaches, the sound waves travel a shorter distance to the ear, increas\
1194 ing the frequency of the waves and the pitch of the sound; as the sound source moves\
1195 away, the sound waves must travel farther and farther, resulting in lower frequenci\
1196 es. A common example of this effect is an approaching emergency vehicle whose siren \
1197 sounds higher as it approaches and lower after it passes. The Doppler Effect can be \
1198 utilized in audio settings, for example, in the Leslie speaker in which an electric\
1199 motor rotates the speakers inside the cabinet, constantly changing the distance bet\
1200 ween the sound source and the listener (or microphone) and creating its signature wa\
1201 rbling vibrato effect."),
1202 quiz::Quiz("Double","1) To record a second performance closely matching the first pe\
1203 rformance, for the purpose of blending the two tracks. 2) To use a delay line with m\
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1204 edium delay to simulate double tracking."),
1205 quiz::Quiz("Driver","1) A transducer in a loudspeaker that converts electrical signal\
1206 ls into sound pressure waves. 2) A computer program that controls an attached device\
1207 or piece of hardware."),
1208 quiz::Quiz("Dropout","A brief loss of audio signal on tape, or a brief loss of data \
1209 in a digital audio file (often due to a dropped sample), that can result in an unwan\
1210 ted dip in audio, a crackle or a pop."),
1211 quiz::Quiz("Drum Machine","An electronic device containing synthesized and/or sample\
1212 d drum sounds in its memory, along with an internal sequencer that can be programmed\
1213 to play drum patterns or loops."),
1214 quiz::Quiz("Drum Pattern","A specific sequence of drum sounds played by a drummer or\
1215 sequenced into a drum machine for use in a song."),
1216 quiz::Quiz("Dry","A sound with no effects is referred to as \"dry\"; a sound with ef\
1217 fects (such as reverb) mixed is referred to as \"wet.\" Effects units or mixers ofte\
1218 n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
1219 d and the fully-effected sound."),
1220 quiz::Quiz("DSP","Any signal processing done after an analog audio signal has been c\
1221 onverted into digital audio."),
1222 quiz::Quiz("Dub (or Dubbing)","1) To copy a recording. 2) To record in real time wit\
1223 h another recording with the intent of mixing the two recordings (see also “Overdub/\
1224 Overdubbing”). 3) “Dub” is an abbreviation for “dubstep,” a style or subgenre of ele\
1225 ctronic music."),
1226 quiz::Quiz("Ducking","A compression-based audio effect in which an audio signal is r\
1227 educed proportionately by the presence of another audio signal, sometimes accomplish\
1228 ed through a “sidechain” connection with the signal processor. A notable example is \
1229 a spoken-word voice-over track recorded over a musical track, where the music drops \
1230 in volume when the speaker begins to speak. A more subtle example is when an audio e\
1231 ngineer “ducks” specific sounds to make room for others in the track; for example, w\
1232 hen a bass guitar signal triggers a slight reduction in the level of drums or guitar\
1233 s. (See also “Sidechain.”)"),
1234 quiz::Quiz("Duophonic","Duophonic means two \"voices.\" Most early synths (including\
1235 modular systems) are monophonic, which means they can play only one note at a time;\
1236 some instruments have enough oscillators, filters, envelopes, and amplifiers that t\
1237 hey could play two separate notes as once. Some MIDI interfaces for modular synths i\
1238 nclude duophonic modes so you can patch up and control two separate voices from your\
1239 keyboard. Some users play fast and loose with terms such as duophonic, monophonic, \
1240 and polyphonic;"),
1241 quiz::Quiz("Duration","Duration is another way of saying length. A clock pulse or a \
1242 gate signal that is “high” for a certain amount of time – say, 100 msec – is said to\
1243 have a duration of 100 msec. The length of time you hold a note down, or the length\
1244 of a step in a sequence, is also called its duration."),
1245 quiz::Quiz("Dynamic Microphone","(Also called Moving Coil Microphone) A microphone i\
1246 n which sound pressure waves are converted to an electrical audio signal by an induc\
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1247 tion coil moving within a magnetic field—a process often compared to a loudspeaker w\
1248 orking in reverse. Dynamic microphones are less sensitive than condenser microphones\
1249 , but can be effective for miking louder sound sources or for close-miking applicati\
1250 ons."),
1251 quiz::Quiz("Dynamic Processing/Dynamic Signal Processing","The process of automatica\
1252 lly changing the level (or gain) to alter the level relationship of the loudest audi\
1253 o to the softest audio. Dynamic processors include compressors, limiters, expanders \
1254 and gates."),
1255 quiz::Quiz("Dynamic Range","1) The ratio (in dB) between the loudest peak and the so\
1256 ftest level of a song or recording. 2) The ratio (in dB) between the softest and lou\
1257 dest possible levels a device or system can provide without distortion."),
1258 quiz::Quiz("Early Reflections","The first sound waves that reach a listener's ear af\
1259 ter bouncing off a surface in the room, usually heard almost immediately after the i\
1260 nitial sound. The first stage of reverberation."),
1261 quiz::Quiz("East Coast Synthesis","This blanket term is applied to most common synth\
1262 esizer configuration pioneered by East Coast based companies such as Moog, Arp, and \
1263 EML (as well as "Far East" companies such as Roland and Korg) where one or more osci\
1264 llators producing waveforms with rich harmonic content (such as a sawtooth or square\
1265 wave) are fed into a filter that removes some of those harmonics, and then onto an \
1266 amplifier to shape the loudness of a note. This approach is also often known as subt\
1267 ractive synthesis, as the filter reduces (subtracts) harmonics that came from the os\
1268 cillators. East Coast synthesizers also regularly have organ-style black & white key\
1269 boards, and four stage ADSR type envelopes. Today it's common to mix both East Coast\
1270 and West Coast approaches in the same system."),
1271 quiz::Quiz("Echo Chamber","An enclosed room designed with reflective, non-parallel s\
1272 urfaces for the purpose of creating acoustic echoes (reverberation)."),
1273 quiz::Quiz("Echo","The distinct repetition of an initial sound, caused by the reflec\
1274 tion of the sound waves upon a surface. We recognize a sound as an echo when the dis\
1275 tance between the source and the reflection is far enough apart that we can detect t\
1276 he time delay between one and the other. Essentially, reverberation is the combinati\
1277 on of many echoes occurring too rapidly to hear each individually. In the studio, ec\
1278 hoes can be reproduced acoustically or simulated by a digital signal processor."),
1279 quiz::Quiz("Edit","To change one or more parameters of a recorded sound after the fa\
1280 ct. This can take many forms, including "punching in" a section of the music that is\
1281 re-recorded to replace the original version; altering the shape/size of waveforms g\
1282 raphically; changing the sequence of playback; and many others. Analog editing would\
1283 typically involve splicing the magnetic tape on which the audio signals were record\
1284 ed. These days, almost all editing in the studio is done via computer using a digita\
1285 l audio workstation (DAW)."),
1286 quiz::Quiz("Effect Loop","Sometimes you might want to send a signal outside your mod\
1287 ular system, process it through an external effects device, and bring it back into y\
1288 our modular for more processing. This going out/coming back in is referred to as an \
1289 effect loop. The trick with modular synths is that their internal signal levels tend\
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1290 to be much higher than those used by external effect equipment, so a modular effect\  
1291 loop will usually have level matching circuitry as well."),  
1292 quiz::Quiz("Effects Processor","(Also called Guitar Processor) A device that adds au\  
1293 dio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, o\  
1294 verdrive, amplifier simulation, etc. Effects processors can occur as individual effe\  
1295 cts boxes or multi-sound pedal boards (see also "Foot Pedals," "Foot Switches") adde\  
1296 d into the signal path between the guitar and the console. They can also be found as\  
1297 presets in guitar amplifiers, or even as digital plug-ins within a DAW."),  
1298 quiz::Quiz("Effects Track","1) In film production audio, a recording of the mixdown \  
1299 of all the sound effects ready to be mixed with the dialogue and music. 2) In music \  
1300 recording, one track with a recording of effects to be added to another track of a m\  
1301 ultitrack recording."),  
1302 quiz::Quiz("Effects","1) Various ways an audio signal can be modified by adding some\  
1303 thing to the signal to change the sound. 2) Short for the term Sound Effects (sounds\  
1304 other than dialogue, narration or music like door closings, wind, etc.) added to fi\  
1305 lm or video."),  
1306 quiz::Quiz("EG","The envelope generator (EG) module is used to shape the loudness or\  
1307 dynamics of a note when connected to a VCA (Voltage Controlled Amplifier), as well \  
1308 as how its frequency content or timbre changes over time when connected to a VCF (Vo\  
1309 ltage Controlled Filter). To do this, an envelope generator creates a voltage that \  
1310 typically rises from zero volts to some maximum level, and back down again. You cont\  
1311 rol how long this takes, usually in various stages: an attack stage as it goes from \  
1312 zero to max, a decay stage as it falls back down from maximum to either zero (in the\  
1313 case of an AD, or Attack/Decay envelope) or an intermediate level known as the sust\  
1314 ain, and then (usually after a key has been released and the corresponding gate sign\  
1315 al has gone back to zero) from the sustain level back to zero over a duration known \  
1316 as its release."),  
1317 quiz::Quiz("Electret Microphone","A variation of condenser microphone that uses an e\  
1318 lectret instead of a capacitor. (Also called "Electret Condenser Microphone.") Becau\  
1319 se the electret is permanently polarized, an electret microphone does not require an\  
1320 external power source as a standard condenser microphone does."),  
1321 quiz::Quiz("Electret","A dielectric plate that is designed with permanent polarity, \  
1322 allowing it to function similarly to a magnet. ("Electret" comes from the words "ele\  
1323 ctricity" and "magnet.") Used in some microphone types in place of a capacitor (cond\  
1324 ensen)."),  
1325 quiz::Quiz("Electromagnetic Field","(Abbreviated EMF) A field of magnetic energy put\  
1326 out because of current traveling through a conductor."),  
1327 quiz::Quiz("Electromagnetic Interference (EMI)","The bane of audio professionals eve\  
1328 rywhere, EMI is a type of interference caused by nearby electromagnetic activity, wh\  
1329 ich can be picked up by audio cables and equipment, causing unwanted noise, hum or b\  
1330 uzz in audio systems. Common causes of EMI in audio systems may include high-current\  
1331 power lines, fluorescent lighting, dimmer switches, computers, video monitors and r\  
1332 adio transmitters."),
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1333 quiz::Quiz("Electrons","Negatively charged particles revolving around the nucleus of\  
1334 an atom. Electrical current is generated by electrons moving along a conductor, lik\  
1335 e a metallic wire."),  
1336 quiz::Quiz("Emphasis","This word can have two meanings. In a normal audio context, i\  
1337 t usually means some form of high frequency boost, as emphasizing the higher harmoni\  
1338 cs can add clarity to a tone and help distinguish it from another. In synthesizers, \  
1339 emphasis usually means the Q or resonance setting on a filter, as increasing this se\  
1340 tting boosts (emphasizes) the harmonics at the cutoff or corner frequency."),  
1341 quiz::Quiz("Envelope Follower","This module follows the loudness contour of a sound,\  
1342 and outputs a voltage that corresponds to how that loudness changes. They tend to p\  
1343 erform some smoothing on this signal so that it's not too nervous or jumpy in nature\  
1344 . Envelope followers often also have a gate output that goes high when the loudness \  
1345 of the input signal went over a certain level, and low when it falls back below that\  
1346 level."),  
1347 quiz::Quiz("Envelope Generator","The envelope generator (EG) module is used to shape\  
1348 the loudness or dynamics of a note when connected to a VCA (Voltage Controlled Ampl\  
1349 ifier), as well as how its frequency content or timbre changes over time when connec\  
1350 ted to a VCF (Voltage Controlled Filter). To do this, and envelope generator creates\  
1351 a voltage that typically rises from zero volts to some maximum level, and back down\  
1352 again. You control how long this takes, usually in various stages: an attack stage \  
1353 as it goes from zero to max, a decay stage as it falls back down from maximum to eit\  
1354 her zero (in the case of an AD, or Attack/Decay envelope) or an intermediate level k\  
1355 nown as the sustain, and then (usually after a key has been released and the corresp\  
1356 onding gate signal has gone back to zero) from the sustain level back to zero over a\  
1357 duration known as its release."),  
1358 quiz::Quiz("Envelope Tracking","This describes the main action of an envelope follow\  
1359 er: a module or section of a module that follows the loudness of a signal and output\  
1360 s a voltage that corresponds to - tracks - that input."),  
1361 quiz::Quiz("Envelope","The collective term for the four elements of the lifespan of \  
1362 a sound: Attack, Decay, Sustain and Release (ASDR). The envelope of a sound describe\  
1363 s how a sound or audio signal varies in intensity over a period of time."),  
1364 quiz::Quiz("Equal Loudness Contours","A drawing of several curves showing how loud t\  
1365 he tones of different frequencies would have to be played for a person to say they w\  
1366 ere of equal loudness. (See also "Fletcher-Munson Curves.")"),  
1367 quiz::Quiz("Equalizer","An audio signal processor that uses one or more filters to b\  
1368 oost or cut the amplitude (volume) of certain frequencies within the sound. The unde\  
1369 rlying principle is to balance or "equalize" the frequency response of the audio sys\  
1370 tem, or to create balance between multiple signals in a sonic space. However, audio \  
1371 engineers may use equalizers to alter or "color" the sound in many different ways.")\  
1372 ,  
1373 quiz::Quiz("Eurorack","Eurorack is arguably the most popular format of modular synth\  
1374 esizer today, with over 100 manufacturers and over 1000 modules available. It was cr\  
1375 eated by Doepfer Musikelektronik in 1995, basing its size off the Eurorack format fo
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1376 r lab equipment. Some users will try to tell you that Eurorack doesn't "sound" as go\
1377 od as other formats, but that's just based on a few substandard manufacturers or mod\
1378 ules; there's nothing inherent to the standard that makes a huge difference in the f\
1379 inal sound (no; the difference between 12 and 15 volt power supplies is not enough t\
1380 o most ears)."),
1381 quiz::Quiz("Expander","A signal processor (or plug-in) that performs the opposite fu\
1382 nction of a compressor, expanding the dynamic range of an audio signal rather than c\
1383 ompressing it. It accomplishes this by further reducing the amplitude of signals tha\
1384 t drop below a set threshold."),
1385 quiz::Quiz("Expansion Ratio","The rate by which an expander attenuates an incoming s\
1386 ignal, measured in decibels. For example, an expansion ratio of 2:1 means the expand\
1387 er will reduce the signal by 2dB for every 1dB it drops below the threshold. If the \
1388 signal falls 3dB below the threshold, the expander attenuates it by 6 dB, and so on.\
1389 "),
1390 quiz::Quiz("Exponential","In general terms, this is a mathematical curve that starts\
1391 out relatively flat and then bends to climb steeply. In synthesizer terms, it most \
1392 often refers to the control voltage scheme where a change of 1 volt corresponds to a\
1393 n increased pitch of one octave, which is doubling in cycles (vibrations) per second\
1394 . This is in contrast to a linear system where 1 volt increase would always result i\
1395 n the same increase of cycles per second."),
1396 quiz::Quiz("Fade","A gradual reduction of the level of the audio signal, or a gradua\
1397 l change of level from one pre-set level to another."),
1398 quiz::Quiz("Fader","A control which adjusts the level (gain or attenuation) of an in\
1399 coming signal to a channel or grouping of channels on a console."),
1400 quiz::Quiz("Far Field","The region away from a loudspeaker at which the sound drops \
1401 6dB for each doubling of the distance, up to the critical distance. The beginning of\
1402 the far field varies according to the size of the speaker, but in most cases the fa\
1403 r field begins around 3 feet from the sound source. Audio engineers often use both n\
1404 ear field and far field monitoring when fine-tuning a mix. (See also "Critical Dista\
1405 nce," "Near Field.")"),
1406 quiz::Quiz("Feed","To send an audio or control signal to."),
1407 quiz::Quiz("Feedback Control","The control on a delay line or delay effects device t\
1408 hat controls the amount of feedback into the system."),
1409 quiz::Quiz("Feedback","The return of a portion of the output signal back into the in\
1410 put of a system. This can be done in a controlled manner through a feedback circuit \
1411 to alter the sound of an instrument (most commonly electric guitars or analog synths\
1412 ). It can also describe the unwanted feedback loop created when an open microphone i\
1413 s picking up the sound from a nearby speaker, generating a loud, oscillating frequen\
1414 cy that increases in intensity until the feedback loop is broken by turning off the \
1415 mic or speaker, or by use of an equalizer to attenuate the frequency."),
1416 quiz::Quiz("Fidelity","A term describing how accurately a sound is reproduced from i\
1417 ts original source."),
1418 quiz::Quiz("Figure-8 Pattern","A microphone pickup pattern which is most sensitive t\
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1419 o picking up sounds directly in front and back of the mic, effectively rejecting sou\
1420 nds coming from the sides."),
1421 quiz::Quiz("Filter","A module that reduced or removes certain frequencies and harmon\
1422 ics from the sound that is passed through it. In a synthesizer, the most typical fil\
1423 ter types are low pass (passes all of the harmonics below its cutoff or corner frequ\
1424 ency untouched, and then reduces the level of higher harmonics the further you go ab\
1425 ove that cutoff frequency), high pass (passes all harmonics above its cutoff frequen\
1426 cy untouched, and reduces the level of progressively lower harmonics below the cutoff\
1427 f), bandpass (harmonics right around the cutoff are passed intact, and then reduced \
1428 more in level the further away they are above or below the cutoff frequency), and no\
1429 tch (harmonics right around the cutoff frequency are reduced or cut out entirely; ot\
1430 hers above or below are allowed to live)."),
1431 quiz::Quiz("Flanger","A signal processor often identified as the one that creates a \
1432 “jet taking off” whoosh. What’s going on behind the panel is that a copy of the input\
1433 t signal is delayed by a very small amount (longer than a chorus effect; shorter tha\
1434 n an echo effect) and mixed in with the original. When the delay is constant, the re\
1435 sult is a “comb filter” where certain harmonics are cancelled out as they are mixed \
1436 back on top of themselves out of phase. When the delay is varied over time, you get \
1437 swooshes and sweeps. The effect was originally created by playing two tape reels of \
1438 the same song, starting them in time with each other, and dragging your finger on th\
1439 e flange of one of the tape reels to delay it."),
1440 quiz::Quiz("Flanging","An audio effect caused by blending the signal with a copy of \
1441 that signal at a slight time delay, then modifying the delayed copy, creating a “swi\
1442 rling” sound. This was originally accomplished in analog tape recording by playing t\
1443 he original tape and the copy on two tape machines simultaneously, then physically p\
1444 ressing on the flange of one of the machines to alter the timing of the duplicate tr\
1445 ack. These days, most flanging is done through delay boxes or digital plug-ins."),
1446 quiz::Quiz("Flat","1) A term used to describe an even frequency response in a device\
1447 or speaker, meaning that the device/speaker treats all frequencies the same without\
1448 the need for EQ. When displayed graphically, the frequency response is shown as a “\
1449 flat” line with no peaks or valleys. 2) In music, describes a note or pitch that is \
1450 out of tune, sounding at a slightly lower frequency than it should. 3) In music nota\
1451 tion, an “accidental” mark that instructs the player to play/sing the note one-half \
1452 step lower."),
1453 quiz::Quiz("Fletcher-Munson Curves","Also known as “Equal Loudness Contours,” a set \
1454 of graphical curves plotted to illustrate how the human ear responds to different fr\
1455 equencies at different volume levels. Named after the two researchers who first plot\
1456 ted the curves. (See also “Equal Loudness Contours.”)"),
1457 quiz::Quiz("Flip-Flop","In binary logic terms, a flip-flop toggles between high and \
1458 low every time it receives an input trigger (i.e. the first trigger would set the ou\
1459 tput high, the second trigger sets it low again, and so on). In clock or audio terms\
1460 , it divides the speed of an input clock or square wave by 2."),
1461 quiz::Quiz("Floating Unbalanced Line","A connection “workaround” in which an unbalan\
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1462 ced output is connected to a balanced input by modifying the connections in the line\  
1463 to resemble a balanced line, alleviating unwanted hum or buzz."),  
1464 quiz::Quiz("Fly In","To add sounds into a mix or recording that have no synchronizat\  
1465 ion."),  
1466 quiz::Quiz("Flying Bus","This is a very simple type of power distribution or bus boa\  
1467 rd that typically uses a ribbon cable with multiple connectors along its length to t\  
1468 ake the output of your power supply and distribute it to your individual modules. Th\  
1469 ey're cheap and easy to install and use, but in a few cases might be a cause of nois\  
1470 e being shared between modules."),  
1471 quiz::Quiz("FM","Frequency modulation (FM for short) refers to a synthesis technique\  
1472 where the pitch of an oscillator is varied (modulated) very quickly - at audio rate\  
1473 s - by another oscillator. The result is a complex side of harmonics that may either\  
1474 be nicely in tune or clangorous and "out of tune" with the fundamental pitch of the\  
1475 main oscillator."),  
1476 quiz::Quiz("FOH","In live audio settings, the location in a venue opposite the stage\  
1477 , where live audio for the show is controlled and mixed."),  
1478 quiz::Quiz("Foldback","A stage monitoring system used in live audio. A set of on-sta\  
1479 ge speakers called monitors or wedges (or "foldback speakers" in British countries) \  
1480 are fed a special mix of audio signals for the onstage performers to hear in order t\  
1481 o play. This mix is usually different from the FOH (front-of-house) mix that the aud\  
1482 ience hears, and is sometimes controlled by a second engineer through amplifiers and\  
1483 speakers separate from the main sound system. This type of stage monitoring is freq\  
1484 uently susceptible to feedback from the microphones, and in certain venues can cause\  
1485 unwanted reflective noise that makes it difficult for FOH engineers to create a goo\  
1486 d mix for the audience. For this reason, many live audio systems now use in-ear moni\  
1487 toring as an alternative to stage monitors to control the onstage noise and reduce t\  
1488 he risk of feedback."),  
1489 quiz::Quiz("Foot Pedal","An effects device controlled by a musician with his foot."),  
1490 quiz::Quiz("Foot Switch","A switch placed on the floor and pressed by a musician to \  
1491 do various functions."),  
1492 quiz::Quiz("Force-Sensing Resistor","In modular systems, an FSR (Force-Sensing or -S\  
1493 ensitive Resistor) usually takes the form of a circular pad that you press on to var\  
1494 y a parameter. It acts as a resistor that decreases in resistance the harder you pre\  
1495 ss."),  
1496 quiz::Quiz("Formant","Many instruments based on vibrating tubes - including our own \  
1497 vocal tract - have certain frequencies that they like to vibrate or "resonate" at. W\  
1498 hen you send a sound down these tubes, they will accentuate the frequency of that so\  
1499 und (or some of its harmonics) to match these resonate frequencies. Each of these re\  
1500 sonant frequencies is known as a formant of that instrument. A common way of synthes\  
1501 izing vocal-like sounds is to pass an oscillator through a filter or equalizer that \  
1502 has several formant peaks, spaced apart in ways that mimic certain vowels. Formant i\  
1503 s an element in the sound of a voice or instrument that does not change frequency as\  
1504 different pitches are sounded. Formants are essentially "fixed" frequencies or reso
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1505 nances that occur as a result of the physical structure of the sound source. These f\
1506 requencies are what create timbre, that element of sound that creates the specific s\
1507 ound of a guitar, a flute, a male or female voice, etc."),
1508 quiz::Quiz("Format","1) One of many different media used to store and reproduce audi\
1509 o, whether in the recording studio or for listening purposes. Examples include curre\
1510 ntly used physical formats such as vinyl records and compact discs; obsolete formats\
1511 such as cassette tape, 8-track tape and DAT; analog recording staples such as reel-\
1512 to-reel multitrack tape; and many different digital audio file formats such as mp3, \
1513 WAV, WMA, AIFF and others. 2) Format can also describe specific parameters when reco\
1514 rding to analog tape, such as number of tracks, width, spacing and order. 3) To prep\
1515 are a hard drive or memory card for use, usually erasing all existing data in the pr\
1516 ocess."),
1517 quiz::Quiz("Four Quadrant Multiplier","A Four-Quadrant Multiplier is a special case \
1518 of Amplitude Modulation (AM). It is also referred to as ring or balanced modulation.\
1519 One signal changes the level of - \"multiplies\" - the level of a second signal. A \
1520 typical use is two VCOs running at audio rates fed into a ring modulator (a four-qua\
1521 drant multiplier). The output is a complex set of component tones that don't follow \
1522 typical "musical" spacing based on octaves above the fundamental that harmonics usua\
1523 lly follow. Namely, the modulation frequency is both added to and subtracted from th\
1524 e carrier's frequency; the resulting harmonics replace the original carrier and modu\
1525 lator. Say the carrier was a sine wave (only the fundamental harmonic present) at 60\
1526 0Hz, and the modulator was a sine wave at 100Hz. The result would be a tone that had\
1527 frequency components at 500 and 700Hz."),
1528 quiz::Quiz("FracRack","A less-common format of modular synthesizers put forward by P\
1529 AiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5" (3.8 cm) wi\
1530 de by 3U, or 5.25" (13.3 cm) high."),
1531 quiz::Quiz("Fractional Rack","A less-common format of modular synthesizers put forwa\
1532 rd by PAiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5" (3.8\
1533 cm) wide by 3U, or 5.25" (13.3 cm) high."),
1534 quiz::Quiz("Frequency Modulation (FM) Synthesis","A method of sound synthesis in whi\
1535 ch the frequencies generated by one oscillator (the carrier) are altered by the outp\
1536 ut of one or more additional oscillators (operators) to create a diversity of harmon\
1537 ically rich sounds."),
1538 quiz::Quiz("Frequency Range","1) The range of frequencies over which an electronic d\
1539 evice puts out a useful signal (see also "Bandwidth"). 2) The range of frequencies t\
1540 hat can be substantially transmitted or received in relation to a sound source. Each\
1541 instrument has a certain frequency range in which it can play; the human ear can al\
1542 so hear within a certain frequency range."),
1543 quiz::Quiz("Frequency Response","The range between high and low frequencies that a c\
1544 omponent of an audio system can adequately handle, transmit or receive."),
1545 quiz::Quiz("Frequency-Agile","In wireless microphone systems, frequency-agile descri\
1546 bes the ability of the system to operate on a choice of different RF frequencies wit\
1547 hin a certain bandwidth. Frequency-agile systems are preferred for live touring and \
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1548 in areas with high concentrations of radio signals (like large cities) because the R\
1549 F frequency of the device can be changed to avoid interference."),
1550 quiz::Quiz("Frequency-Shift Key (FSK)","A now out-of-date protocol in which a sync t\
1551 one is recorded onto a spare track of a multi-track tape recorder to enable electron\
1552 ic devices (mainly drum machines) to perform in sync with the tape. While some older\
1553 devices still read FSK, an updated protocol (Smart FSK) is now more commonly used. \
1554 (See also "Smart FSK.")"),
1555 quiz::Quiz("Frequency","The number of occurrences of a particular event within a cer\
1556 tain amount of time. In audio and acoustics, frequency specifically refers to the nu\
1557 mber of complete cycles a vibration or waveform makes in a second, measured in cycle\
1558 s per second, or Hertz (Hz). In sound, frequency determines what we hear as pitch. T\
1559 he longer the wavelength, the fewer the cycles per second, and the lower the pitch."\
1560 ),
1561 quiz::Quiz("Front-of-House","(Abbreviated FOH) In live audio settings, the location \
1562 in a venue opposite the stage, where live audio for the show is controlled and mixed\
1563 ."),
1564 quiz::Quiz("FSR","In modular systems, an FSR (Force-Sensing or -Sensitive Resistor) \
1565 usually takes the form of a circular pad that you press on to vary a parameter. It a\
1566 cts as a resistor that decreases in resistance the harder you press."),
1567 quiz::Quiz("Full-Normalled","Describes the configuration within a patch bay in which\
1568 the jacks form a connected pathway until a patch cord is inserted to change the pat\
1569 h. When a patch bay is "full-normalled," the connection is altered by inserting a co\
1570 rd into either the input or output side; when it is "half-normalled," the path chang\
1571 es only when a cord is plugged into the input. "Non-normalled" or "open" means there\
1572 are no internal connections, and each input sends the signal through its correspond\
1573 ing output."),
1574 quiz::Quiz("Full-Wave Rectifier","A full-wave rectifier takes any negative voltages \
1575 and inverts them so they become positive. This effectively doubles the frequency of \
1576 many simple waveforms, like the triangle and sine."),
1577 quiz::Quiz("Function Generator","The term function generator can have two meanings i\
1578 n the world of synthesis. One, test equipment that generates waveforms such as sine \
1579 or square waves are often called "function generators." Two, envelope generators are\
1580 sometimes referred to as "function generators." In both cases, "function" means to \
1581 execute an equation of some sort, such as creating a periodic waveform such as a sin\
1582 e or creating a rise & fall in response to a trigger."),
1583 quiz::Quiz("Fundamental","(Also called fundamental frequency or first harmonic) The \
1584 lowest frequency present in the sounding of a note by musical instrument or voice.")\
1585 ,
1586 quiz::Quiz("Gain Control","A device that changes the gain of an amplifier or circuit\
1587 , often a knob (potentiometer) that can be turned. In a mixing console, each channel\
1588 usually has its own gain control to regulate the gain of the signal coming into the\
1589 board—not to be confused with the channel "fader," which regulates the output of an\
1590 already-amplified signal."),
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1591 quiz::Quiz("Gain Reduction","The action of a compressor or limiter in regulating the\
1592 amplitude of the audio signal."),
1593 quiz::Quiz("Gain Structure","A term that describes the interconnection of multiple c\
1594 omponents in an audio system, and the amount of gain increase or reduction that occu\
1595 rs at each point. A configuration with a good gain structure means that the componen\
1596 ts are working properly together to provide optimal gain with minimal distortion or \
1597 noise."),
1598 quiz::Quiz("Gain","The amount of increase in audio signal strength, often expressed \
1599 in dB."),
1600 quiz::Quiz("Gate Detector","This is one of the main signal types that are passed aro\
1601 und inside a modular synthesizer. It jumps to high level – typically 5 volts – when \
1602 a new note is supposed to start (such as when you press a key on a keyboard controll\
1603 er), or when a sequencer jumps to the next “stage” or note. A gate typically stays a\
1604 t that level for the duration of the note (i.e. while the key is being held down), a\
1605 nd suddenly drops or “goes low” to its resting level – typically 0 volts, but someti\
1606 mes -5 volts or another number – when the note ends (i.e. when the key is released).\
1607 In practice, when a gate signal is sent to a typical envelope generator, the start \
1608 of the gate (when it “goes high”) tells the envelope to go through its Attack and De\
1609 cay stages; while the gate remains high, the envelope stays at its Sustain level, an\
1610 d when the gate goes low again, the envelope moves onto its Release stage."),
1611 quiz::Quiz("Generation Loss","The amount of clarity lost when recorded audio is copi\
1612 ed, due to added noise and distortion."),
1613 quiz::Quiz("Generation","A term used to describe the number of times that the record\
1614 ed audio signal has been copied."),
1615 quiz::Quiz("Glide","Refers to a note that glides from one pitch to another while it \
1616 is still audible. The music term for this effect is portamento, which is a slurring \
1617 between notes. In a synthesizer, this effect is created by causing the control volta\
1618 ge for the pitch of a note to slide from the pitch of the previous note rather than \
1619 make a discrete jump. The module that creates this effect is sometimes known as a sl\
1620 ew generator, slew limiter, slope generator, or lag. Some use the terms glide, gliss\
1621 ando, and portamento interchangeably, but if you want to split musical hairs, a glis\
1622 sando (gliss) is a different effect where the intermediate notes are more distinct –\
1623 such as played rapidly in order – rather than slurred through."),
1624 quiz::Quiz("Golden Section","(also called Golden Ratio) A ratio of height to width t\
1625 o length, where the width is approximately 1.6 times the height, and the length appr\
1626 oximately 2.6 times the height. First calculated by the ancient Greeks, this ratio (\
1627 known mathematically as “phi”) is used as an optimal ratio in many applications, inc\
1628 luding room dimensions and studio design (to achieve “optimal acoustics” in the room\
1629 ), and even in the design of certain acoustic instruments."),
1630 quiz::Quiz("Granular Synthesis","Granular synthesis can be thought of as particle th\
1631 eory applied to sound. The concept is that a sound can be broken down into very smal\
1632 l “grains” – typically 1-50 or 100 msec in duration. These tiny snippets are then pl\
1633 ayed back to reproduce the original sound, or to create new sounds by changing the s\
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1634 peed, pitch, volume, playback order, and direction of the individual grains. You can\  
1635 crossfade between these modified grains, or layer more grains on top. The result ca\  
1636 n range from audio processing tricks such as changing speed without changing pitch a\  
1637 nd vice versa, to creating psychedelic “clouds” of sound (and indeed, there is a pop\  
1638 ular module called Clouds).”),  
1639 quiz::Quiz("Graphic Equalizer","A type of equalizer that can adjust various frequenc\  
1640 ies of the incoming signal using sliders that are assigned to specific frequency ban\  
1641 ds. (See also “Equalizer.”)”),  
1642 quiz::Quiz("Ground Lift Plug","An adapter that enables a three-prong power cord to p\  
1643 lug into two-prong outlet. Some engineers wrongly use this plug to interrupt the gro\  
1644 und connection and prevent buzz, but it is a VERY unsafe practice to break the groun\  
1645 d connection using this plug without grounding the unit by another means.”),  
1646 quiz::Quiz("Ground Lift Switch","A switch that breaks the connection between the gro\  
1647 und point in one circuit and the ground point in another circuit, for the purpose of\  
1648 eliminating hum or buzz caused by ground loops.”),  
1649 quiz::Quiz("Ground Loop","A situation caused when one or more electronic devices are\  
1650 connected to the same ground at different points. The devices operate at different \  
1651 ground potentials, which creates voltage along the ground, resulting in a low-freque\  
1652 ncy hum that can be annoying at best and cause damage to gear at worst. The best res\  
1653 olution for ground loops is to ground all devices at the same point using a central \  
1654 power source. An alternative solution is to break the loop via ground lift switches \  
1655 or plugs, but this should be avoided when possible as it is considered an unsafe man\  
1656 agement of electricity.”),  
1657 quiz::Quiz("Group (or Grouping)","A number of input channels on a console that can b\  
1658 e controlled and adjusted as a single set before sending the combined signal to the \  
1659 master output. Sometimes also called “Submix,” “Bus” or just “Group.””),  
1660 quiz::Quiz("Group Delay","In audio, group delay is a phenomenon within all electroni\  
1661 c audio devices (e.g., speakers, amplifiers) in which different frequencies in the s\  
1662 ignal are output at slight delays from one another. In simpler terms, lower frequenc\  
1663 ies are delivered slightly more slowly than higher ones. In all devices, there is an\  
1664 inherent delay between input and output of the signal, but group delay specifically\  
1665 deals with the time delays between specific frequencies of the sound. The goal in a\  
1666 ny configuration is to keep the group delay as small as possible; in cases of extrem\  
1667 ely poor configurations, the delays between highs and lows can be audible.”),  
1668 quiz::Quiz("Guitar Controller","An electric guitar (or device played like a guitar) \  
1669 that transmits MIDI data that can be used to control synthesizers and sound modules.\  
1670 "),  
1671 quiz::Quiz("Guitar Processor","A device that adds audio effects to a direct guitar s\  
1672 ignal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, \  
1673 etc. Effects processors can occur as individual effects boxes or multi-sound pedal b\  
1674 oards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between t\  
1675 he guitar and the console. They can also be found as presets in guitar amplifiers, o\  
1676 r even as digital plug-ins within a DAW.”),
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1677 quiz::Quiz("Haas Effect", "(Also called Precedence Effect) Simply stated, a factor in\  
1678 human hearing in which we perceive the source of a sound by its timing rather than \  
1679 its sound level. In his research, Helmut Haas determined that the first sound waves \  
1680 to reach our ears help our brains determine where the sound is coming from, rather t\  
1681 han its reflection or reproduction from another source. The reflection of the sound \  
1682 must be at least 10dB louder than the original source, or delayed by more than 30ms \  
1683 (where we can perceive it as an echo), before it affects our perception of the direc\  
1684 tion of the sound. This is what helps us distinguish the original sound source witho\  
1685 ut being confused by reflections and reverberations off of nearby surfaces. Understa\  
1686 nding the Haas effect is particularly useful in live audio settings, especially in l\  
1687 arge venues where loudspeakers are time-delayed to match the initial sound waves com\  
1688 ing from the source."),  
1689 quiz::Quiz("Half Step", "A change in pitch equivalent to adjacent keys on a piano. Al\  
1690 so known as a “semitone.””),  
1691 quiz::Quiz("Half-Normalled", "Describes the configuration within a patch bay in which\  
1692 the jacks form a connected pathway until a patch cord is inserted to change the pat\  
1693 h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\  
1694 rd into either the input or output side; when it is “half-normalled,” the path chang\  
1695 es only when a cord is plugged into the input. “Non-normalled” or “open” means there\  
1696 are no internal connections, and each input sends the signal through its correspond\  
1697 ing output.”),  
1698 quiz::Quiz("Half-Wave Rectifier", "A half-wave rectifier passes only positive voltage\  
1699 s, and replaces anything negative with 0v. In other words, anything “below zero” is \  
1700 clipped off.”),  
1701 quiz::Quiz("Hall Program", "A setting of a digital delay/reverb effects unit that app\  
1702 roximates concert halls. Hall programs are characterized by pre-delay of up to 25 ms\  
1703 .”),  
1704 quiz::Quiz("Hard Knee", "In compression, refers to a more abrupt introduction of comp\  
1705 ression of the signal once the sound level crosses the threshold. (See also “Knee.”)\  
1706 "),  
1707 quiz::Quiz("Hard Sync", "This is the most common type of oscillator sync where the sl\  
1708 ave oscillator will reset its waveform whenever it receives a sync pulse. If the typ\  
1709 e of sync is not specified, then it’s probably hard sync.”),  
1710 quiz::Quiz("Harmonic Distortion", "The presence of harmonics in the output signal of \  
1711 a device which were not present in the input signal, usually for the purpose of chan\  
1712 ging the instrument’s timbre.”),  
1713 quiz::Quiz("Harmonic", "A single harmonic is the purest sound possible: It contains n\  
1714 o overtones or other identifying characteristics aside from its pitch and loudness. \  
1715 The shape of its vibration – whether it be vibrating the air so you can hear it, or \  
1716 causing the electrical vibrations of a voltage going up and down – is a sine wave. M\  
1717 ost of the time, overtones have a very specific pitch relationship to each other. Th\  
1718 e first or lowest harmonic – known as the ‘fundamental’ – is the pitch of the sound,\  
1719 just as the lowest note of a chord is its ‘root.’ The other harmonics are higher, a\  

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1720 nd spaced out as integer multiples of the fundamental: two times its frequency, thre\  
1721 e times, four times, and so forth. The first few harmonics happen to have a nice mus\  
1722 ical spacing: an octave; an octave and a fifth; two octaves. But the higher they get\  
1723 , the less musical they may seem."),  
1724 quiz::Quiz("Harmonics","Whole number multiples of the fundamental frequency that occ\  
1725 ur naturally within the playing of a tone. Mathematically, if the fundamental freque\  
1726 ncy is x, the harmonics would be 2x, 3x, 4x, etc. For example, if the fundamental fr\  
1727 equency of the note played is 440Hz (or A-440), the harmonics would be 880Hz, 1320Hz\  
1728 , 1760Hz, and so on. The presence of harmonics in the tone is what creates the timbr\  
1729 e of an instrument or voice."),  
1730 quiz::Quiz("Head","In tape recording, an electromagnetic transducer that magneticall\  
1731 y affects the tape passing over it. Recording/playback heads change the audio signal\  
1732 from electrical energy to magnetic energy and back, for recording and playback purp\  
1733 poses. An erase head creates a powerful electromagnetic field to the tape to erase pr\  
1734 evious signals from the tape."),  
1735 quiz::Quiz("Headroom","The difference in dB between normal operating level and clipp\  
1736 ing level in an amplifier or audio device. Also describes the difference in dB betw\  
1737 een the peak levels of a recording and the point at which the signal distorts. (Also\  
1738 called "Margin.")"),  
1739 quiz::Quiz("Hertz/Volt","A system where a change of 1 volt at the input results in a\  
1740 change in pitch of a fixed number of hertz (cycles per second), rather than a fixed\  
1741 musical interval."),  
1742 quiz::Quiz("Hertz","(Abbreviated Hz) 1) The unit of measurement for frequency, speci\  
1743 fically, the number of complete wave cycles that occur in a second (cycles per secon\  
1744 d). 1 Hz = 1 complete wave per second. 2) A popular rental car company (not typicall\  
1745 y used in recording except for transport to the studio)."),  
1746 quiz::Quiz("Hi-Hat","In drum sets, double cymbal on a stand, usually positioned next\  
1747 to the snare, which can be played with a foot pedal and/or by the top cymbal being \  
1748 hit with a stick."),  
1749 quiz::Quiz("Hi-Z","(abbreviated Hi-Z) Described as an impedance or resistance of sev\  
1750 eral thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more o\  
1751 hms. (See also "Impedance.")"),  
1752 quiz::Quiz("High (gate)","When a gate signal is at the voltage level (typically 5 vo\  
1753 lts, although it can be more) that indicates it is "on" - such as when a note is bei\  
1754 ng held down on a keyboard controller - it is said that the gate is high."),  
1755 quiz::Quiz("High Impedance","(abbreviated Hi-Z) Described as an impedance or resista\  
1756 nce of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000\  
1757 or more ohms. (See also "Impedance.")"),  
1758 quiz::Quiz("High Pass Filter","An audio filter that attenuates signals below a certa\  
1759 in frequency (the cut-off frequency) and passes signals with frequencies that are hi\  
1760 gher."),  
1761 quiz::Quiz("High-End","Highs or High-End - Short for "high frequencies," loosely the\  
1762 frequencies above 4000 Hz. Usually meant in the context of "highs, mids and lows" i\  

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1763 n an audio signal."),
1764 quiz::Quiz("High-Pass Filter","The high pass filter (HPF) design passes harmonics ab\
1765 ove its cutoff or corner frequency untouched, and reduces the level of lower harmoni\
1766 cs depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) hi\
1767 gh pass filter, harmonics one octave below the cutoff frequency (in other words, one\
1768 half the cutoff frequency) are reduced in level by 12 dB; harmonics two octaves bel\
1769 ow the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pa\
1770 ss filters are typically used to create bright sounds where the higher harmonics are\
1771 much stronger than the fundamental and lower harmonics – for example, the sound of \
1772 a harpsichord."),
1773 quiz::Quiz("Horizontal Pitch","HP = Horizontal Pitch. In the Eurorack format for syn\
1774 thesizer modules, the width of a module is defined as the number of hp (horizontal p\
1775 itch) units. Each hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, alt\
1776 hough some are odd numbers. Also, modules tend to be ever so slightly less than exac\
1777 tly some multiple of 0.2” wide, just to make sure you don’t run into problems with e\
1778 ver so slightly too wide modules overlapping."),
1779 quiz::Quiz("Horn","1) A speaker or speaker enclosure where sound waves are sent by a\
1780 speaker cone or driver into a narrow opening which flares out to a larger opening. \
1781 2) One of several different types of brass musical instruments."),
1782 quiz::Quiz("House Sync","A reference signal such as SMPTE time code that is used to \
1783 keep all devices in the room in sync."),
1784 quiz::Quiz("HP","HP = Horizontal Pitch. In the Eurorack format for synthesizer modul\
1785 es, the width of a module is defined as the number of hp (horizontal pitch) units. E\
1786 ach hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, although some are\
1787 odd numbers. Also, modules tend to be ever so slightly less than exactly some multi\
1788 ple of 0.2” wide, just to make sure you don’t run into problems with ever so slightl\
1789 y too wide modules overlapping."),
1790 quiz::Quiz("HPF","The high pass filter (HPF) design passes harmonics above its cutof\
1791 f or corner frequency untouched, and reduces the level of lower harmonics depending \
1792 on how far below the cutoff they are. In a 12dB/oct (decibel/octave) high pass filte\
1793 r, harmonics one octave below the cutoff frequency (in other words, one half the cut\
1794 off frequency) are reduced in level by 12 dB; harmonics two octaves below the cutoff\
1795 (one quarter the frequency) are reduced by 24dB, and so forth. High pass filters ar\
1796 e typically used to create bright sounds where the higher harmonics are much stronge\
1797 r than the fundamental and lower harmonics – for example, the sound of a harpsichord\
1798 ."),
1799 quiz::Quiz("Hum","1) The low-frequency pitch that occurs when power line current is \
1800 accidentally induced or fed into electronic equipment. The hum reflects the fundamenta\
1801 l frequency of the current (60 Hz in the U.S., and 50 Hz in many European countries)\
1802 . 2) To vocalize a pitch without opening one’s mouth."),
1803 quiz::Quiz("Hybrid Power Supply","A hybrid power supply uses a lower weight, more ef\
1804 ficient switching power supply to perform most of the drop in voltage – say, from 12\
1805 0v AC to 15v DC – and then uses a linear power supply for the remaining much smaller\
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1806 drop, such as from 15v to 12v. These are becoming the preferred design in many modu\
1807 lar synthesizer enclosures. Shortcomings with the power supply – too noisy, or not e\
1808 nough – tend to be at the cause of many unexpected problems in modular synthesizers.\
1809 "),
1810 quiz::Quiz("Hypercardioid","A variation of the cardioid microphone pick up sensitivi\
1811 ty pattern in which the shape of the optimal pickup area is tighter and more directi\
1812 onal than cardioid. Hypercardioid microphones are most sensitive directly on-axis in\
1813 front of the microphone, and begins rejecting sounds between 90-150 degrees off-axi\
1814 s, depending on the tightness of the pattern."),
1815 quiz::Quiz("Hz/V","A system where a change of 1 volt at the input results in a chang\
1816 e in pitch of a fixed number of hertz (cycles per second), rather than a fixed music\
1817 al interval."),
1818 quiz::Quiz("Hz","An abbreviation for the term Hertz, or the unit of frequency."),
1819 quiz::Quiz("IADSR","This is an Attack/Decay/Sustain/Release (ADSR) envelope generato\
1820 r that allows you to start the attack phase at an initial level – the “I” – rather t\
1821 han the customary 0 volts. The envelopes in the Prophet VS, as well as a module from\
1822 Ladik, have this capability."),
1823 quiz::Quiz("IC","Integrated Circuit – A miniature circuit of many components set on \
1824 semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”\
1825 "),
1826 quiz::Quiz("Imaging","Refers to the ability to localize a specific sound within the \
1827 sound space. In recording environment, it refers to “placing” instruments within the\
1828 stereo or surround field so that it when the sound is played through speakers, it f\
1829 ools our ears into thinking the sound source is in emanating from a specific point i\
1830 nstead of from the speakers. In live audio and sound reinforcement, the principle of\
1831 imaging is the same, the goal being to make the audience perceive the sounds as com\
1832 ing from performers on the stage, rather than from the speakers."),
1833 quiz::Quiz("Impedance","Refers to the resistance of a circuit or device to alternati\
1834 ng current, which can be mathematically described as the ratio of voltage to current\
1835 . Differences in impedance between devices in the studio can affect how they work to\
1836 gether. Impedance is abbreviated by the letter Z, and measured in ohms (W)."),
1837 quiz::Quiz("In Line Console","An audio mixing console that is designed and configure\
1838 d so each channel strip can be used for both recording and monitoring functions duri\
1839 ng multitrack recording. This configuration is in contrast to split mixing consoles,\
1840 which requires separate channels on the board for recording and monitoring function\
1841 s."),
1842 quiz::Quiz("In Phase","The desirable situation in which two or more devices (and the\
1843 ir respective audio signals) are on the same side of the polarity spectrum, producin\
1844 g waveforms that do not conflict or cancel each other out."),
1845 quiz::Quiz("In Port","A jack on a MIDI device or computer that will accept an incomi\
1846 ng data signal."),
1847 quiz::Quiz("Inductance","A characteristic of electrical conductors in which electric\
1848 al charge (voltage) is produced or stored magnetically due to the natural resistance\
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1849 to change in the electrical current. Inductance is an electromagnetic principle tha\
1850 t can either assist in audio applications (as in loudspeakers) or cause resistance (\
1851 as in using speaker wire whose gauge is too low for the application)."),
1852 quiz::Quiz("Inductor","A device (usually a coil of wire) that converts electrical en\
1853 ergy into stored magnetic energy as electrical current passes through it. Commonly f\
1854 ound in a variety of audio applications such as guitar pickups and loudspeakers."),
1855 quiz::Quiz("Infinite Baffle","A loudspeaker mount or enclosure designed so that soun\
1856 d waves coming from the front theoretically do not reach the back, preventing the so\
1857 und waves from cancelling each other out. The term "infinite" comes from the idea th\
1858 at mounting the speaker on a wall with no end points would not allow sound waves to \
1859 migrate behind it. Of course, this is physically impossible, so infinite baffles are\
1860 designed to replicate this as much as possible. Examples of infinite baffles are mo\
1861 unting the speaker on a wall of an enclosed room, or building it inside a sealed cab\
1862 inet large enough to prevent rear sounds from affecting the cone from the back."),
1863 quiz::Quiz("Initial/Attack/Decay/Sustain/Release","This is an Attack/Decay/Sustain/R\
1864 elease (ADSR) envelope generator that allows you to start the attack phase at an ini\
1865 tial level – the "I" – rather than the customary 0 volts. The envelopes in the Proph\
1866 et VS, as well as a module from Ladik, have this capability."),
1867 quiz::Quiz("Input / Output (I/O)","I/O – An abbreviation for "Input/Output." In audi\
1868 o, it refers to any device, program or system involving the transferring of electric\
1869 al/audio signals or data."),
1870 quiz::Quiz("Input Impedance","The opposition to current flow by the first circuits o\
1871 f a device."),
1872 quiz::Quiz("Input Monitoring","A setting on many DAWs that allows you to monitor the\
1873 live input signal coming into the DAW (as opposed to the recorded signal)."),
1874 quiz::Quiz("Input","The jack or physical location where a device receives a signal. \
1875 Also refers to the incoming signal itself."),
1876 quiz::Quiz("Insert","An access in the signal chain (usually in the mixing console or\
1877 virtually within a DAW) in which a device, signal processor or digital plug-in can \
1878 be "inserted" into the circuit between pre-amplification and the channel or bus outp\
1879 ut. Commonly used to add processing such as reverb, compression or EQ to a channel o\
1880 r group of channels."),
1881 quiz::Quiz("Instrument Amplifier","A device that has a power amplifier and speaker t\
1882 o reproduce the signal put out by an electric instrument."),
1883 quiz::Quiz("Instrument Out Direct","Feeding the output of an electric instrument (li\
1884 ke an electric guitar) directly to the recording console or tape recorder, as oppose\
1885 d to miking the amplifier."),
1886 quiz::Quiz("Insulator","A substance such as glass, air, plastic, etc., that will (fo\
1887 r all practical purposes) not conduct electricity."),
1888 quiz::Quiz("Integrated Circuit","Integrated Circuit (Abbreviated "IC") – A miniature\
1889 circuit of many components set on semiconductor material, used in electronics. A fa\
1890 ncency term for "chip" or "microchip.""),
1891 quiz::Quiz("Integrator","This function smoothens out an incoming signal so that the \
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1892 change in voltage level. “Integrator” is the technical name for this math function; \
1893 you are more likely to see this module called a slew limiter (where I go into more d\
1894 etail on its uses) or less often as a lag generator or processor."),
1895 quiz::Quiz("Interface","Any device or connection point that allows one unit to work,\
1896 drive or communicate with another unit, or that allows a human to interact with a c\
1897 omputer or other electronics. There are many examples of interfaces in professional \
1898 audio situations, including MIDI (Musical Instrument Digital Interface); audio inter\
1899 faces which connect audio inputs to your computer; and even your DAW program, which \
1900 displays a screen that enables you to assign instruments, adjust settings, record, m\
1901 ix and playback. Even the mixing console is an interface of sorts, connecting the ma\
1902 ny elements of the control room."),
1903 quiz::Quiz("Intermodulation (IM) Distortion","Distortion caused by two or more audio\
1904 signals of different frequencies interacting with one another. The sum and differen\
1905 ce of the frequencies produce new (usually unwanted frequencies) that didn't exist i\
1906 n any of the original frequencies."),
1907 quiz::Quiz("Inverse Square Law","A mathematical rule that describes an inverse relat\
1908 ionship between one quantity and the square of another quantity. In plain English, o\
1909 ne number goes down by a certain amount each time the other number doubles. In audio\
1910 and acoustics, the inverse square law says that in an open sound field with no obst\
1911 ructions, the sound pressure level will drop by half (6dB) each time the distance fr\
1912 om the sound source is doubled. (This equation is quite useful to audio engineers tr\
1913 ying to provide sound in open-air settings, for example.)"),
1914 quiz::Quiz("Inverter","An inverter multiplies an incoming control voltage by -1. In \
1915 the case of a gate or logic inverter, it reverses the high and low states so that (f\
1916 or example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pola\
1917 rizer, as it changes the polarity (+ versus -) of a signal. A control voltage invert\
1918 er is often combined with an offset voltage to adjust the output voltage into the de\
1919 sired range. For example, if you had an envelope generator that had an output range \
1920 of 0 to +8 volts, and you just inverted it, the result would be 0 to -8 volts. Since\
1921 some modules such as voltage controlled amplifiers usually expect only positive vol\
1922 tages, you would then need to add 8 volts to that result to get an upside-down (inve\
1923 rted) envelope that still had an overall range of 0 to +8v."),
1924 quiz::Quiz("Inverting Mixer","Most signal mixers make an effort to keep the same pol\
1925 arity of a signal as it passes through the mixer. However, some mixers may invert th\
1926 e polarity or “phase” of a signal (as it’s a simpler design); other mixers may allow\
1927 you to invert a signal on purpose so that you can experiment with tricks like addin\
1928 g one waveform or filter mode output out of phase with another coming from the same \
1929 oscillator or filter."),
1930 quiz::Quiz("Isolation","The process of containing sound within a certain area so tha\
1931 t it doesn't interact with other sounds. For example, acoustically treated isolation\
1932 booths are often used to record vocals or instruments in the studio to keep outside\
1933 noises from bleeding into the recording microphone, or likewise to keep vocals or o\
1934 ther sounds away from instrument mics during live recording sessions."),
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1935 quiz::Quiz("IV Cable","You often need to send one signal to multiple destinations. 0\  
1936 ptions for doing this include using dedicated multiples, free-floating widgets with \  
1937 multiple jacks wired together, or fancy cables that allow you plug one or two extra \  
1938 cables into them. The IV cable is one the latter: Made by Erthenvar, it has an extra\  
1939 3.5mm jack molded into the mid-point of the cable (loosely resembling an intravenou\  
1940 s or "IV" drip), in addition to having 3.5mm plugs at either end."),  
1941 quiz::Quiz("Jack","That hole you plug your patch cables into on the face of your syn\  
1942 thesizer modules? That's called a jack. The size and type of jack - 3.5mm, banana, o\  
1943 r 1/4" - often is one of the defining features of different synth module formats: 3U\  
1944 /Eurorack, 4U, and 5U/MU respectively. (No, a plug is not called a Jill. Actually, i\  
1945 t's the other way around: A plug is sometimes referred to as a male connector, and a\  
1946 jack is referred to as a female connector.)"),  
1947 quiz::Quiz("Jam Sync","A process available on some clock or syncing devices which re\  
1948 ads an external time code and recreates (or "jams") a new time code identical to the\  
1949 original external code for the syncing of devices. This function is mainly used for\  
1950 replacing code that has become degraded."),  
1951 quiz::Quiz("Karplus Strong","This is a physical modeling synthesis algorithm designe\  
1952 d to replicate the sound of plucked, vibrating strings - although it has also proven\  
1953 useful for some percussion sounds as well. A short sample - originally noise, altho\  
1954 ugh it can be a high frequency chirp or other sound - is sent to both the output, an\  
1955 d to a delay line. The output of a delay line is connected to a filter - originally \  
1956 a one-pole low pass filter; changing the filter has a huge effect on the character o\  
1957 f the sound - and then back to both the main output and the input of the delay line\  
1958 A few modules implement Karplus Strong synthesis, although it is an interesting cha\  
1959 llenge to patch yourself and play with the results."),  
1960 quiz::Quiz("Key","1) In music, the note scale in which a piece of music is written o\  
1961 r played, identified by the first note (tonic) of the scale, as in, "Key of C." 2) T\  
1962 he control of a dynamics processing device by an external audio signal through the u\  
1963 se of a side chain. 3) A digital or data code that unlocks the use of a device or so\  
1964 ftware. Example: Pro Tools is licensed through an iLok ID via the use of a physical \  
1965 USB key."),  
1966 quiz::Quiz("Keyboard Controller","A piano-styled keyboard that sends out MIDI signal\  
1967 s to control other MIDI devices. Most keyboard instruments are equipped with MIDI co\  
1968 ntrol capabilities, but dedicated MIDI keyboard controllers emit no audio signals, o\  
1969 nly MIDI data."),  
1970 quiz::Quiz("Keyboard Tracking","Most modular synths follow a strict relationship bet\  
1971 ween voltage and pitch, such as 1 volt per octave; any deviation would cause tuning \  
1972 errors. Because of this sensitivity, 1v/oct and similar signals and connections are \  
1973 sometimes specifically distinguished as keyboard tracking rather than just "CV" (con\  
1974 trol voltage) to make it clear they are not attenuated or otherwise modified when co\  
1975 ntrolling a function on a module."),  
1976 quiz::Quiz("Keyboard","Any musical instrument or computer controlled by pressing a k\  
1977 ey."),
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1978 quiz::Quiz("Keytar","A strap-on, lightweight, portable keyboard meant to allow keybo\
1979 ardists the same freedom (not to mention posturing opportunities) as guitarists."),
1980 quiz::Quiz("Kick Drum","The bass drum on a trap drum set, so called because it is pl\
1981 ayed with a kick pedal."),
1982 quiz::Quiz("Kilohertz (kHz)","kHz - An abbreviation for kilohertz (1000 Hz, or 1000 \
1983 cycles per second). Example: 2000 Hz = 2 kHz. Most commonly used in the studio for d\
1984 escribing audio frequency ranges or digital sampling rates."),
1985 quiz::Quiz("Knee","A function on a compressor that determines how abruptly or gradua\
1986 lly compression begins once the sound level crosses the threshold. So-called because\
1987 the graphic "bend" in the response curve is reminiscent of a knee. "Hard knee" refe\
1988 rs to an abrupt activation of the compressor, while "soft knee" refers to a more gra\
1989 dual change."),
1990 quiz::Quiz("Krell Patch","Recreating this patch is a challenge many modular musician\
1991 s like to tackle. It is based on the 1959 movie Forbidden Planet, in a segment where\
1992 they supposedly play the music of the ancient Krell race. In general terms, each no\
1993 te has a random pitch, envelope, and duration."),
1994 quiz::Quiz("Lag Generator","This function smoothes out an incoming signal so that th\
1995 e change in voltage level cannot exceed a certain number of volts per second. This c\
1996 auses the result to "lag behind" changes in the input. It is sometimes called a slew\
1997 limiter or technically as an integrator."),
1998 quiz::Quiz("Layering","Refers to almost any blending of similar multiple musical par\
1999 ts or sounds at once, often combined on one channel or assigned to one controller. I\
2000 n audio recording, layering usually involves recording similar takes of the same ins\
2001 trument or vocal (or duplicating parts with slight delays or chorusing effects) to c\
2002 reate a fuller, richer sound than the vocal/instrument by itself. In sound design, i\
2003 t also refers to blending multiple samples (example: two or more drum sounds) to cre\
2004 ate a fuller sound."),
2005 quiz::Quiz("Lead Sheet","A shorthand form of music notation (similar to a chord char\
2006 t) that displays the basic essential elements of a song so musicians can follow alon\
2007 g without the full notation of every note or expression. Lead sheets most commonly i\
2008 nclude a melody line written in music notation with chord changes above the staff, a\
2009 nd lyrics below it. (See also "Chord Chart.")"),
2010 quiz::Quiz("Leakage","Sounds from other instruments and sound sources that were not \
2011 intended to be picked up by the microphone."),
2012 quiz::Quiz("Level","The amount of signal strength; the amplitude, especially the ave\
2013 rage amplitude."),
2014 quiz::Quiz("LFO","This module produces repetitive, cycling waves ranging in frequenc\
2015 y from the low end of the audio spectrum to as slow as many seconds or even minutes \
2016 per cycle. They are used to produce effects such as tremolo (when controlling the lo\
2017 udness of a signal), vibrato (when controlling the pitch of a signal), repetitive fi\
2018 lter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an \
2019 oscillator, and more."),
2020 quiz::Quiz("Limiter","A type of compressor that sharply reduces (limits) the gain of\
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2021 the signal when the audio level reaches a certain threshold, typically used to prev\
2022 ent overload and signal peaking. A compressor effectively becomes a limiter when its\
2023 ratio is 10:1 or higher. (See also "Compressor.")"),
2024 quiz::Quiz("Line Input","Line Input ("Line In") – An input designed to take a line l\
2025 evel signal."),
2026 quiz::Quiz("Line Level","Most consumer and lower-cost professional audio equipment u\
2027 se a signal level reference known as line level or -10dBV (decibel volts). The most \
2028 common connectors are RCA (phono) or 3.5mm, although 1/4" is also used; the signal i\
2029 s "unbalanced" (it uses two wires: signal and ground). In the line level standard, a\
2030 sine wave that varies between +/-0.447 volts is considered to be at -10dBV. By cont\
2031 rast, a typical oscillator signal in a modular synthesizer is +/-5 to +/-8 volts. As\
2032 a result, you will need either an output module in your modular synth or one heckuv\
2033 a input attenuator on your mixer or recorder to plug your synth into equipment that \
2034 runs at line level. Similarly, you will need to substantially boost a line level sig\
2035 nal to get it up to modular standards to process in your modular synth."),
2036 quiz::Quiz("Line Output","Line Output ("Line Out") – Any output that sends out a lin\
2037 e level signal, such as the output of a console that feeds a recorder."),
2038 quiz::Quiz("Linear FM","This is often the preferred input response for frequency mod\
2039 ulating (FM'ing) an oscillator, as the result stays in tune while you change the mod\
2040 ulator."),
2041 quiz::Quiz("Linear Power Supply","A linear power supply design takes a higher incomi\
2042 ng voltage and reduces it to a lower voltage using components such as transformers. \
2043 In very general terms, they tend to introduce less noise into the output power signa\
2044 l, at the cost of increased heat and weight (they're not very efficient). Many are m\
2045 oving to a hybrid power supply that combines a switcher with a small linear supply o\
2046 r regulator to get the best of both worlds."),
2047 quiz::Quiz("Linear VCA","A linear voltage-controlled amplifier (VCA) uses a simple m\
2048 athematical relationship between control voltage input and signal level output – for\
2049 example, 50% of nominal control voltage in would result in the output signal being \
2050 at 50% of the level of the input signal. This, however, is not how our ears perceive\
2051 loudness; a sound must be amplified by 10x in order to be perceived as twice as lou\
2052 d. This makes a linear VCA desirable for scaling control voltages, but perhaps less \
2053 so for scaling audio signals. If you connect an envelope generator with an exponenti\
2054 al output to a linear VCA, then you will get the desired aural result. Confusing? Th\
2055 at's why it's great when an envelope generator or VCA has a switch or control to var\
2056 y it between linear and exponential response. A linear mixer is similar to a linear \
2057 VCA: "half" on the input level control equals the output having half the voltage swi\
2058 ng as the input. Again, this is fine for altering control voltages, but not for mixi\
2059 ng audio signals; in that case you want a mixer with exponential controls."),
2060 quiz::Quiz("Linear VCO","A linear voltage-controlled oscillator (VCO) follows the vo\
2061 lts/hertz (v/Hz) standard; more common is the exponential volts/octave (v/oct) stand\
2062 ard. The exception is frequency modulation (FM), where a linear control voltage inpu\
2063 t is often preferred to recreate classic style FM as it does not change the fundamen\
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2064 tal pitch of the carrier oscillator."),
2065 quiz::Quiz("Live Recording","A recording session where all the musicians are playing\
2066 at once with no overdubbing."),
2067 quiz::Quiz("Live Room","The large, main room of the recording studio where most of t\
2068 he instruments and/or vocalists perform. So called, not just because there is room f\
2069 or live performances, but because the room has been acoustically treated to produce \
2070 a pleasing amount of live reverberation."),
2071 quiz::Quiz("Live","1) A term describing a space with a reverberant or reflected soun\
2072 d. In a “live” space, the sound waves are active or “live.” 2) Occurring in real ti\
2073 me, as opposed to previously recorded."),
2074 quiz::Quiz("Local On/Off","Local On/Off – A MIDI message that controls the internal \
2075 sound module of a synthesizer or MIDI controller. “Local On” triggers the internal m\
2076 odule when the keyboard is played; “Local Off” disconnects it. “Local Off” is freque\
2077 ntly used to prevent unwanted looping of MIDI messages in some configurations, or wh\
2078 en controlling the internal module via another controller."),
2079 quiz::Quiz("Logic Functions","In a modular synth, control voltages tend to be contin\
2080 uous in nature, while gate and trigger signals are binary: on or off; high or low. T\
2081 his is the same as logic signals in digital circuitry. Therefore, some make digital \
2082 logic modules. A common logic function is OR: If either signal A or signal B is high\
2083 (on), then output a high gate signal (on); otherwise output a low gate (off). Anoth\
2084 er is AND: If and only if signal A and signal B are both, then output a high gate (o\
2085 n); otherwise, output a low gate (off). These are great functions for combining beat\
2086 triggers from different timing sources."),
2087 quiz::Quiz("Logic","Binary or Boolean logic is a way of combining gate signals (on o\
2088 r off voltages) to create new outputs. Each section of a logic module typically incl\
2089 udes 1 to 3 inputs, with 2 being the most common. An OR function says if there is a \
2090 gate on (or “high”) signal at any of the inputs (i.e. input 1 or input 2 or input 3,\
2091 etc.), to output a gate on signal. An AND function says only output a gate on signa\
2092 l if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all ha\
2093 ve gate ons). Adding an “N” to the front of a function’s name says “not” this functi\
2094 on – in other words, a NOR function would only output a high signal if all inputs we\
2095 re low (not input 1 nor input 2 are high)."),
2096 quiz::Quiz("Loop","1) Effectively, any piece of music or data that repeats endlessly\
2097 . Before digital audio and sampling, loops were created by looping tape. Today, loop\
2098 s are used in samples to sustain a sampled note for as long as the note is triggered\
2099 , while drum loops and other music loops are common in modern music production. 2) A\
2100 nother term for antinode, or the points of maximum displacement of motion in a vibra\
2101 ting stretched string or a sound wave. (See also “Standing Wave.”)"),
2102 quiz::Quiz("Looping","Sometimes it’s useful to have a module loop or repeat its func\
2103 tions. For example, an envelope generator that can be set to loop becomes a low freq\
2104 uency oscillator: as it attacks to a maximum value and decays back to zero, it start\
2105 s that attack phase again. Quite often you want a note sequencer to loop: When it re\
2106 aches the last note in the sequence, it would be useful for it to then look back to \
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2107 or return to the first note and start over. Audio recorders with looping features ar\  
2108 e also popular for live performance."),  
2109 quiz::Quiz("Loudness","A term referring to how the human ear perceives incoming soun\  
2110 d waves. This term seems self-explanatory, but it's deceptive. We commonly think of \  
2111 loudness as it relates to the volume of a sound, but this is an indirect relationshi\  
2112 p. In acoustic terms, volume is more about the amplitude of the sound waves, while l\  
2113 oudness describes how our ears hear the intensity of those waves."),  
2114 quiz::Quiz("Low (gate)","Most often, this is shorthand for saying a gate or trigger \  
2115 signal is in its "off" condition (typically 0 or -5 volts, in contrast to a "high" o\  
2116 r "on" signal of +5 volts)."),  
2117 quiz::Quiz("Low Frequency Oscillator","This module produces repetitive, cycling wave\  
2118 s ranging in frequency from the low end of the audio spectrum to as slow as many sec\  
2119 onds or even minutes per cycle. They are used to produce effects such as tremolo (wh\  
2120 en controlling the loudness of a signal), vibrato (when controlling the pitch of a s\  
2121 ignal), repetitive filter wah-wah effects, pulse width modulation to vary the wavesh\  
2122 ape of a pulse in an oscillator, and more."),  
2123 quiz::Quiz("Low Impedance","(abbreviated Lo-Z) Described as impedance of 500 ohms or\  
2124 less. (See also "Impedance.")"),  
2125 quiz::Quiz("Low Pass Filter","The low pass filter (LPF) design passes harmonics belo\  
2126 w its cutoff or corner frequency untouched, and reduces the level of lower harmonics\  
2127 depending on how far above the cutoff they are. In a 12dB/oct (decibel/octave) low \  
2128 pass filter, harmonics one octave above the cutoff frequency (in other words, double\  
2129 cutoff frequency) are reduced in level by 12 dB; harmonics two octaves above the cu\  
2130 toff (four times the frequency) are reduced by 24dB, and so forth. This is the most \  
2131 common type of filter used, as most natural sounds have stronger low harmonics and w\  
2132 eaker high harmonics - especially as a note fades to silence."),  
2133 quiz::Quiz("Low Pass Gate","By strict definition, a low pass gate (LPG) is a low pas\  
2134 s filter whose cutoff frequency goes down into the subsonic range as its control vol\  
2135 tage goes towards 0 volts, resulting in the input signal being filtered almost into \  
2136 silence. Some replicate this by combining a low pass filter and a voltage controlled\  
2137 amplifier into the same module, with both following the same control voltage. In ei\  
2138 ther case, as an input envelope falls from a high level to 0 volts, the output gets \  
2139 duller (higher harmonics are filtered more) as it falls to silence. This mimics the \  
2140 way many natural sounds work."),  
2141 quiz::Quiz("Low-Frequency Oscillator (LFO)","A circuit that emits low-frequency elec\  
2142 tronic waveforms below the audible level of human hearing (20 Hz or less). This low-\  
2143 frequency waveform creates a rhythmic pulse that is used to modulate various paramet\  
2144 ers in the audio signal, such as pitch or volume. LFOs are frequently used in sampl\  
2145 rs, synthesizers and signal processors to create such effects as vibrato, tremolo, a\  
2146 nd phasing."),  
2147 quiz::Quiz("low-pass-filter","An audio filter or device that attenuates signals abov\  
2148 e a certain frequency (the cut-off frequency) and passes signals with frequencies th\  
2149 at are lower than the cut-off."),
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2150 quiz::Quiz("Lows or Low-End", "Short for “low frequencies,” loosely referring to bass\  
2151 -frequency signals below 250 Hz. Usually meant in the context of “highs, mids and lo\  
2152 ws” in an audio signal."),  
2153 quiz::Quiz("LPF", "The low pass filter (LPF) design passes harmonics below its cutoff\  
2154 or corner frequency untouched, and reduces the level of lower harmonics depending o\  
2155 n how far above the cutoff they are. In a 12dB/oct (decibel/octave) low pass filter,\  
2156 harmonics one octave above the cutoff frequency (in other words, double cutoff freq\  
2157 uency) are reduced in level by 12 dB; harmonics two octaves above the cutoff (four t\  
2158 imes the frequency) are reduced by 24dB, and so forth. This is the most common type \  
2159 of filter used, as most natural sounds have stronger low harmonics and weaker high h\  
2160 armonics – especially as a note fades to silence."),  
2161 quiz::Quiz("LPG", "By strict definition, a low pass gate (LPG) is a low pass filter w\  
2162 hose cutoff frequency goes down into the subsonic range as its control voltage goes \  
2163 towards 0 volts, resulting in the input signal being filtered almost into silence. S\  
2164 ome replicate this by combining a low pass filter and a voltage controlled amplifier\  
2165 into the same module, with both following the same control voltage. In either case,\  
2166 as an input envelope falls from a high level to 0 volts, the output gets duller (hi\  
2167 gher harmonics are filtered more) as it falls to silence. This mimics the way many n\  
2168 atural sounds work."),  
2169 quiz::Quiz("M2.5", "A common screw thread size used to mount Eurorack modules. This s\  
2170 ize is most common when using a system of loose nuts that slide along the rails that\  
2171 the modules are attached to."),  
2172 quiz::Quiz("M3", "A common screw thread size used to mount Eurorack modules. This siz\  
2173 e is most common when using module mounting rails that have been pre-drilled."),  
2174 quiz::Quiz("Magnetic Tape", "Recording tape consisting of a plastic strip coated by m\  
2175 agnetic materials, finely ground iron oxide (rust) particles. Commonly used for anal\  
2176 og recording."),  
2177 quiz::Quiz("Magnetism", "A natural attractive energy of iron based-materials toward o\  
2178 ther iron-based materials."),  
2179 quiz::Quiz("MArF", "The rare Buchla Model 248 MArF (Multiple Arbitrary Function Gener\  
2180 ator) is a cross between a sequencer and an envelope generator (both described elsew\  
2181 here in this glossary) in that it typically contains 16 or 32 stages (sometimes refe\  
2182 rred to as “segments”), and a rate control to interpolate between these stages. This\  
2183 means very complex envelope shapes and other control voltage sequences can be creat\  
2184 ed. Later on, Buchla used the term MARF to describe the multi-step envelopes in inst\  
2185 ruments such as the Buchla 400."),  
2186 quiz::Quiz("Margin", "See “Headroom.”"),  
2187 quiz::Quiz("Masking", "The characteristic of hearing by which loud sounds prevent the\  
2188 ear from hearing softer sounds of similar frequency. Also refers to the obscuring o\  
2189 f softer sounds by louder ones."),  
2190 quiz::Quiz("Master", "1) The main output control of a console or DAW, setting the lev\  
2191 el of the mixed signal as it leaves the console. (Also called “master fader.”) 2) Th\  
2192 e final-mixed original recording from which copies are made."),
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2193 quiz::Quiz("Mastering","The final process of fine-tuning and “sweetening” the mix on\  
2194 a song or collection of songs, from which the master will be created."),  
2195 quiz::Quiz("Measure","The grouping of a number of beats in music. (See also “Bar.”)\  
2196 ),  
2197 quiz::Quiz("Meg","A slang abbreviation based on the prefix “Mega-, meaning 1,000,000\  
2198 . Often used as shorthand for megahertz (1,000,000 Hertz, Mhz) or megabytes (1,000,0\  
2199 00 bytes, MB)."),  
2200 quiz::Quiz("Meter","1) A device that measures and displays the signal level in audio\  
2201 or digital equipment. Meters usually measure peak values or RMS values. (See also “\  
2202 Peak Value,””RMS Value.”) 2) The rhythmic structure of music, typically describing t\  
2203 he number of beats in a measure."),  
2204 quiz::Quiz("Mic / Line Switch","Mic, Mike - Abbreviations for “microphone.”"),  
2205 quiz::Quiz("Microphone (Mic) Input","The input of a console or other device designat\  
2206 ed for a microphone signal."),  
2207 quiz::Quiz("Microphone (Mic) Level","The very low audio voltage level emitted by a s\  
2208 tudio microphone. The signal must go through a preamplifier to be increased to line \  
2209 level before entering the console. (See also “Line Level,” “Preamplifier.”)"),  
2210 quiz::Quiz("Microphone (Mic) Pad","A setting on a microphone or preamp, or a separat\  
2211 e adapter/connector, that reduces the level of the microphone signal before it enter\  
2212 s the preamplifier to prevent overload."),  
2213 quiz::Quiz("Microphone","A transducer which converts sound pressure waves into elect\  
2214 rical signals."),  
2215 quiz::Quiz("Mid-Side Miking (M/S)","(Abbreviated M/S) A stereo coincident microphone\  
2216 placement technique in which one cardioid pattern microphone is aimed directly at t\  
2217 he sound source, and a bi-directional microphone placed sideways and as close as pos\  
2218 sible to the first mic."),  
2219 quiz::Quiz("MIDI Clock","A clock signal conveyed by MIDI that is used by the connect\  
2220 ed sequencers and musical devices to stay in sync with one another. Not to be confus\  
2221 ed with MIDI time code (MTC), MIDI clock is tied to the Beats-Per-Minute (BPM) tempo\  
2222 , advancing 24 steps per quarter note."),  
2223 quiz::Quiz("MIDI Controller","Can refer to two different elements of MIDI, depending\  
2224 on the context. 1) A device or software that sends MIDI data to connected devices, \  
2225 either through pre-programmed sequencing or through live performance by a musician. \  
2226 2) Any of a number of smaller controls on a MIDI device that is assigned to control \  
2227 specific parameters of the sound or performance."),  
2228 quiz::Quiz("MIDI Interface","A device that converts a MIDI signal into the digital f\  
2229 ormat of a computer so it can store and use the MIDI signal."),  
2230 quiz::Quiz("MIDI over Bluetooth","Bluetooth Low Energy (BLE) is a wireless connectio\  
2231 n specification supported by the majority of mobile computing devices. BLE (also cal\  
2232 led Bluetooth SMART) can extend battery life for mobile devices using connected acce\  
2233 ssories (such as MIDI keyboards and controllers) that don't continuously stream data\  
2234 . An MMA Working Group evaluated Bluetooth LE MIDI performance (latency and jitter) \  
2235 and decided on a specification for MIDI over Bluetooth which would enable products f\
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2236 rom different manufacturers to interoperate. The Specification for MIDI over Bluetoo\
2237 th Low Energy (BLE-MIDI) is based on Apple's implementation which appeared in iOS8 a\
2238 nd OSX 10.10, so that products from early adopters would remain compatible with the \
2239 industry standard."),
2240 quiz::Quiz("MIDI Sample Dump Standard (SDS)","A sub-protocol that was added into MID\
2241 I to enable the transfer of digitally recorded samples between instruments, storage \
2242 units or sound modules without converting them to analog."),
2243 quiz::Quiz("MIDI Sequencer","A device or software that can record and play back MIDI\
2244 data, controlling the performance of MIDI musical instruments or devices in a serie\
2245 s of timed steps. MIDI sequencers can exist on board MIDI controllers, keyboards or \
2246 workstations, as standalone devices, or as computer software."),
2247 quiz::Quiz("MIDI Thru Box","A unit with one MIDI In Port and several MIDI Thru Ports\
2248 to relay the MIDI signal to multiple devices. MIDI users often prefer this as an al\
2249 ternative to “daisy chaining” devices, which can cause slight delays in the MIDI sig\
2250 nal."),
2251 quiz::Quiz("MIDI Thru","A port that puts out a MIDI signal that is the same as the i\
2252 ncoming MIDI signal, effectively relaying the signal to another device without alter\
2253 ing or changing it. (Many MIDI devices have three MIDI ports: In, Out and Thru.)"),
2254 quiz::Quiz("MIDI Time Code (MTC)","The translation of the information in SMPTE time \
2255 code into MIDI data, enabling MIDI sequencers and connected devices to sync with SMT\
2256 PE code (usually in relation to video). (See also “SMPTE Time Code.”)"),
2257 quiz::Quiz("MIDI","Short for Musical Instrument Digital Interface. MIDI is a common \
2258 language to connect one synthesizer to another, and synthesizers to a computer. Alth\
2259 ough it is a digital language, it is easy to buy a MIDI to CV/Gate (control voltage \
2260 and gate) converter module that handles both note events and MIDI clocks for driving\
2261 sequencers and the such. The biggest thing to watch out for is what type of connect\
2262 or is required: the traditional 5-pin DIN, or a USB computer-style connection."),
2263 quiz::Quiz("Mids","Abbreviation for “mid-range frequencies,” the audio frequencies f\
2264 rom about 250 Hz through 6000 Hz. Meant in the context of “highs, mids and lows” in \
2265 an audio signal."),
2266 quiz::Quiz("Mini Keys","A number of keyboard controllers and even keyboard synths us\
2267 e a key size that is much smaller than a typical piano key. Mini keys is the term co\
2268 mmonly used (sometimes derisively, although the space and cost savings can be quite \
2269 significant) to refer to this hardware choice."),
2270 quiz::Quiz("Mix Down","Mixdown or Mix Down – The processes of creating a final mix b\
2271 y combining multiple audio tracks into a single track (or two-channel stereo track) \
2272 prior to the mastering stage. This can include the traditional method of mixing the \
2273 multiple channels of analog tape into a two-track master, or the more modern method \
2274 of creating a digital mixdown using a DAW (which is often referred to as “rendering”\
2275 )."),
2276 quiz::Quiz("Mix","1) The blending of audio signals together into one composite signa\
2277 l. 2) Can also refer to the blending of a portion of an effected audio signal back i\
2278 nto the direct signal."),
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2279 quiz::Quiz("Mixer","This module combines signals together. You may use a mixer to co\
2280 mbine audio signals, in which case you may want one with exponential level controls \
2281 and perhaps stereo panning, or to combine control voltages, in which case you may wa\
2282 nt linear level controls plus additional functions to invert and offset the voltages\
2283 going through it."),
2284 quiz::Quiz("Modular","A modular synth breaks down the main components of a synthesiz\
2285 er - the tone-generating oscillators, the tone-modifying filters, the amplitude-shap\
2286 ing VCAs, and the modulation sources that create envelopes, tremolos, and more - int\
2287 o individual modules you can purchase and install. At the most basic level, this all\
2288 ows you to play mix-and-match in building your own custom synth."),
2289 quiz::Quiz("Modulation Noise","Noise that is present only when the audio signal is p\
2290 resent."),
2291 quiz::Quiz("Modulation","When you vary a parameter of a synthesizer module using vol\
2292 tage control, it is said that you're modulating that parameter. For example, when a \
2293 low frequency oscillator (LFO) varies the cutoff frequency of a filter to create a w\
2294 ah-wah effect, it is said that the LFO is modulating the cutoff. When an envelope ge\
2295 nerator causes a voltage controlled amplifier (VCA) to open up to allow a sound to b\
2296 ecome suddenly loud, and then fades it back down to silence, you can also say the en\
2297 velope is modulating the amp (although some like to restrict the term "modulate" to \
2298 a repetitive action). Therefore, we call the sources of these changes modulators."),
2299 quiz::Quiz("Modulator","We touched on the general subject of modulation and modulato\
2300 rs in the definition above. However, quite often when someone uses the term modulato\
2301 r, they're usually discussing a synthesis techniques where one usually audio-rate si\
2302 gnal "modulates" (varies) another audio signal. For example, in frequency modulation\
2303 (FM) synthesis, the modulator (or modulating oscillator) varies the frequency (pitc\
2304 h) of the main signal generator (oscillator), called the carrier. In ring, balanced,\
2305 or amplitude modulation, the modulator is varying the loudness of the carrier signa\
2306 l. So the term modulator is a way to make it clear which component you're talking ab\
2307 out in one of these patches: not the main tone generator, but the module that is dri\
2308 ving that generator crazy."),
2309 quiz::Quiz("Module","A self-contained group of circuits and controls. In the recordi\
2310 ng studio, modules are often contained in interchangeable housing for installation o\
2311 n rack mounts, and can include amplifiers, equalizers, effects processors and sound \
2312 modules (MIDI instruments to be activated by an external controller). In the digital\
2313 space, plug-ins, software synths, samplers and plug-ins are also described as modul\
2314 es."),
2315 quiz::Quiz("Monaural (Mono)","(Abbreviated "Mono") Describing an audio signal coming\
2316 through a single, as opposed to stereo, which is two channels. (See also "Monophoni\
2317 c.")"),
2318 quiz::Quiz("Monitor Mix","A mix of the live and/or recorded audio signals that is fe\
2319 d to the musicians so the can hear the music while performing, whether live onstage \
2320 or in the studio. Monitor mixes are on a separate signal path from the main mix (oft\
2321 en controlled by a separate, smaller console) and do not affect the FOH mix (in live\
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2322 audio) or the signal going into the multitrack recorder/DAW. In live performance se\
2323 ttings, the monitor mix is often controlled by a separate audio engineer running a s\
2324 eparate sound board."),
2325 quiz::Quiz("Monitor Mixer Section","Monitor Section/Monitor Mixer Section – The sect\
2326 ion of the console that is used to create a rough mix so the engineer can hear what \
2327 is being recorded without effecting the levels being fed to the multitrack recorder \
2328 or DAW."),
2329 quiz::Quiz("Monitor Path","A signal path separate from the channel path that allows \
2330 the engineer to listen to what is being recorded without affecting the signal being \
2331 fed to the multitrack recorder or DAW. (See also “Channel Path.”)"),
2332 quiz::Quiz("Monitor","1) To listen to the music for the purpose of checking quality \
2333 or avoiding peaks. 2) A speaker in the studio (usually one of a pair) that is used t\
2334 o listen to the audio signals. This can include studio monitors in the control room \
2335 for listening to the mix, and headphones in the booths or live room for the performe\
2336 rs to hear a mix of the tracks while they are performing."),
2337 quiz::Quiz("Monophonic","(Abbreviated “Mono”) 1) A single sound source or single-cha\
2338 nnel transmission (as opposed to stereo). 2) A melody line in which only one note at\
2339 a time is played. 3) Describing an instrument or synthesizer setting that only play\
2340 s one pitch (or “voice”) at a time. (See also “Voice.”)"),
2341 quiz::Quiz("Morphing","In the context of a modular synth, morphing refers to an osci\
2342 llator that can more or less smoothly change the shape of its output waveform – and \
2343 therefore, the resulting sound – as you play it. This is usually the domain of digit\
2344 al oscillators which internally crossfade (or in some cases, switch) from one wavesh\
2345 ape to another, although it is sometimes applied to analog oscillators that give you\
2346 real time control over waveshapes."),
2347 quiz::Quiz("Mother-32","A very popular semi-modular synthesizer by Moog. It comes in\
2348 its own case, but can be mounted in a Eurorack-format case. It comes with one VCO (\
2349 sawtooth and pulse waveforms), one LFO (triangle and square waveforms), one Moog-sty\
2350 le transistor ladder filter that can be low pass or high pass, and one AD or AR enve\
2351 lope generator. It also has a very capable step sequencer plus a miniature one-octav\
2352 e keyboard. What makes it a semi-modular is a nice patch panel that allows alternat\
2353 e routings for the way the synth voice is internally wired, and for it to be patched \
2354 to external modules. As so many of these were sold, I’m using it as a representative\
2355 of a typical semi-modular or “starter” synthesizer voice when discussing how to exp\
2356 and a basic modular system. I have an online introductory course to the Mother-32 co\
2357 ming out this spring, and will have a course plus ongoing weekly series on adding di\
2358 fferent modules to this starter system."),
2359 quiz::Quiz("Moving Coil Microphone","A microphone in which sound pressure waves are \
2360 converted to an electrical audio signal by an induction coil moving within a magneti\
2361 c field—a process often compared to a loudspeaker working in reverse. Dynamic microp\
2362 hones are less sensitive than condenser microphones, but can be effective for miking\
2363 louder sound sources or for close-miking applications."),
2364 quiz::Quiz("Moving Fader Automation","A feature in some consoles in which fader chan\
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2365 ges can be pre-programmed to occur automatically during playback of a multitrack rec\
2366 ording."),
2367 quiz::Quiz("MU", "Refers to modules that are 5U (rack units) or 8.75" (22.2 cm) high,\
2368   which is most often associated with the vintage Moog standard and those who have fo\
2369 llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
2370 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
2371 fers to a standardized width of 2.125" (5.4 cm) wide per MU. Given that this standar\
2372 d is both historical and physically large, some users "5U" as a badge of honor that \
2373 they're traditional and cool. (And the are.) There was also a briefly popular 5U for\
2374 mat from MOTM that used a different width and power connection. It has since been di\
2375 scontinued, but there are still diehard MOTM format users today."),
2376 quiz::Quiz("Multi-Tap Delay", "A delay works by in essence putting audio in one end o\
2377 f a pipe and grabbing it again when it comes out the other. A multi-tap delay says "\
2378 Why wait until the audio snapshots go all the way through the pipe? Let's grab it wh\
2379 en it's only part way through the pipe." Those points where it's prematurely grabbed\
2380 are the "taps" - kind of like additional water taps added along a long pipe."),
2381 quiz::Quiz("Multimeter", "A small device that tests electrical voltage, current, and \
2382 resistance. Multimeters are useful in recording studios for calibrating electrical s\
2383 ystems and troubleshooting problems."),
2384 quiz::Quiz("Multiple Arbitrary Function Generator", "The rare Buchla Model 248 MARF (\
2385 Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelop\
2386 e generator (both described elsewhere in this glossary) in that it typically contain\
2387 s 16 or 32 stages (sometimes referred to as "segments"), and a rate control to inter\
2388 polate between these stages. This means very complex envelope shapes and other contr\
2389 ol voltage sequences can be created. Later on, Buchla used the term MARF to describe\
2390 the multi-step envelopes in instruments such as the Buchla 400."),
2391 quiz::Quiz("Multiple", "Quite often you need to split or copy a signal to send to mor\
2392 e than one destination. This is commonly done with a multiple ("mult" for short) whe\
2393 re you plug one source in, and then plug in additional patch cables to go off to mul\
2394 tiple destinations."),
2395 quiz::Quiz("Multiplexer", "Multiplexing is a technical way to describe signal routing\
2396 , where multiple signals may be routed to one destination. In synth modules, this is\
2397 usually extended to include the possibility of one input being switched between mult\
2398 iple outputs. A sequential switch is a type of multiplexor, as it chooses among mult\
2399 iple inputs to decide which one to send to the output (or the other way around). The\
2400 re are some modules that do this at audio rate, using an oscillator's output to swit\
2401 ch between variations of another waveshape to create complex, chopped mixtures of th\
2402 ose waveforms."),
2403 quiz::Quiz("Multitimbral", "Refers to the ability of a synthesizer or module to play \
2404 several different sounds, patches or "timbres" at once."),
2405 quiz::Quiz("Multitrack Recording", "Also called tracking or multitracking) The heartb\
2406 eat of the recording studio, multitrack recording is process of recording a collecti\
2407 ve of sound sources onto separate tracks, each with its own audio channel, then comb\
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2408 ining the tracks to play back simultaneously. Recording can be done either one track\  
2409 or instrument at a time (to be combined later) or by recording the performers onto \  
2410 separate tracks as they play together live. These signals were originally recorded o\  
2411 nto multitrack analog tape, but today they can also be recorded digitally as separat\  
2412 e audio files into a digital audio workstation (DAW)."),  
2413 quiz::Quiz("Multitrack Tape","A piece/reel of magnetic tape which can be used to sto\  
2414 re two or more discrete signals in sync with each other."),  
2415 quiz::Quiz("Musical Instrument Digital Interface (MIDI)","Short for Musical Instrume\  
2416 nt Digital Interface. MIDI is a common language to connect one synthesizer to anothe\  
2417 r, and synthesizers to a computer. Although it is a digital language, it is easy to \  
2418 buy a MIDI to CV/Gate (control voltage and gate) converter module that handles both \  
2419 note events and MIDI clocks for driving sequencers and the such. The biggest thing t\  
2420 o watch out for is what type of connector is required: the traditional 5-pin DIN, or\  
2421 a USB computer-style connection."),  
2422 quiz::Quiz("Mute Switch","A switch on a console or other piece of audio equipment th\  
2423 at turns off the input or output, or a matching button on the virtual audio control \  
2424 space of a DAW. The individual channels on a console each have a mute switch that ca\  
2425 n cut the signal for that channel."),  
2426 quiz::Quiz("Mute","Sometimes you need to silence or disconnect a signal. A circuit t\  
2427 hat allows you to do so is called a mute."),  
2428 quiz::Quiz("Nanowebers per Meter (NW/m)","The standard unit in measuring the amount \  
2429 of magnetic strength on analog tape. A Weber is a unit of magnetic strength, but it \  
2430 is too large a unit to apply to the magnetism in tape recorders, so nanowebers is us\  
2431 ed instead. Nanowebers per meter of tape effectively describes the signal strength t\  
2432 hat is being recorded to tape."),  
2433 quiz::Quiz("Narrowband Noise","Noise (random energy) that occurs over a limited freq\  
2434 uency range."),  
2435 quiz::Quiz("Near Field","The area between 1-5 feet from the sound source. Studio mon\  
2436 itors are generally considered "near-field" speakers because they are meant to be li\  
2437 stened to at close range. (See also "Far Field.")"),  
2438 quiz::Quiz("Near-Coincident Miking","A stereo miking technique in which two micropho\  
2439 nes are placed near each other at an outward angle to create a stereo image (as oppo\  
2440 sed to "Coincident Miking" which angles the microphones toward each other). Common \  
2441 versions of near-coincident miking include DIN stereo (90-degree angle, 20cm apart),\  
2442 NOS stereo (90-degree angle, 30 cm apart) and ORTF (110-degree angle, 17 cm apart).\  
2443 "),  
2444 quiz::Quiz("Negative Feedback","A portion of the output signal that is fed back to t\  
2445 he input of an amplifier with its phase inverted from the original output signal. Th\  
2446 is has a dampening effect on the output, effectively cancelling out a portion of the\  
2447 volume."),  
2448 quiz::Quiz("Noise Floor","The level of the noise present below the audio signal, mea\  
2449 sured in dB. Every electronic device emits a minimum level of noise, even when no au\  
2450 dio is traveling through it; this is described as its noise floor. Generally speakin\
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2451 g, the lower the noise floor in these devices, the higher the quality of the device.\
2452 The noise floor also translates to the recorded signal; the noise floor of a record\
2453 ing is the sum of all the noise generated by connected devices. The objective is alw\
2454 ays to keep the noise floor as low as possible."),
2455 quiz::Quiz("Noise Gate","A gate that is used reduce audible noise by automatically t\
2456 urning off an audio channel when the signal is not present."),
2457 quiz::Quiz("Noise Reduction","Any of a number of processes to remove noise from a si\
2458 gnal, device or system."),
2459 quiz::Quiz("Noise","Describes any unpleasant, objectionable or unintended sound freq\
2460 uencies present in the audio signal. All electronic equipment produces some type of \
2461 noise, which may be described as a hiss or buzz that can be heard during quiet or ot\
2462 herwise silent passages. (See also "Noise Floor.") Bad connections, improper groundi\
2463 ng, radio interference and other issues can also cause introduce noise into the sign\
2464 al. Engineers may also deliberately run a noise signal through a sound system for te\
2465 sting purposes. (See also "White Noise, "Pink Noise.")"),
2466 quiz::Quiz("Non-destructive Editing","A feature in recording systems (most common in\
2467 Digital Audio Workstations, or DAWs) in which the original signal or content stays \
2468 intact while edits are performed, allowing the engineer to revert to the original ve\
2469 rsion at any time. (Sometimes also called "Nonlinear editing.")"),
2470 quiz::Quiz("Nondirectional","In microphones, picking up evenly from all directions."\
2471 ),
2472 quiz::Quiz("Normalize","To apply a fixed amount of gain to audio so that the highest\
2473 peak is set at the highest acceptable recording level."),
2474 quiz::Quiz("Normalled","The power of modular synthesizers is that you can patch a si\
2475 gnal to flow the way you prefer through your system. This can also be a time-consumi\
2476 ng bummer when you're just trying to patch a "typical" signal flow. Therefore, some \
2477 manufacturers have created "semi-modular" synths that have all of these typical conn\
2478 ections pre-wired for you, with the important feature that many of these wirings can\
2479 be overridden by inserting patch cables into the correct jacks. These pre-wired con\
2480 nections are often referred to as being normalled. For example: An internal noise so\
2481 urce may normally be connected to one channel of a mixer that appears before the fil\
2482 ter, but if you insert a patch cable into a jack usually labeled external input, thi\
2483 s "normalled" connection is broken and replaced by your external connection."),
2484 quiz::Quiz("Notch Filter","This is a particular type of filter mode where audio freq\
2485 uencies or harmonics around the corner or cutoff frequency setting are removed, nor \
2486 "notched out" of the overall spectrum. It is the opposite of a bandpass filter, whic\
2487 h only passes harmonics around the cutoff frequency. Notch filters tend to have a su\
2488 btle effect on the sound; moving (modulating) the cutoff frequency can result in a w\
2489 eak phasing sort of sound. Notch filters are often used in sound systems to weaken o\
2490 r remove a problematic frequency, such as ground loop hum, a resonance in a room, or\
2491 other annoying peak in the harmonic spectrum of a sound. Think of using a notch fil\
2492 ter in a patch to hollow out a sound, leaving room in the harmonic spectrum for othe\
2493 r sounds to exist with less competition, or just to create a sound more likely to ca\
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2494 tch the ear because something that is expected is instead missing."),
2495 quiz::Quiz("Notch","A narrow band of audio frequencies."),
2496 quiz::Quiz("NW/m","The standard unit in measuring the amount of magnetic strength on\
2497 analog tape. A Weber is a unit of magnetic strength, but it is too large a unit to \
2498 apply to the magnetism in tape recorders, so nanowebers is used instead. Nanowebers \
2499 per meter of tape effectively describes the signal strength that is being recorded t\
2500 o tape."),
2501 quiz::Quiz("Nybble","Nybble (or Nibble) - One half byte of computer data, or 4 bits.\
2502 "),
2503 quiz::Quiz("Nyquist Frequency","In digital recording, the highest frequency that can\
2504 be recorded and reproduced properly, equivalent to a one-half the sampling rate. (F\
2505 or example, with the common sampling rate of 44,100 kHz per second, the Nyquist freq\
2506 uency would be 22,050 kHz.) Aliasing begins to occur with frequencies that exceed th\
2507 is threshold. (See also "Aliasing.")"),
2508 quiz::Quiz("Nyquist Rate","he lowest sampling rate that can be used to record and re\
2509 produce a given audio signal, equivalent to twice the highest frequency. If the high\
2510 est frequency found in an analog signal or sound is 18,000 kHz, theoretically the si\
2511 gnal must be sampled at a minimum of 36,000 kHz per second—otherwise, the signal is \
2512 considered to be undersampled and aliasing will occur. This is essentially the inver\
2513 se principle of the Nyquist Frequency. (NOTE: the sample rate of 44,100 kHz/second i\
2514 s considered the standard sample rate because it easily covers the upper range of hu\
2515 man hearing, which is about 20,000 kHz.)"),
2516 quiz::Quiz("Octave Divider","A module that creates a new tone one or two octaves bel\
2517 ow the fundamental harmonic - the "pitch" - of the sound coming into it, to emphasiz\
2518 e the bass. Sometimes also known as a suboctave or sub bass function."),
2519 quiz::Quiz("Octave","An octave is a typical musical interval. For example, all of th\
2520 e "C" notes on a keyboard are octaves apart from each other. To play a note that is \
2521 one octave higher in tuning, you need to double its pitch; to play an octave lower, \
2522 you need to cut the pitch in half. In patch terms, this typically means adding or su\
2523 btracting 1 volt to get a one octave change in pitch; some oscillators also have oct\
2524 ave switches on their front panels that add or subtract these voltages for you (all \
2525 they are not always perfectly accurate; you often need to re-tune after switching oc\
2526 taves). Suboctave or subharmonic generators divide the input pitch by 2 or 4 to crea\
2527 te new waveforms that are one or two octaves lower in pitch, which adds bass."),
2528 quiz::Quiz("Off Axis","Veering away from the imaginary line (axis) directly in front\
2529 of the receiving end of a microphone. Measured as degrees of an angle. (For example\
2530 , a sound coming from directly behind the microphone is said to be 180 degrees off-a\
2531 xis.)"),
2532 quiz::Quiz("Offset Time","1) The SMPTE time that will trigger a MIDI sequencer to be\
2533 gin. 2) The amount of position difference needed to get two reels to play the music \
2534 in time."),
2535 quiz::Quiz("Offset","In simple terms, Offset modules usually add or subtract a volta\
2536 ge from a signal passing through - such as shifting a 0 to +10v signal to instead va\
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2537 ry between -5 and +5 volts."),
2538 quiz::Quiz("Ohm's Law","The mathematical relationship between voltage, current and r\
2539 esistance."),
2540 quiz::Quiz("Ohm","The unit used to measure the amount of opposition (impedance) to e\
2541 lectrical current flow in a signal or device. (See also "Impedance.")"),
2542 quiz::Quiz("Omni Mode","A setting that enables a MIDI device to recognize and respon\
2543 d to all MIDI channels at once."),
2544 quiz::Quiz("Omni","A prefix meaning "all.""),
2545 quiz::Quiz("Omnidirectional Pattern","In microphones, picking up evenly from all dir\
2546 ections (sometimes also called "Nondirectional"). 2) In speakers, sending out the si\
2547 gnal evenly in all directions."),
2548 quiz::Quiz("On Axis","The position directly in front of the diaphragm of a microphon\
2549 e, in line with its movement."),
2550 quiz::Quiz("Open Circuit","An electrical circuit that is disconnected, interrupted o\
2551 r incomplete, preventing the flow of electricity."),
2552 quiz::Quiz("Operating Level","(Sometimes called "Reference Level") The maximum level\
2553 that should not be exceeded in normal operation."),
2554 quiz::Quiz("Operational Amplifier","(Abbreviated "Op Amp") An amplifying circuit use\
2555 d in most audio and electronic devices."),
2556 quiz::Quiz("Operational Transconductance Amplifier","An OTA (operational transconduc\
2557 tance amplifier) circuit is one that converts an input voltage to an output current.\
2558 This is a popular amplifier design as it can be less prone to going into saturation\
2559 (clipping), has good bandwidth, and is also known for a "warm" sound. Therefore, yo\
2560 u may find it in VCAs (voltage controlled amplifiers). Current can be thought of as \
2561 the inverse of resistance, so what you have in an OTA circuit is in essence a voltag\
2562 e to resistance device that makes it possible to add voltage control to circuits suc\
2563 h as filters. In general, when someone touts they have an OTA based filter, they usu\
2564 ally mean it has a "warm" sound...unless it's an MS-20 filter clone, in which case i\
2565 t's thinner and more edgy. In reality, using an OTA is more about convenience of des\
2566 ign than creating a specific sound."),
2567 quiz::Quiz("Operator","There are a few different synthesis techniques where one usua\
2568 lly audio-rate signal does something to another audio signal. For example, in freque\
2569 ncy modulation (FM), a second signal (called the modulator) varies the frequency (pi\
2570 tch) of the main signal, called the carrier. These two signals or oscillators are of\
2571 ten referred to as operators, particularly in FM patches. You're more likely to hear\
2572 this term used when working with a dedicated FM synthesizer like a Yamaha DX-7 and \
2573 its descendants, than with a modular system."),
2574 quiz::Quiz("OR function","One of the most common Boolean or binary logic functions, \
2575 OR says if there is a gate on (or "high") signal at any of the inputs (i.e. input 1 \
2576 or input 2 or input 3, etc.), to output a gate on signal. A NOR function has an inve\
2577 rted output: it would only be on (high) if all inputs were low (not input 1 nor inpu\
2578 t 2 are high). An XOR (Exclusive OR) would only output a high signal if one of the i\
2579 nputs was high, but not if both inputs were high (or low). Finally, an XNOR is the i\
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2580 nvert of an XOR function."),
2581 quiz::Quiz("Oscillator","At its core, to oscillate means to vary back and forth in a \
2582 repeating pattern. The main sound generator in a modular system is called an oscill\
2583 ator because its output varies up and down (oscillates) in voltage in a repeating pa\
2584 ttern. This pattern is referred to as its waveshape (such as a square wave, that alt\
2585 ernates between high and low voltages); how fast this pattern repeats is called its \
2586 frequency or pitch. An acoustic instrument equivalent of an oscillator is a string t\
2587 hat vibrates back and forth on a guitar, a drum head that vibrates up and down, or t\
2588 he vibrations in the reed of a woodwind instrument. The vibrations of a modular synt\
2589 h's oscillator just happen with electricity going down a wire rather than a physical\
2590 object vibrating in air. (Eventually this electricity is routed to a speaker, which\
2591 then vibrates the air with the same pattern sent to it over a wire.)"),
2592 quiz::Quiz("Oscilloscope","This is a piece of test equipment that displays voltage fl\
2593 uctuations as graphical waveforms. A 'scope can run at a wide range of frequencies,\
2594 displaying slowly changing voltages like LFOs or envelopes, or quickly changing vol\
2595 tages like oscillators and noise. Oscilloscopes used to be bulky pieces of external \
2596 equipment, but now you can get USB scopes that offload the display portion of the jo\
2597 b to your computer, or scopes as modules."),
2598 quiz::Quiz("OTA","An OTA (operational transconductance amplifier) circuit is one tha\
2599 t converts an input voltage to an output current. This is a popular amplifier design\
2600 as it can be less prone to going into saturation (clipping), has good bandwidth, an\
2601 d is also known for a "warm" sound. Therefore, you may find it in VCAs (voltage cont\
2602 rolled amplifiers). Current can be thought of as the inverse of resistance, so what \
2603 you have in an OTA circuit is in essence a voltage to resistance device that makes i\
2604 t possible to add voltage control to circuits such as filters. In general, when some\
2605 one touts they have an OTA based filter, they usually mean it has a "warm" sound...u\
2606 nless it's an MS-20 filter clone, in which case it's thinner and more edgy. In reali\
2607 ty, using an OTA is more about convenience of design than creating a specific sound.\
2608 "),
2609 quiz::Quiz("Out of Phase","1) Being similar to another signal in amplitude, frequenc\
2610 y and wave shape but being offset in time by part of a cycle. 2) Having the opposite\
2611 polarity."),
2612 quiz::Quiz("Outboard Equipment","Equipment that is used with, but is not a part of, \
2613 a console."),
2614 quiz::Quiz("Output Impedance","The opposition to the flow of electrical current by t\
2615 he output circuits of an amplifier (or other device)."),
2616 quiz::Quiz("Output Level","The signal level at the output of a device."),
2617 quiz::Quiz("Output","1) The jack or physical location of where a device sends out a \
2618 signal. 2) The signal put out by a device."),
2619 quiz::Quiz("Overdubbing","The process of recording an additional musical performance\
2620 over an existing recording, usually on its own track. Overdubbing has become a comm\
2621 on recording technique with the advent of multitrack recording, first on multitrack \
2622 analog tape, and more recently via computers and Digital Audio Workstations (DAWs)."
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2623 ),
2624 quiz::Quiz("Overtone","Any harmonic in a tone except the fundamental frequency. (See\
2625 also "Partial.")"),
2626 quiz::Quiz("Pad","1) A device or circuit that attenuates an incoming signal, usually\
2627 to prevent overload of an amplifier that follows along the signal path. (Also somet\
2628 imes called "Attenuator pad.") 2) A device with a surface that can be hit by a drum \
2629 stick; hitting the pad produces an output signal pulse (or MIDI command) that causes\
2630 a drum machine or synthesizer to sound a drum sound. 3) A type of synthesizer patch\
2631 /program used to create sustained background or atmospheric sounds."),
2632 quiz::Quiz("Pan (Panning)","The process of "placing" a particular sound within the s\
2633 tereo field. This is accomplished by controlling the balance of the signal between t\
2634 he left and right speakers so the ear hears the sound as coming from a particular po\
2635 int in the sonic space between left and right. This sonic space is sometimes called \
2636 the "stereo panorama," from which the word "panning" is derived. In surround sound, \
2637 panning occurs in a 360° sound space, not just left-right."),
2638 quiz::Quiz("Panpot (or Pan Pot)","Short for "Panoramic Potentiometer," a panpot is a\
2639 knob in the channel strip that controls the panning of the audio signal in the ster\
2640 eo (or surround) space by controlling how much of the signal is sent to each speaker\
2641 or channel."),
2642 quiz::Quiz("Parallel Jacks","Several jacks that are wired so that each connection is\
2643 wired to the corresponding connection of other jacks."),
2644 quiz::Quiz("Parallel Port","A connector that is able to transmit and receive digital\
2645 data at the same time though different pins."),
2646 quiz::Quiz("Parameter","Parameter is the fancy name given to any value or property o\
2647 r control of a synthesizer module that you're trying to change. For example, an osci\
2648 llator's parameters typically include its pitch and the width of its pulse wave. A f\
2649 ilter's parameter will include its cutoff frequency (pitch), the amount of resonance\
2650 (feedback), and possibly other controls such as a blend between its different outpu\
2651 ts. Parameter was a popular term to describe a value you could change in software, a\
2652 nd it's been carried over by some to hardware modular synths."),
2653 quiz::Quiz("Parametric Equalization","An equalizer in which all parameters of equali\
2654 zation can be adjusted to any amount, including the center frequency, the amount of \
2655 boost or cut, and the bandwidth."),
2656 quiz::Quiz("Paraphonic","A paraphonic synth is one where all of the notes being play\
2657 ed go through a single filter (VCF) and amplifier (VCA). This was a popular scheme i\
2658 n the early days of polyphonic synths in that a separate oscillator (or organ-like f\
2659 requency divider, in the case of "string synths" and the such) was used for each not\
2660 e played, but they were mixed before all going to the filter and amp to articulate t\
2661 he note(s). It was not uncommon for some monophonic synths to allow two to four inde\
2662 pendent notes to independently control the pitch of its oscillators, while still goi\
2663 ng through a single filter. This works great for chords; it doesn't always work all \
2664 that great for when a new note is played while others are being held as all of the n\
2665 otes will be re-articulated together."),
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2666 quiz::Quiz("Partial","1) Another word for overtone. 2) One of a number of sine waves\  
2667 that makes up a complex sound, helping to define the timbre. This concept is a key \  
2668 part of creating sounds in synthesizers: in additive synthesis, a number of partials\  
2669 are combined to create a certain tone."),  
2670 quiz::Quiz("Pass Band","The frequency range of signals that will be “passed” by a fi\  
2671 lter, rather than reduced."),  
2672 quiz::Quiz("Passive Device","A component that does not generate or control electrica\  
2673 l current (as opposed to an “Active Device”). In audio applications, this usually re\  
2674 fers to a piece of gear that does not include an amplifier as part of its design. Fo\  
2675 r example, active speakers are self-powered, while passive speakers require an exter\  
2676 nal amplifier in order to reproduce sound. (See also “Active Device.”)"),  
2677 quiz::Quiz("Passive","Means no active (i.e. connected to a power supply) electronics\  
2678 are involved – such as sending a signal straight through a potentiometer control, i\  
2679 nstead of using op amps and other electronics to create a mixer circuit around it. P\  
2680 assive is cheap and easy, and does not add noise to a signal. But passive electronic\  
2681 s cannot buffer one signal from another (meaning they might interact in undesirable \  
2682 ways), and cannot boost, offset, or invert a signal."),  
2683 quiz::Quiz("Patch Bay","Patch Bay (or Patchbay, Patch Field, Patch Panel) – A panel \  
2684 or component containing a series of jacks with connections for most of the inputs an\  
2685 d outputs of the console and components in the studio, used for the purpose of organ\  
2686 izing, managing and regulating signal flow."),  
2687 quiz::Quiz("Patch Cable","The cables used to connect together the different inputs a\  
2688 nd outputs in a modular synthesizer, carrying electrical control voltages and audio.\  
2689 The term came from the old telephone patch boards where an operator had to physical\  
2690 ly connect two callers together using electrical cables. As different modular format\  
2691 s often use different connector standards, you need to make sure the connectors at t\  
2692 he ends of the wire in a patch cord are the size you need (3.5mm for Eurorack, 1/4” \  
2693 for 5U/Moog Unit, or banana for Serge or Buchla control voltages)."),  
2694 quiz::Quiz("Patch Cord (or Patch Cable)","An insulated cable with plugs on each end \  
2695 used to route audio signals. Patch cords are typically thought of as short cables us\  
2696 ed to make connections in the patch bay (hence the name); however, patch cords facil\  
2697 itate almost any kind of audio connection between devices, can come in a wide range \  
2698 of lengths, and can include a number of different types of connectors."),  
2699 quiz::Quiz("Patch Field","A panel or component containing a series of jacks with con\  
2700 nections for most of the inputs and outputs of the console and components in the stu\  
2701 dio, used for the purpose of organizing, managing and regulating signal flow."),  
2702 quiz::Quiz("Patch Librarian","A computer program allowing for the storing of sound p\  
2703 atches outside of a synthesizer via MIDI."),  
2704 quiz::Quiz("Patch Panel","A panel or component containing a series of jacks with con\  
2705 nections for most of the inputs and outputs of the console and components in the stu\  
2706 dio, used for the purpose of organizing, managing and regulating signal flow."),  
2707 quiz::Quiz("Patch","The shorthand term used to refer how a series of modules are int\  
2708 erconnected to create a sound, derived from the fact that patch cords are used to co\
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2709 nnect the modules together. 1) To route or reroute the signal in an audio system (su\
2710 ch as a console) by using short cables with plugs inserted into jacks. 2) A sound se\
2711 tting or program on a synthesizer."),
2712 quiz::Quiz("Path","Short for Signal Path, the way in which current does or may trave\
2713 l in a circuit or through a device."),
2714 quiz::Quiz("PCM","Pulse Code Modulation - A process by which analog signals are tran\
2715 slated to digital code. This is done by taking samples of the amplitude of the analo\
2716 g signal at regular rapid intervals, then translating it into binary numbers as a di\
2717 gital representation of the original signal. The faster the sample rate, the better \
2718 the digital reproduction. PCM is the most common form of A/D conversion in digital a\
2719 udio."),
2720 quiz::Quiz("PD","Phase Distortion synthesis was used by Casio originally in the 80s \
2721 in the CZ line of synths. It is related to FM (frequency modulation), with enough di\
2722 fferences to avoid problems with the patent used by Yamaha's FM synths of the era. I\
2723 ntriguingly, it did a good job at mimicking many "analog" synth effects including th\
2724 e sound of a resonant filter."),
2725 quiz::Quiz("Peak Filter","An EQ circuit/filter that boosts or cuts the middle (cente\
2726 r frequencies in an audio signal, as opposed to high-pass or low-pass filters. (NOT \
2727 to be confused with amplitude peaks.)"),
2728 quiz::Quiz("Peak Meter","A meter which detects the absolute peak value of a waveform\
2729 , as opposed to the RMS value. (See also "Peak Value," "Root-Mean-Square," "RMS Mete\
2730 r.")"),
2731 quiz::Quiz("Peak to Peak Value","The measure of the total amplitude between positive\
2732 and negative peaks in an audio signal. Equal to twice the peak value for a sine wav\
2733 e. (See also "Peak Value.")"),
2734 quiz::Quiz("Peak Value","Peak Value (also called Peak Level) - The measure of the max\
2735 imum positive or negative value (amplitude) of a waveform at any moment. In audio, t\
2736 his is visually depicted as the farthest point of the waveform above or below the ze\
2737 ro axis."),
2738 quiz::Quiz("Pedal Board","A board with several guitar pedals attached and inter-conn\
2739 ected so that a guitar player can conveniently activate a number of different effect\
2740 s."),
2741 quiz::Quiz("Phantom Power","A system used to supply DC voltage to condenser mics and\
2742 other components through the audio cables, eliminating the need for external power \
2743 supplies."),
2744 quiz::Quiz("Phase Addition","The increased audio energy that happens when waveforms \
2745 are in similar phase relationships, resulting in an increase in volume up to twice w\
2746 hat it should be."),
2747 quiz::Quiz("Phase Cancellation","The opposite of phase addition, this is the reducti\
2748 on of energy that occurs when two similar waveforms that are out of phase with one a\
2749 nother and begin cancelling each other out, either greatly reducing or eliminating t\
2750 he volume. When two identical wave forms are completely out of phase (by 180 degrees\
2751 ), the result in theory is a total silencing or cancellation of the signal."),
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2752 quiz::Quiz("Phase Distortion Synthesis", "Phase Distortion synthesis was used by Casi\
2753 o originally in the 80s in the CZ line of synths. It is related to FM (frequency mod\
2754 ulation), with enough differences to avoid problems with the patent used by Yamaha's\
2755 FM synths of the era. Intriguingly, it did a good job at mimicking many "analog" sy\
2756 nth effects including the sound of a resonant filter."),
2757 quiz::Quiz("Phase Distortion", "A change in the sound because of a phase shift in the\
2758 signal. Sometimes used in synthesizers as a method of altering the wave shape or ad\
2759 ding harmonics to the sound."),
2760 quiz::Quiz("Phase Lock", "Any of a number of processes used to help synchronize signa\
2761 ls or devices by correcting phase differences. For example, in analog tape machines,\
2762 phase locking helps to keep multiple machines synced together by sensing phase diff\
2763 erences in the playback of pilot tones by the two machines and adjusting the speed t\
2764 o eliminate the phase difference. In synthesizers, phase locking controls one tone g\
2765 enerator so that it begins its waveform in phase with the signal from another tone g\
2766 enerator. Phase-locked loops (PLL) are reference signals used in the clock functions\
2767 of electronic devices."),
2768 quiz::Quiz("Phase Locked Loop", "A phase locked loop is, in essence, an oscillator th\
2769 at tries to match the frequency of – or more importantly, a division or multiple of \
2770 the frequency of – another signal. This is most commonly used to create a frequency \
2771 that is much higher than the incoming reference signal – such as a timing module tha\
2772 t can create an output clock that is 2, 4, 8, or more times the tempo of an incoming\
2773 clock, or a very high frequency oscillator that is locked to a multiple of an incom\
2774 ing pitch – perhaps to drive a special circuit such as a switched-capacitor filter." \
2775 ),
2776 quiz::Quiz("Phase Modulation", "Some would say this is the pedantically correct term \
2777 for frequency modulation (FM), as the act of causing a carrier oscillator to play ba\
2778 ck faster and slower (quickly changing its frequency to be higher and lower) is the \
2779 same as advancing and retarding position (phase) of the normal playback of a wavefor\
2780 m. But don't get bogged down by terminology when creating an FM patch; just connect \
2781 the output of one oscillator to the pitch input of another and go for it."),
2782 quiz::Quiz("Phase Reversal", "A change in a circuit to get the waveform to shift by 1\
2783 80 degrees."),
2784 quiz::Quiz("Phase Shift", "A delay introduced into an audio signal measured in degree\
2785 s delayed."),
2786 quiz::Quiz("Phase Shifter", "This effect splits a signal into two copies. One copy is\
2787 fed through an "all pass filter" which does not attenuate any of the original harmo\
2788 nics like a low pass or high pass filter does, but which does alter the phase of the\
2789 signal, causing those harmonics to have varying amounts of phase shift in relation \
2790 to the original depending on their frequency. Mix these two copies back together, an\
2791 d different harmonic components of the original sound cancel each other out (see Pha\
2792 se), resulting in a notch filter effect. Each "stage" – all-pass filter section – of\
2793 a phase shifter creates one of these notches. More stages create more notches, and \
2794 a deeper effect."),
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2795 quiz::Quiz("Phase-Locked Loop","PLL - Any of a number of processes used to help syn\
2796 chronize signals or devices by correcting phase differences. For example, in analog \
2797 tape machines, phase locking helps to keep multiple machines synced together by sens\
2798 ing phase differences in the playback of pilot tones by the two machines and adjusti\
2799 ng the speed to eliminate the phase difference. In synthesizers, phase locking contr\
2800 ols one tone generator so that it begins its waveform in phase with the signal from \
2801 another tone generator. Phase-locked loops (PLL) are reference signals used in the c\
2802 lock functions of electronic devices."),
2803 quiz::Quiz("Phase","A measurement (expressed in degrees) of the time difference betw\
2804 een two similar waveforms. One cycle of a waveform is considered to have 360 degrees\
2805 , just like a circle. How far you move around the circle (or through the waveform) c\
2806 an be defined by the phase. For example, if you are one-quarter of the way through a\
2807 waveform's cycle, your phase is 90°."),
2808 quiz::Quiz("Phasing","An effects sound created by varying the phase shift of an audi\
2809 o signal, then mixing it with the direct signal."),
2810 quiz::Quiz("Phon","A unit of apparent loudness, numerically equal to the same number\
2811 of dB as a tone playing at 1000 Hz. For example, a sound is said to be 60 phon if i\
2812 t is perceived to be as loud as a 1000-Hz tone playing at 60dB."),
2813 quiz::Quiz("Phone Plug","A plug (or its mating jack) with a diameter of 1/4 inch and\
2814 a length of 1 1/4 inches used for interconnecting audio."),
2815 quiz::Quiz("Phono Plug","A common audio connector found on most stereo systems with \
2816 a center pin as one connection and an outer shell as the second connection."),
2817 quiz::Quiz("Physical Modeling","One approach to (often digital) synthesis is to recr\
2818 eate the components of actual instruments – such as a vibrating string or tube, or a\
2819 resonating body such as the shell of a guitar or drum – and string those together t\
2820 o create sounds. There are a handful of modules available which perform this modelin\
2821 g to create their sounds."),
2822 quiz::Quiz("Pickup Pattern","The shape of the area in front of or around the microph\
2823 one from where it evenly picks up sound. Many use this term interchangeably with “po\
2824 lar pattern,” but a polar pattern gives more detail about microphone sensitivity. (S\
2825 ee also “Polar Pattern.”),
2826 quiz::Quiz("Pickup","1) A device on an electric guitar or other instrument that puts\
2827 out an audio signal according to the string motion on the instrument. 2) See “Conta\
2828 ct Microphone.”),
2829 quiz::Quiz("Pinch Roller","A rubber (or plastic) wheel on a tape recorder that pinch\
2830 es the tape between it and the capstan, allowing the capstan to pull the tape."),
2831 quiz::Quiz("Ping-Ponging (Bouncing)","The technique of combining and mixing multiple\
2832 tracks onto one or two tracks (mono or stereo). This can be done in real-time or an\
2833 analog by playing the tracks through the console and recording them onto separate trac\
2834 ks, or digitally through a digital audio workstation. Bouncing was once used frequen\
2835 tly by engineers to free up additional tracks for recording, but in digital workstat\
2836 ions where tracks are virtually unlimited, this practice is basically obsolete. Toda\
2837 y, engineers typically bounce tracks for the purpose of creating a preliminary or fi\
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2838     nal mix of a song."),
2839     quiz::Quiz("Pink Noise","A noise signal similar to white noise, containing all audib\
2840     le frequencies, but with equal energy per octave as opposed to all frequency bands. \
2841     Engineers frequently use pink noise as a tool to tune and calibrate audio equipment.\
2842     (See also "White Noise.") Noise is a random, unpitched signal that, at audio rates,\
2843     can sound like hissing or the wind. Pink noise has equal energy (sound level) per o\
2844     ctave. As each higher octave has double the frequency of the octave below it which s\
2845     preads out the energy over a wider range of frequencies, pink noise tends have a mor\
2846     e natural, less electronic sound with more bass and less high end – especially when \
2847     compared to white noise, which has an equal energy per number of hertz (frequency) a\
2848     nd therefore tends to sound very bright."),
2849     quiz::Quiz("Pitch Bend","A mechanism on a synth, keyboard or controller that can cau\
2850     se the pitch of the note to move up or down by a small amount."),
2851     quiz::Quiz("Pitch to Voltage Converter","A device that detects the frequency of an a\
2852     udio waveform and changes it into a control voltage, which is in turn fed to an osci\
2853     llator that produces a pitch at the same frequency."),
2854     quiz::Quiz("Pitch-to-MIDI Converter","A device that detects pitch in an analog audio\
2855     signal and translates it into MIDI information. (Also called "Audio-to-MIDI-Convert\
2856     er.")"),
2857     quiz::Quiz("Pitch-to-Voltage Converter","A device that detects the frequency of an a\
2858     udio waveform and changes it into a control voltage, which is in turn fed to an osci\
2859     llator that produces a pitch at the same frequency."),
2860     quiz::Quiz("pitch","1) The perception of frequency by the ear (a higher or lower ton\
2861     e of music). 2) A control on a tape transport which adjusts the speed slightly up or\
2862     down, changing the pitch and time of the music."),
2863     quiz::Quiz("Plate Reverb","A device that produces artificial reverberation by sendin\
2864     g vibrations across a metal plate via a transducer similar to a speaker driver. Phys\
2865     ical plate reverbs today are considered a vintage form of artificial reverb; nowaday\
2866     s, most plate reverb effects are emulated digitally by plugins or reverb units."),
2867     quiz::Quiz("Playback Head","A transducer that converts magnetic flux recorded on tap\
2868     e into an audio signal for playback."),
2869     quiz::Quiz("Playback Mode","A configuration on a console that allows quick playback \
2870     of the signal previously recorded on tape or via DAW via the monitor mixer."),
2871     quiz::Quiz("Playback","1) The reproduction of recorded audio. 2) In motion picture o\
2872     r video production, the reproduction of the music over loudspeakers so the performer\
2873     s/musicians can perform in time to the music for the camera."),
2874     quiz::Quiz("Playlist","1) See "Take." 2) A user-defined selection of songs; a featur\
2875     e available on most streaming and digital media players."),
2876     quiz::Quiz("PLL","A phase locked loop is, in essence, an oscillator that tries to ma\
2877     tch the frequency of – or more importantly, a division or multiple of the frequency \
2878     of – another signal. This is most commonly used to create a frequency that is much h\
2879     igher than the incoming reference signal – such as a timing module that can create a\
2880     n output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a v\
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2881 ery high frequency oscillator that is locked to a multiple of an incoming pitch - pe\  
2882 rhaps to drive a special circuit such as a switched-capacitor filter."),  
2883 quiz::Quiz("Plug","A connector, usually on a cable, that mates with a jack."),  
2884 quiz::Quiz("Polar Pattern","1) In microphones, a graphic display of the area around \  
2885 the microphone that is sensitive to sound waves, detailing the audio output levels i\  
2886 n dB of sound arriving from different directions. Similar to "Pickup pattern," but m\  
2887 ore specific. 2) In speakers, a graphic display of the speaker's dispersion of sound\  
2888 ."),  
2889 quiz::Quiz("Polarity","The direction of current flow or magnetizing force."),  
2890 quiz::Quiz("Polarizer","An inverter multiplies an incoming control voltage by -1. In\  
2891 the case of a gate or logic inverter, it reverses the high and low states so that (\  
2892 for example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pol\  
2893 arizer, as it changes the polarity (+ versus -) of a signal. A control voltage inver\  
2894 ter is often combined with an offset voltage to adjust the output voltage into the d\  
2895 esired range. For example, if you had an envelope generator that had an output range\  
2896 of 0 to +8 volts, and you just inverted it, the result would be 0 to -8 volts. Sinc\  
2897 e some modules such as voltage controlled amplifiers usually expect only positive vo\  
2898 ltages, you would then need to add 8 volts to that result to get an upside-down (inv\  
2899 erted) envelope that still had an overall range of 0 to +8v."),  
2900 quiz::Quiz("Polarizing Voltage","In condenser and electret microphones, the introduc\  
2901 tion of a small amount of electrical current to create the magnetism by which the ca\  
2902 pacitor converts audio signals to electrical current. In condenser microphones, pola\  
2903 rizing voltage is provided externally (see also "Phantom Power"); in electret microp\  
2904 hones, the polarizing voltage is permanently impressed on the condenser during manuf\  
2905 acturing."),  
2906 quiz::Quiz("Pole Pieces","Iron or other magnetic material that conducts magnetic for\  
2907 ce for use in transducers like record heads, playback heads, microphones, speakers, \  
2908 etc."),  
2909 quiz::Quiz("Pole","This is a technical term that helps describe the design of a filt\  
2910 er. Each pole of a filter attenuates frequencies beyond its cutoff or corner frequen\  
2911 cy by 6 decibels (dB)/octave; the more poles, the stronger the filtering effect. A 4\  
2912 -pole low pass filter, for example, attenuates frequencies one octave above its cuto\  
2913 ff frequency by 24 dB; frequencies two octaves above the cutoff are attenuated by 48\  
2914 dB and so forth."),  
2915 quiz::Quiz("Polyphonic","The term \"polyphonic\" refers to a synthesizer that can pl\  
2916 ay more than one individually articulated note at a time; in most cases, those notes\  
2917 all play a similar sound or patch. Able to play more than one pitch or "voice" at t\  
2918 he same time. A term commonly used to describe synths and keyboards. (See also "Voic\  
2919 e.")"),  
2920 quiz::Quiz("Ponging (Bouncing)","The technique of combining and mixing multiple trac\  
2921 ks onto one or two tracks (mono or stereo). This can be done in real-time or analog \  
2922 by playing the tracks through the console and recording them onto separate tracks, o\  
2923 r digitally through a digital audio workstation. Bouncing was once used frequently b\  

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2924 y engineers to free up additional tracks for recording, but in digital workstations \
2925 where tracks are virtually unlimited, this practice is basically obsolete. Today, en\
2926 gineers typically bounce tracks for the purpose of creating a preliminary or final m\
2927 ix of a song."),
2928 quiz::Quiz("Pop Filter","A device that is placed over a microphone or between the mi\
2929 crophone and vocalist to prevent loud "pop" sounds created by the vocalist's breath \
2930 directed toward the microphone."),
2931 quiz::Quiz("Port","1) A connection point in computer or electronic device for transm\
2932 itting and receiving digital data, similarly to how a jack receives and transmits au\
2933 dio signals. 2) An opening or vent in a speaker case that resonates with air movemen\
2934 t in the speaker, used in bass reflex speakers and woofers to enhance low frequencie\
2935 s."),
2936 quiz::Quiz("Portamento","A pitch change that smoothly glides from one pitch to anoth\
2937 er. Also refers to the synthesizer mode or MIDI command that allows or causes this t\
2938 o happen."),
2939 quiz::Quiz("Post Production","Refers to the work of adding tracks, editing and other\
2940 fine tuning after primary recording or filming has taken place. Post-production in \
2941 recording includes such things as additional overdubs, editing, mixing and mastering\
2942 . Post-production in film includes a wide range of additional audio and visual effec\
2943 ts. NOTE: We mention film in this context because film post-production includes a l\
2944 ot of audio work (e.g., voiceovers, foley, audio mixing and editing) to the point th\
2945 at many audio engineers are involved in film post-production as a full-time career."\
2946 ),
2947 quiz::Quiz("Post Roll","A segment of blank tape (or track silence, on a DAW) that ru\
2948 ns past the end of the recording. (See also "Pre-Roll.")"),
2949 quiz::Quiz("Post-Fader","Refers to an aux send position or setting that places the s\
2950 end after the channel fader within the signal path. Sending a signal post-fader mean\
2951 s the fader itself affects the level of the send signal, as opposed to pre-fader. (S\
2952 ee also Pre-Fader.)"),
2953 quiz::Quiz("Post","Refers to an aux send position or setting that places the send af\
2954 ter the channel fader within the signal path. Sending a signal post-fader means the \
2955 fader itself affects the level of the send signal, as opposed to pre-fader. (See als\
2956 o Pre-Fader.)"),
2957 quiz::Quiz("Pot","Often thought of as a fancy word for "knob," a potentiometer is ba\
2958 sically any mechanism that controls input or output voltage by varying amounts (for \
2959 example, panning a signal left/right, volume control, or the amount of signal sent t\
2960 o an aux send or bus. Potentiometers can be knobs or faders, meaning that almost eve\
2961 ry control on a console that isn't a button or switch is a potentiometer. However, m\
2962 any engineers commonly refer to faders as "faders" and knobs as "pots.""),
2963 quiz::Quiz("Potentiometer","(Abbreviated "Pot") Often thought of as a fancy word for\
2964 "knob," a potentiometer is basically any mechanism that controls input or output vo\
2965 ltage by varying amounts (for example, panning a signal left/right, volume control, \
2966 or the amount of signal sent to an aux send or bus. Potentiometers can be knobs or f
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2967 aders, meaning that almost every control on a console that isn't a button or switch \
2968 is a potentiometer. However, many engineers commonly refer to faders as "faders" and\
2969 knobs as "pots.""),
2970 quiz::Quiz("Power Amplifier","(abbreviated "Power Amp") A device that amplifies a li\
2971 ne level signal to drive a speaker or set of speakers. (See also "Line Level.""),
2972 quiz::Quiz("Power Distribution Board","This simple circuit board takes the output of\
2973 your modular system's power supply and creates multiple copies of it, routed to con\
2974 nectors that go to your individual modules."),
2975 quiz::Quiz("PPQN","When you send a clock signal (usually a gate signal or other elec\
2976 trical pulse) around a modular synth to move sequencers through their steps and the \
2977 such, it's good to know how fast that clock is pulsing. This is usually defined in t\
2978 erms of how many pulses there are per quarter note - PPQ or PPQN for short. If the c\
2979 lock is just happening every quarter note, then the clock speed is 1 PPQN; in the ca\
2980 se of DIN Sync (a popular standard among early Roland synths, with DIN being the typ\
2981 e of electrical connector used) or MIDI clocks, the standard is 24 PPQN. This means \
2982 the master pulse can define a triplet for every 8th note (8 x 3)."),
2983 quiz::Quiz("Pre / Post Switch","A switch on the input module that determines whether\
2984 the send control comes before or after the main channel fader in the signal path (S\
2985 ee also "Pre-Fader," "Post-Fader.""),
2986 quiz::Quiz("Pre Emphasis","A boosting of high frequencies during the recording proce\
2987 ss to keep the audible signal above the noise floor."),
2988 quiz::Quiz("Pre Fader","Refers to an aux send position or setting that places the se\
2989 nd before the channel fader within the signal path. Sending a signal pre-fader means\
2990 the fader does not affect the level of the send signal, as opposed to pre-fader."),
2991 quiz::Quiz("Pre-Delay","A parameter on a reverb unit or plugin that determines the a\
2992 mount of time (delay) between the original dry sound and the early reflections of re\
2993 verberation. This feature is often used to simulate the natural acoustic properties \
2994 of a room, but can also be used to create interesting unnatural effects."),
2995 quiz::Quiz("Pre-Echo","(Also called "Forward Echo") A compression artifact that ofte\
2996 n occurs in digital audio in which an "echo" of a sound (or part of a sound) is hear\
2997 d ahead of the sound itself, often due to the data inconsistencies in certain compre\
2998 ssed digital formats. A type of pre-echo can also sometimes occur in the end product\
2999 of a recording, occurring on tape as a result of low-level leakage caused by print-\
3000 through, and also on vinyl records due to physical differences and/or deformities in\
3001 the grooves between silence and a loud transient. In digital formats, pre-echo is g\
3002 enerally an unwanted problem that requires additional signal processing to resolve-b\
3003 ut in some cases it can also be used on purpose as a sound effect (not to be confuse\
3004 d with "Reverse Echo")."),
3005 quiz::Quiz("Pre-Fade Listen (PFL)","A function on the channel strip of a mixer or DA\
3006 W that allows a channel signal to be heard and often metered before the channel fade\
3007 r."),
3008 quiz::Quiz("Preamplifier (Preamp)","A low-noise amplifier designed to take a low-lev\
3009 el signal (for example, from a microphone) and bring it up to normal line level befo\
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3010 re sending it into the mixing console."),
3011 quiz::Quiz("Precedence Effect (Haas Effect)","Simply stated, a factor in human hearing in which we perceive the source of a sound by its timing rather than its sound level. In his research, Helmut Haas determined that the first sound waves to reach our ears help our brains determine where the sound is coming from, rather than its reflection or reproduction from another source. The reflection of the sound must be at least 10dB louder than the original source, or delayed by more than 30ms (where we can perceive it as an echo), before it affects our perception of the direction of the sound. This is what helps us distinguish the original sound source without being confused by reflections and reverberations off of nearby surfaces. Understanding the Haas effect is particularly useful in live audio settings, especially in large venues where loudspeakers are time-delayed to match the initial sound waves coming from the source."),
3023 quiz::Quiz("Precision Adder","Synthesizers are very sensitive to unintentional variations in pitch control voltage – any error can result in the oscillators under control going out of tune. Therefore, whenever you add together pitch control voltages inside a modular synth, you really should be using a precision adder that precisely adds together the pitch voltages without introducing an error. Ordinary mixers might slightly attenuate or amplify a voltage passed through them, which in most cases would create tuning errors."),
3030 quiz::Quiz("Premix","1) The process of mixing a set of tracks as group, then managing the mixed group in the context of the other tracks by routing them to an auxiliary channel. Consolidating tracks by bouncing is a form of premixing, but a premix is not necessarily pre-recorded. (See also “Bouncing.”) 2) An important part of film post-production in which the process of mixing a section of audio for combination with the others. Dialogue, Foley, SFX and music may all be premixed before being combined together under the video."),
3037 quiz::Quiz("Presence Frequencies","The range of audio frequencies between 4 kHz and 6 kHz that when boosted, can increase the sense of presence, especially on voices.")\
3039 ,
3040 quiz::Quiz("Presence","1) In amplification and mixing, the boosting of upper-mid frequencies to cause a sound or instrument to cut through, creating the impression that the sound source is more “present,” right next to the listener. 2) See “Room Tone.”),
3044 quiz::Quiz("Preset","A factory programmed set of parameters on a synth, signal processor, plug-in or other electronic device."),
3046 quiz::Quiz("Pressure Microphone","(Also called “pressure operative microphone”) – A microphone whose diaphragm responds to incoming sound wave pressure as it works against the normal or controlled air pressure inside the microphone case. This design makes the diaphragm sensitive to pressure regardless of direction, giving it an omnidirectional pickup pattern. (See also “Omnidirectional Pattern.”)"),
3051 quiz::Quiz("Pressure Sensitivity (Aftertouch)","A feature in some keyboard instruments by which applying additional pressure to a key after it has been pressed can acti\
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3053 vate an additional MIDI control command. a synthesizer or Keyboard Controller of Aft\  
3054 er Touch (a control or operational function of a synthesizer where pressing a key af\  
3055 ter it has been pressed, and before it is released, will activate a control command \  
3056 that can be set by the player)."),  
3057 quiz::Quiz("Pressure Zone Microphone (Boundary Microphone)","An omnidirectional micr\  
3058 ophone designed to be placed flush against a flat surface (or boundary), effectively\  
3059 creating a "half-Omni" pickup pattern while eliminating the danger of phase issues \  
3060 from reflected sounds. A popular type of boundary microphone is Crown Audio's tradem\  
3061 ark Pressure Zone Microphone (PZM)."),  
3062 quiz::Quiz("Pressure-Gradient Microphone","(Also called "Velocity Microphone") A mic\  
3063 rophone whose diaphragm is exposed front and back, with diaphragm movement being cau\  
3064 sed by the pressure difference between its front and back. This creates a bi-directi\  
3065 onal or "figure-8" pickup pattern (See also "Bi-Directional Pattern.")"),  
3066 quiz::Quiz("Pressure","Some keyboards measure how hard you press down on the keys, a\  
3067 nd convert this to a voltage (or other control signal such as MIDI, which can then b\  
3068 e converted into a control voltage) that you can use to add expression to a note, su\  
3069 ch as adding vibrato or opening the filter wider. Monophonic aftertouch measures one\  
3070 pressure value for the entire keyboard, regardless of which key(s) you are pressing\  
3071 ; polyphonic aftertouch produces a signal for each individual key. Important trivia:\  
3072 Touch plate keyboards actually measure the surface area of the skin touching them r\  
3073 ather than pressure or force - so you can increase or decrease the aftertouch amount\  
3074 by rolling between the tip and length of your finger."),  
3075 quiz::Quiz("Print Through","The unwanted transfer of magnetic flux from one layer of\  
3076 analog tape to another."),  
3077 quiz::Quiz("Pro Tools","Avid's trade name for its digital audio workstation (DAW) th\  
3078 at has become an industry standard in professional recording studios."),  
3079 quiz::Quiz("Producer","In music, the producer is the director of an audio recording \  
3080 project; the person responsible for getting a final product of desired quality withi\  
3081 n a budget."),  
3082 quiz::Quiz("Production Studio","Broadly speaking, any space dedicated to production \  
3083 within the arts, for example, film/video, animation or post production. In the conte\  
3084 xt of audio, a production studio is effectively a recording studio that specializes \  
3085 in the assembly and mixing of commercials and radio programs from pre recorded music\  
3086 and effects with newly recorded dialogue."),  
3087 quiz::Quiz("Production","1) The collective actions that go into producing music. 2) \  
3088 Describing the quality of a recording—the end result of production decisions during \  
3089 the recording and mixing process."),  
3090 quiz::Quiz("Program Change","A MIDI message that tells the receiving device to chang\  
3091 e presets."),  
3092 quiz::Quiz("Programmable","Able to have the parameters changed by the user, especial\  
3093 ly in a computer controlled device."),  
3094 quiz::Quiz("Prompt","A set of instructions for the user to follow, which appears on \  
3095 a computer screen."),
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3096 quiz::Quiz("Protocol","In digital and information technology, a set of rules governi\  
3097 ng the structuring and transmitting of data in a standardized format so all related \  
3098 devices can properly interpret the data."),  
3099 quiz::Quiz("Proximity Effect","The natural boost in the microphone's output for bass\  
3100 frequencies as the mic is placed closer to the sound source."),  
3101 quiz::Quiz("Psychoacoustics","The study of how humans perceive and respond to sound,\  
3102 not just in the context of interpreting the physical sound waves, but also taking p\  
3103 sychological and emotional factors into account. This branch of science is helpful t\  
3104 o audio engineers in understanding how the brain interprets various sounds and frequ\  
3105 encies."),  
3106 quiz::Quiz("Puck","Any circular piece of metal, fiber, rubber, etc., which drives so\  
3107 mething from a rotating power source. A common example in the recording studio is th\  
3108 e puck in a rotating Leslie speaker."),  
3109 quiz::Quiz("Pulse Code Modulation (PCM)","A process by which analog signals are tran\  
3110 slated to digital code. This is done by taking samples of the amplitude of the analo\  
3111 g signal at regular rapid intervals, then translating it into binary numbers as a di\  
3112 gital representation of the original signal. The faster the sample rate, the better \  
3113 the digital reproduction. PCM is the most common form of A/D conversion in digital a\  
3114 udio."),  
3115 quiz::Quiz("Pulse Per Quarter Note","When you send a clock signal (usually a gate si\  
3116 gnal or other electrical pulse) around a modular synth to move sequencers through th\  
3117 eir steps and the such, it's good to know how fast that clock is pulsing. This is us\  
3118 ually defined in terms of how many pulses there are per quarter note - PPQ or PPQN f\  
3119 or short. If the clock is just happening every quarter note, then the clock speed is\  
3120 1 PPQN; in the case of DIN Sync (a popular standard among early Roland synths, with\  
3121 DIN being the type of electrical connector used) or MIDI clocks, the standard is 24\  
3122 PPQN. This means the master pulse can define a triplet for every 8th note (8 x 3).\  
3123 ),  
3124 quiz::Quiz("Pulse Width Modulation","Most oscillators that output a square waveform \  
3125 also have an additional control voltage input that sets the width of the top portio\  
3126 n of the "square" wave (obviously, making the top portion wider makes the bottom port\  
3127 ion narrower and vice versa). The act of varying the width of the resulting pulse wa\  
3128 ve creates a sort of Doppler shift; varying the width back and forth - for example, \  
3129 by modulating the pulse width with a low frequency oscillator - creates a chorusing \  
3130 effect that can sound like a detuned pair of oscillators. The resulting effect is re\  
3131 ferred to as pulse width modulation. The process of using a control voltage to vary \  
3132 the width of a pulse wave form, essentially switching between square waves and pulse\  
3133 waves. This has the effect of creating richer timbres, giving sounds a thicker, mor\  
3134 e lush feel, or of giving a digital sound more analog properties."),  
3135 quiz::Quiz("Pulse","Pulse has a couple of different meanings in a modular synth. Whe\  
3136 n you alter the shape of a square wave so that one portion is narrower than the othe\  
3137 r, it is referred to a pulse wave (see Pulse Wave Modulation below). Also, a narrow \  
3138 gate or trigger used as a clocking signal for sequencers and the such is often refer\
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3139 red to as a pulse. 1) The steady beat in music based on its tempo, whether audible o\  
3140 r perceived. 2) A type of sound wave commonly created and manipulated by synthesizer\  
3141 s whose waveform is characterized by sharp rises and drops in amplitude like a squar\  
3142 e wave, but whose peaks are shorter than its troughs, giving the wave a pulse-like f\  
3143 eel. Also called "Pulse Wave.""),  
3144 quiz::Quiz("Pumping and Breathing","In studio jargon, an effect created when a compr\  
3145 essor is rapidly compressing and releasing the sound, creating audible changes in th\  
3146 e signal level. "Pumping" generally refers to the audible increase of sound levels a\  
3147 fter compression has taken place; "breathing" refers to a similar effect with vocals\  
3148 , raising the signal volume just as the vocalist is inhaling. Pumping and breathing \  
3149 is a sign of cheap compression or over-compression, and is usually undesirable, alth\  
3150 ough some engineers and musicians use it on purpose occasionally to create a particu\  
3151 lar effect."),  
3152 quiz::Quiz("Punch In / Punch Out Recording","The process of activating and/or deacti\  
3153 vating the record function on tape or DAW during playback of a passage, usually as t\  
3154 he performer plays/sings along. This can be used either as a method of doing quick o\  
3155 verdubs, or as a way of getting a better take on a certain passage without having to\  
3156 start the track from the beginning."),  
3157 quiz::Quiz("Pure Tone","A tone consisting of only the fundamental frequency with no \  
3158 overtones or harmonics, graphically represented as a simple sine wave."),  
3159 quiz::Quiz("PVC","PVC stands for pitch to voltage conversion. In the quest to play a\  
3160 voltage-controlled synthesizer with something other than a keyboard-like thingy (to\  
3161 uch plates included), some have designed modules or other equipment that attempt to \  
3162 detect the pitch of an audio signal - say, from a guitar, flute, or singer - and con\  
3163 vert that pitch to a corresponding voltage that can drive a VCO in unison with the o\  
3164 riginal sound."),  
3165 quiz::Quiz("PWM","Most oscillators that output a square waveform also have an additi\  
3166 onal control voltage input that sets the width of the top portion of the "square" wa\  
3167 ve (obviously, making the top portion wider makes the bottom portion narrower and vi\  
3168 ce versa). The act of varying the width of the resulting pulse wave creates a sort o\  
3169 f Doppler shift; varying the width back and forth - for example, by modulating the p\  
3170 ulse width with a low frequency oscillator - creates a chorusing effect that can sou\  
3171 nd like a detuned pair of oscillators. The resulting effect is referred to as pulse \  
3172 width modulation."),  
3173 quiz::Quiz("PZM","Abbreviation for Crown Audio's Pressure Zone Microphone. (See also\  
3174 "Boundary Microphone.")"),  
3175 quiz::Quiz("Q - (Also called "Q Factor")","Stands for "Quality Factor," defining the\  
3176 bandwidth of frequencies that will be affected by an equalizer. The lower the Q, th\  
3177 e broader the bandwidth curve of frequencies that will be boosted or cut. If you com\  
3178 e from the pro audio world, you may be used to Q referring to the width or narrownes\  
3179 s of a peak or notch filter. In a synthesizer filter, when you increase the resonanc\  
3180 e (feedback), a peak forms around the cutoff frequency of the filter's curve or shap\  
3181 e. The higher the resonance, the higher and narrower this peak. As a result, some us\
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3182 ed to use the audio term Q to refer to the resonance amount, although you don't hear\  
3183 that term used nearly as much today."),  
3184 quiz::Quiz("Quadraphonic","A now rarely-used system of four-channel sound where the \  
3185 channels are designated as left front, left back, right front, right back, intended \  
3186 to deliver sound from all four corners of a room. Quadraphonic sound was a precursor\  
3187 to the surround-sound systems of today."),  
3188 quiz::Quiz("Quadrature","You can define a full cycle of a waveform as consisting of \  
3189 360 degrees, akin to a circle. One quarter of the way around this circle – or moving\  
3190 to a point that is one quarter of the way through a cyclical wave – is 90°. A sine \  
3191 and cosine wave are shifted 90° degrees or a quarter cycle out of alignment (phase) \  
3192 with each other. Since this is a quarter of a cycle, this is often referred to as a \  
3193 quadrature relationship."),  
3194 quiz::Quiz("Quantization Distortion","Quantization Distortion/Quantization Error – T\  
3195 he effective “error in translation” between an analog signal and its sampled counter\  
3196 part due to the rounding of a large number of analog values to the nearest digital q\  
3197 uantity. This often results in additional random frequencies in the sound, often hea\  
3198 rd as noise."),  
3199 quiz::Quiz("Quantization Noise","The modulation noise in a signal resulting from qua\  
3200 ntization error. "),  
3201 quiz::Quiz("Quantization","1) In digital music, the process of adjusting the rhythmi\  
3202 c performance of music by moving the notes to precise locations on the time line, ef\  
3203 fectively “rounding” the note occurrences to the nearest defined increment. 2) In an\  
3204 analog-to-digital conversion, the use of the same mathematical quantization principles\  
3205 to convert an analog signal into a smaller set of steps (a digital quantity)."),  
3206 quiz::Quiz("Quantizer","A quantizer auto-corrects the input voltage to the nearest d\  
3207 esired target, such as the voltage that corresponds to a semitone or other note in a\  
3208 scale. These are occasionally built into modules like sequencers or oscillators, bu\  
3209 t quite often they are standalone modules."),  
3210 quiz::Quiz("Rack Ears","Rack Ears/Rack Flanges – Mounting brackets that can be atta\  
3211 ched to equipment so it can be mounted in a standard equipment rack."),  
3212 quiz::Quiz("Rack Mounted","Describing outboard gear that can be housed in an equipme\  
3213 nt rack."),  
3214 quiz::Quiz("Rack Rash","When you mount a module into a case, the head of the screw o\  
3215 r bolt used to mount the module can scratch the faceplate of the module. These scrat\  
3216 ches are referred to as rack rash. You can almost never see it when you mount a modu\  
3217 le, as the scratches are behind the screw or bolt head, but nonetheless some will pa\  
3218 y more for a used module that is unscratched. So buy a bag of plastic washers and pu\  
3219 t them behind the screw or bolt head just to remove another reason for someone to no\  
3220 t buy your used module."),  
3221 quiz::Quiz("Rack Unit","Rack-mounted equipment usually follows a standard set of dim\  
3222 ensions, including 19” (48.3 cm)for width, and a “rack unit” (or U for short) for he\  
3223 ight equaling 1.75” (4.4 cm) per U. Many common modular synthesizer formats follow t\  
3224 he rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25” o\  

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3225 r 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (\
3226 sometimes referred to as MU for Moog Unit)."),
3227 quiz::Quiz("Radiation Pattern","A graphic depiction of speaker coverage. This is not\
3228 unlike the polar pattern of a microphone, with the exception that a polar pattern d\
3229 escribes the area where sound arrives at the microphone, while a radiation pattern d\
3230 escribes how sound is dispersed from the loudspeaker."),
3231 quiz::Quiz("Radiation","The angle and pattern of coverage of a speaker."),
3232 quiz::Quiz("Ramp","In general, a ramp refers to any voltage that is steadily raising\
3233 or falling; quite often it resets when it reaches a target voltage and starts over \
3234 again. A sawtooth oscillator waveform is sometime referred to as a ramp. Sometimes, \
3235 the individual stages of an envelope generator are also referred to a ramp as it rai\
3236 ses from 0 volts to a maximum level such as 5v for the attack stage, then falls from\
3237 this peak to the sustain level for the decay stage."),
3238 quiz::Quiz("Random Access Memory (RAM)","The “short-term” memory in a computer that \
3239 is used in tandem with the processor for performing immediate tasks (as opposed to h\
3240 ard-drive storage memory where projects are saved and recalled). In the recording st\
3241 udios, the more RAM a computer has, the more ability it has to handle large amounts \
3242 of data at a time (for example, in multi-track recording or working with virtual MID\
3243 I instruments)."),
3244 quiz::Quiz("Random Note Generator","A device that generates random pitches at a set \
3245 rate, used in synthesizers."),
3246 quiz::Quiz("Random","Most voltages moving around inside a modular synth are very pur\
3247 poseful in their variations: the repeating waveforms of an audio rate or low frequen\
3248 cy oscillator; the rising then falling voltages of an envelope generator. However, i\
3249 t can also be useful to have randomly wandering voltages to create everything from s\
3250 ubtle variations in pitch to wildly varying volumes or filterings. Noise is an exampl\
3251 le of an audio-rate random signal."),
3252 quiz::Quiz("Rap","To perform a spoken rhythmic part to a music or percussion perform\
3253 ance."),
3254 quiz::Quiz("Rarefaction","The reduced density of air particles during the trough of \
3255 a sound wave; in the context of “compression and rarefaction,” it is the opposite of\
3256 compression. (See also “Compression.”)"),
3257 quiz::Quiz("Ratcheting","This is a trick used with sequencers where one stage of the\
3258 sequence may be triggered quickly multiple times, rather than just once as you step\
3259 to that stage. For example, the result may be a series of quarter notes, with a bur\
3260 st of four sixteenth notes appearing instead for one or more stages."),
3261 quiz::Quiz("Rate","This word is used sometimes to refer to the speed or frequency of\
3262 a low frequency oscillator or similar repetitive function, such a sequencer’s tempo\
3263 clock."),
3264 quiz::Quiz("Rated Load Impedance","The input impedance, or opposition to current flo\
3265 w by an input of a device, that a piece of equipment is designed to feed."),
3266 quiz::Quiz("RCA Plug","(Also called Phono Plug) A common audio connector found on mo\
3267 st stereo systems with a center pin as one connection and an outer shell as the seco\
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3268 nd connection."),
3269 quiz::Quiz("Read Only Memory (ROM)","A type of data storage that cannot be erased or\
3270 reprogrammed by the user. The most common form of ROM in audio/video settings today\
3271 is optical storage media (i.e, CD, DVD, CD-ROM and DVD-ROM)."),
3272 quiz::Quiz("Read","To retrieve information bits from a storage device; in digital au\
3273 dio, the reproduction of digital signals."),
3274 quiz::Quiz("Reason","Popular music software program from Propellerhead Software. It\
3275 offers the digital equivalent of hardware synthesizers, samplers, signal processors\
3276 , sequencers and mixers. Reason works as a virtual music studio, or as a set of virt\
3277 ual musical instruments which can be played live or used with other sequencing softw\
3278 are."),
3279 quiz::Quiz("Recapping","Electronic components can age. Certain types of capacitors -\
3280 namely, electrolytic and tantalum, often used in the power supply section - are the\
3281 most likely to deteriorate over time; some put the maximum safe life of an electrol\
3282 ytic capacitor to be 25 years. Therefore, serious vintage synth owners "recap" (repl\
3283 ace the age-sensitive capacitors in) their older equipment."),
3284 quiz::Quiz("Record Head","A device on an analog tape machine that changes electrical\
3285 current to magnetic energy; the changes of the magnetism match the waveshape of the\
3286 audio signal fed to the head."),
3287 quiz::Quiz("Record Level","A control on a tape machine that determines the amount of\
3288 magnetic flux recorded on the tape, or the DAW control that determines the level of\
3289 the digital signal recorded to the sound file."),
3290 quiz::Quiz("Record Monitor","On some tape machines, a switch position that allows th\
3291 e VU meter and sound output of the tape machine electronics to monitor the input sig\
3292 nal to the tape machine."),
3293 quiz::Quiz("Record Ready","A control state of a multitrack tape recorder where the d\
3294 esignated track will begin recording when the record function of the tape recorder i\
3295 s activated."),
3296 quiz::Quiz("Recording Bus","A bus that sends a mix signals from the console channels\
3297 to the multitrack recorder or DAW. (See also "Bus.")"),
3298 quiz::Quiz("Recording Session","A bloc of time in which music is being recorded in t\
3299 he studio."),
3300 quiz::Quiz("Rectifier","A circuit that makes sure a voltage stays only positive or n\
3301 egative. In power supplies, it is used to remove the negative component of AC voltag\
3302 e, or to protect you from plugging in module's power connector backwards. As a modul\
3303 e, a half-wave rectifier passes only positive voltages and replaces anything negativ\
3304 e with 0v; a full-wave rectifier takes any negative voltages and inverts them so the\
3305 y become positive. This effectively doubles the frequency of many simple waveforms, \
3306 like the triangle and sine."),
3307 quiz::Quiz("Red Noise","Also referred to as brown noise, technically it's a type of \
3308 noise whose power density (spectral loudness) decreases 6 dB per octave with increas\
3309 ing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distanc\
3310 e. It can also be used a slowly changing random control voltage or modulation signal\
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3311 , instead of as an audio source."),
3312 quiz::Quiz("Reel","1) The hub and flanges onto which analog tape is spooled; recordi\
3313 ng and playback involves unspooling the tape from one reel and onto another. 2) Some\
3314 times also called "demo reel," a compilation of audio or video that demonstrates the\
3315 abilities of a musician, audio engineer, actor, or other audio/visual professional.\
3316 Unlike a demo, which is intended to pitch one or more songs, a reel is a demo inten\
3317 ded to promote the abilities of the professional rather than the product itself. The\
3318 term itself is a holdover from the days when this promotional material was delivere\
3319 d on reels."),
3320 quiz::Quiz("Reference Level","1) A standard baseline level of volume used to measure\
3321 how much level is present in dB above or below the baseline. 2) See "Operating Leve\
3322 l.""),
3323 quiz::Quiz("Reference Tone","A single-frequency tone (often at 1000 kHz) used to cal\
3324 ibrate the levels of sound equipment; the tone used to set reference level. (See als\
3325 o "Test Tones.")"),
3326 quiz::Quiz("Reflected Sound","Sound that reaches a microphone or listener after one \
3327 or more reflections from surrounding surfaces."),
3328 quiz::Quiz("Reflection","In acoustics, the bouncing of sound waves off of a flat sur\
3329 face, as opposed to absorption. Reflection can have a great impact on how we perceiv\
3330 e the collective sound; reflected sounds from a distance is perceived as echo, while\
3331 reverberation is created from thousands of reflections. (See also "Absorption," "Ea\
3332 rly Reflection," "Echo," "Reverberation.")"),
3333 quiz::Quiz("Regeneration","Regeneration can have a couple of different meanings insi\
3334 de a synth, both meaning feedback. An echo unit can feed some of its output back int\
3335 o its input, causing the delayed signal to be repeated again; this is sometimes refe\
3336 rred to as regeneration. Also, very rarely you will hear resonance in a filter refer\
3337 red to as regeneration."),
3338 quiz::Quiz("Regulated Power Supply","A device to supply power to electronic equipmen\
3339 t whose output voltage will not fluctuate when more equipment is turned on, or if th\
3340 ere is a change in voltage of the power line. A regulated power supply is designed t\
3341 o protect sensitive electronics from destructive power surges."),
3342 quiz::Quiz("Relay","An electromagnetically activated switch that connects or disconn\
3343 ects two terminals when a control voltage is applied."),
3344 quiz::Quiz("Release Time","In dynamics signal processors, the time it takes for the \
3345 output signal to return to original levels when the input signal crosses the designa\
3346 ted threshold."),
3347 quiz::Quiz("Release","This refers to the final stage of an envelope that typically f\
3348 alls back to zero volts, usually resulting in silence. It is often used in the conte\
3349 xt of talking about an Attack/Release (AR) or Attack/Decay/Sustain/Release (ADSR) en\
3350 velope generator, but can refer to any final stage of an envelope."),
3351 quiz::Quiz("Remote","1) A device that controls the functions of another device wirel\
3352 essly. 2) Describing on-site recording, as opposed to recording in the studio."),
3353 quiz::Quiz("Reset","The Reset input on a module accepts a trigger or gate signal, an\
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3354 d tells the module to go back the beginning of whatever it was doing. In the case of\  
3355 a clock divider, this means pretend the next clock is the first clock you should be\  
3356 counting in the division (more on that in the full definition). In the case of a se\  
3357 quencer, it means go back to the first stage. In the case of an envelope, it means g\  
3358 o back to the start of the attack. In the case of a gate delay, it means to re-start\  
3359 the timer for the delay."),  
3360 quiz::Quiz("Residual Magnetization","The amount of magnetism left in a magnetic mate\  
3361 rial after the magnetizing force is removed. Residual magnetism can accumulate in ta\  
3362 pe machines over time, either creating distortions and noise in the sound output or \  
3363 partially erasing the tape."),  
3364 quiz::Quiz("Residual Noise","The noise level left on recording tape after it has bee\  
3365 n erased."),  
3366 quiz::Quiz("Resistance","The opposition of a substance to the flow of electrical cur\  
3367 rent, measured in ohms."),  
3368 quiz::Quiz("Resistor","An electrical component with a specific amount of resistance \  
3369 to electrical current, used within the circuit to regulate the flow of current."),  
3370 quiz::Quiz("Resonance","The natural tendency of physical substances to vibrate with \  
3371 more energy at certain frequencies. The principle of resonance is a key element in t\  
3372 he design of acoustic instruments; for example, the hollow chamber of a guitar or vi\  
3373 olin is designed to resonate with the vibrations of the string. Resonance also plays\  
3374 a role the acoustic design of a space, and even in developing good vocal technique \  
3375 to project the voice. When the output of a filter is fed back into its input, the re\  
3376 sult is an increased boost in the harmonics right around the filter's cutoff or corn\  
3377 er frequency. The audible result is similar to playing a sound in a room that has a \  
3378 resonance - sympathetic, reinforcing echo or vibration - at a certain frequency. The\  
3379 refore, the term resonance is often used to refer to a filter's feedback amount."),  
3380 quiz::Quiz("Resonant Frequency","A frequency at which a physical item vibrates natur\  
3381 ally."),  
3382 quiz::Quiz("Resonate","To vibrate at the resonant frequency. Also refers to the ling\  
3383 ering reverberation that causes a sound to continue after the sound source has stopp\  
3384 ed. This continuing sound is due to the sympathetic resonance of nearby objects."),  
3385 quiz::Quiz("Resonator","Many acoustic instruments include a body or sound chamber th\  
3386 at "resonates" - sympathetically vibrates at, or reinforces - one or more frequencie\  
3387 s. To simulate this effect in modular synths, you can get a specialized filter or eq\  
3388 ualization module that boosts the sound at typically three or so user-definable freq\  
3389 uencies, each usually within a narrow band. This is one of the secrets of synthesizi\  
3390 ng real-world sounds or spaces."),  
3391 quiz::Quiz("Reverb (Reverberation)","1) Short for "Reverberation." (See "Reverberati\  
3392 on.") 2) A signal processor or plug-in that creates artificial reverb to a signal.")\  
3393 ,  
3394 quiz::Quiz("Reverb Time (RT)","The time it takes for the reverberation or echoes of \  
3395 a sound source to die out after the direct sound has stopped. Specifically, the reve\  
3396 rb time is measured between the point at which the sound source stops and the point \
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3397 at which the reverberation levels fall by 60 dB."),
3398 quiz::Quiz("Reverb","Short for reverberation. This is an effect device that mimics b\
3399 eing in a room where you can hear the original sound reflect off the walls multiple \
3400 times, bouncing around in a wash of sound until it eventually decays into silence. A\
3401 reverb can greatly enhance the sound of a synthesizer, adding lushness and dimensio\
3402 n to what might otherwise be a stark sound. There are relatively few modules that im\
3403 plement a reverb effect, and even fewer that allow you to voltage control some of it\
3404 s parameters (the ErbeVerb being the most famous); many just use an external reverb \
3405 effect."),
3406 quiz::Quiz("Reverberant Field","Describes the space that is far enough from the soun\
3407 d source that the reverberations are louder than the direct sound."),
3408 quiz::Quiz("Reverberation Chamber","A device built to simulate room reflections."),
3409 quiz::Quiz("Reverberation Envelope","The attack, decay, sustain and release of the r\
3410 everberation volume; or how fast the reverberation reaches peak level and its rate o\
3411 f decay."),
3412 quiz::Quiz("Reverberation","The persistence of a sound after the source stops emitti\
3413 ng it, caused by many discrete echoes arriving at the ear so closely spaced in time \
3414 that the ear cannot separate them."),
3415 quiz::Quiz("RF Interference","The unwanted noise introduced into electronics, circui\
3416 ts and/or audio systems by the presence of RF signals. RF interference in a system c\
3417 an result in humming, buzzing, static or even the reproduction of radio transmission\
3418 s."),
3419 quiz::Quiz("RF Signals","RF Signals (or RF) – Short for Radio Frequency Signals, ele\
3420 ctromagnetic waves that carry wireless radio and television signals. The vast majori\
3421 ty of RF signals exist at frequencies higher than 100 kHz."),
3422 quiz::Quiz("Rhythm Section","The musical instruments in a band or ensemble that are \
3423 responsible for playing rhythmic parts rather than melody parts. In contemporary mus\
3424 ic, rhythm sections typically consist of drums and bass, along with some combinatio\
3425 of percussion, piano/keyboard and/or guitars."),
3426 quiz::Quiz("Ribbon Controller","This is a long strip that is capable of measuring th\
3427 e position where you press it along its length, and the pressure used to press it. I\
3428 t can be used as an alternate keyboard or as a pitch bend controller, with the posit\
3429 ion determining pitch. Shorter versions also appeared sometimes as alternate control\
3430 lers on synthesizers, such as the Yamaha CS-80."),
3431 quiz::Quiz("Ribbon Microphone","A microphone that converts sound waves to electrical\
3432 current via a thin conductive ribbon set between magnetic poles. Ribbon microphones\
3433 are almost always responsive to sound on both sides of the ribbon, creating a bi-di\
3434 rectional or figure-8 pattern."),
3435 quiz::Quiz("Riff","A short melody repeatedly played in a tune often with variation b\
3436 etween vocal lines."),
3437 quiz::Quiz("Ring Modulator","Balanced or ring modulation is a special type of amplit\
3438 ude modulation, where one bipolar (swinging both above and below 0 volts) signal – t\
3439 he modulator – is used to vary the amplitude of a second bipolar signal, known as th\
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3440 e carrier. The modulator's frequency is both added to and subtracted from the carrier's
3441 r's frequency; the resulting harmonics replace the original carrier and modulator.")\
3442 ,
3443 quiz::Quiz("Ringing Out a Room","The process of identifying and compensating for pro\
3444 blem frequencies within a room for the purpose of optimizing live audio within that \
3445 space. This is typically done by sending pink noise through the speakers, turning up\
3446 the microphones to the point of feedback, and using EQ to notch out the offending f\
3447 requencies."),
3448 quiz::Quiz("Rise Time","The rate at which an audio waveform makes a sudden increase \
3449 to a higher amplitude."),
3450 quiz::Quiz("RMS Meter","A meter that recognizes and responds to the effective averag\
3451 e, the RMS level, or the effective average value of an AC waveform, rather than to t\
3452 he peak level. (See also "Root-Mean-Square," "Peak Meter.")"),
3453 quiz::Quiz("Roll Off","The reduction of signal level as the frequency of the signal \
3454 moves away from the cut-off frequency, especially when the cut-off rate is mild."),
3455 quiz::Quiz("Room Equalization","In live audio, an equalizer inserted in the monitor \
3456 system that attempts to compensate for frequency response changes caused by room aco\
3457 ustics."),
3458 quiz::Quiz("Room Sound","The natural ambience of a room, including the reverberation\
3459 and background noise."),
3460 quiz::Quiz("Room Tone","The natural background noise occurring in a room without mus\
3461 ic playing or people speaking. In recording audio for film and TV, on-set sound mixe\
3462 rs capture a take of room tone for the purpose of providing continuity between clips\
3463 of dialogue during post-production."),
3464 quiz::Quiz("Root Mean Square (RMS)","The effective average value of an AC waveform. \
3465 Used as a measure of the overall level of the sound rather than just measuring by th\
3466 e peaks. (See also "RMS Metering," "Peak Metering.")"),
3467 quiz::Quiz("Rotating Head","A circular head with two (or more) gaps that rotates aga\
3468 inst the direction of tape motion at a slight angle to the tape travel."),
3469 quiz::Quiz("Rumble","A low-frequency noise, typically caused by earth/floor vibratio\
3470 n or by uneven surfaces in the drive mechanism of a tape recorder or playback unit."\
3471 ),
3472 quiz::Quiz("Rythm Tracks","The recording of the rhythm instruments in a music produc\
3473 tion."),
3474 quiz::Quiz("S-trig","Some systems – such as the original Moog modular – use an s-tri\
3475 gger (switch or shorting trigger) instead of a normal gate, which was a wire that wa\
3476 s shorted to 0 volts ground, like the closing of a switch wired to ground. You canno\
3477 t interconnect these two systems without some form of conversion between the two, wh\
3478 ich can be as simple as a special cable."),
3479 quiz::Quiz("S/H","A sample and hold (S/H) module has two inputs: a signal that is be\
3480 ing sampled, and a trigger input that indicates when the first input should be sampl\
3481 ed. When a trigger is received, the current voltage at the first input is sampled (m\
3482 easured) and held (stored), and presented at the output. This stable voltage is held\
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3483   until a new trigger is received. Sample and holds are most often associated with cr\
3484   eating stepped random voltages. To do this, noise is fed to the main input; whenever\
3485   a trigger is received, the voltage present at that input is some random value, whic\
3486   h is then dutifully sent to the output."),
3487   quiz::Quiz("S/PDIF","Abbreviation for “Sony/Phillips Digital Interface,” a protocol \
3488   for sending and receiving digital audio signals using a common RCA connector."),
3489   quiz::Quiz("Safety Take (ST)","An additional take of audio captured for good measure\
3490   after a take of acceptable quality has been recorded."),
3491   quiz::Quiz("Sallen-Key","The Sallen-Key filter topology or design creates a \"second\
3492   order\" or two-pole low, high, or bandpass filter and is capable of high resonance \
3493   or Q. This is the design used in the Korg MS-20 filter and its clones, among others.\
3494   "),
3495   quiz::Quiz("Sample & Hold","A sample and hold (S/H) module has two inputs: a signal \
3496   that is being sampled, and a trigger input that indicates when the first input shoul\
3497   d be sampled. When a trigger is received, the current voltage at the first input is \
3498   sampled (measured) and held (stored), and presented at the output. This stable volta\
3499   ge is held until a new trigger is received. Sample and holds are most often associat\
3500   ed with creating stepped random voltages. To do this, noise is fed to the main input\
3501   ; whenever a trigger is received, the voltage present at that input is some random v\
3502   alue, which is then dutifully sent to the output."),
3503   quiz::Quiz("Sample Dump Standard (SDS)","See “MIDI Sample Dump Standard.”"),
3504   quiz::Quiz("Sample Rate Conversion","The conversion of digital audio taken at one sa\
3505   mple rate to a different sample rate without first converting the signal to analog."\
3506   ),
3507   quiz::Quiz("Sample Rate","This is a specification of digital audio: How fast the ind\
3508   ividual measurements (samples) that reconstruct a sound are recorded or played back.\
3509   The bandwidth of that audio file (which corresponds to the highest frequency that c\
3510   an be reproduced) is in practice a bit less than half of the sample rate. In digital\
3511   recording, the number of times per second that samples are taken. The higher the sa\
3512   mple rate, the more realistic the digital reproduction of the sound, and the higher \
3513   frequencies of the sound can be reproduced. In digital audio, the quality and resolu\
3514   tion of a digitally reproduced sound are described as a combination of sample rate a\
3515   nd bitrate. (See also “Bitrate.”"),
3516   quiz::Quiz("Sample","1) In digital recording, the numerical measure of the level of \
3517   a waveform at a given instant of time. Analog music is represented digitally by many\
3518   samples taken in rapid succession. 2) A short segment of audio recorded for the pur\
3519   pose of reproducing and manipulating the sound digitally."),
3520   quiz::Quiz("Sampler","A device that records and plays samples, often with features f\
3521   or editing, manipulating and storing the samples."),
3522   quiz::Quiz("Saturation","On a simple level, saturation is a fancy word for clipping:\
3523   Once the input voltage goes higher (or lower) than a circuit can handle, it is inst\
3524   ead held at that limit. However, saturation usually implies a more rounded, shaped a\
3525   pproach to that clipping limit, resulting in a more pleasing (or at least less annoy\
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3526 ing) form of distortion. Tubes circuits are often associated with this soft clipping\  
3527 behavior, although it can be emulated in other circuits or even digital signal proc\  
3528 essing. Different devices may be sought out for specific sonic character of the way \  
3529 they. 1) The point at which magnetic tape reaches full magnetization due to an exces\  
3530 s of sound level. This creates some distortion that some audiophiles describe as “an\  
3531 alog warmth” a desirable quality in certain instances. 2) The audio distortion that \  
3532 occurs by overdriving a signal through a tube amplifier or preamp—again producing co\  
3533 lor and warmth in the sound that engineers often find appealing. 3) A digital plugin\  
3534 that emulates tape or tube saturation.”),  
3535 quiz::Quiz("Sawtooth Wave","A waveform that jumps from a zero value to a peak value \  
3536 and then immediately drops to a zero value for each cycle. (Sometimes also called “R\  
3537 amp Wave.”)”),  
3538 quiz::Quiz("Sawtooth","One of the most common waveforms produced in a synthesizer. T\  
3539 his ramp-shaped wave contains both even and odd harmonics, strongest at the fundamen\  
3540 tal frequency (the note being played) and diminishing at the higher frequencies. The\  
3541 result is very bright, loud, “brassy” sound.”),  
3542 quiz::Quiz("Schmitt Trigger","This is a type of gate detector that looks at a varyin\  
3543 g input signal and outputs either a “high” (typically 0, 10, or even 15 volts) signa\  
3544 l or a “low” signal (typically 0 volts). When the input goes above one reference thr\  
3545 eshold – say, 4 volts – the output goes high. When the input then goes back below a \  
3546 second, different threshold – say, 1 volt – then the output goes back low.”),  
3547 quiz::Quiz("scope","This is a piece of test equipment that displays voltage fluctuat\  
3548 ions as graphical waveforms. A 'scope can run at a wide range of frequencies, displa\  
3549 ying slowly changing voltages like LFOs or envelopes, or quickly changing voltages l\  
3550 ike oscillators and noise. Oscilloscopes used to be bulky pieces of external equipme\  
3551 nt, but now you can get USB scopes that offload the display portion of the job to yo\  
3552 ur computer, or scopes as modules.”),  
3553 quiz::Quiz("Scratch","1) A descriptive term meaning “temporary”. 2) A scratch vocal \  
3554 is a vocal done during a basic recording session to help the musicians play their pa\  
3555 rts. At a later date the final vocal track is overdubbed. 3) The action of a musicia\  
3556 n or disc jockey quickly moving a record back and forth on a turntable reproducing t\  
3557 he stylus motion to create a rhythm pattern of sound.”),  
3558 quiz::Quiz("Scrubbing","The action or function of shuttling a piece of recorded audi\  
3559 o back and forth while monitoring it, typically to locate a certain point in the rec\  
3560 ording. In earlier days, scrubbing was done with reel-to-reel analog tape by manuell\  
3561 y turning the reels to pull the tape across the playhead. Today, scrubbing is primar\  
3562 ily done digitally on a DAW by dragging the cursor back and forth across the wavefor\  
3563 m.”),  
3564 quiz::Quiz("Second Engineer","An assistant recording engineer.”),  
3565 quiz::Quiz("SEM","The Oberheim SEM (Synthesizer Expander Module) was one of their ea\  
3566 rliest products. It was an entire synthesizer voice – two oscillators, two simple en\  
3567 velopes, VCA, and a very popular two-pole state variable filter design with a knob t\  
3568 hat crossfaded between low pass, notch, and high pass outputs plus a separate bandpa
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3569 ss setting – in a cube-like case. Most often today, when a modular manufacturer uses\  
3570 the magic letters "SEM", they're referring to a filter meant to emulate that in t\  
3571 he original Oberheim synth."),  
3572 quiz::Quiz("Semi-modular","The components of a semi-modular synth – such as the osci\  
3573 llator, filter and amplifier – are pre-wired behind the front panel in what the manu\  
3574 facturer considers to be a typical, logical way. However, they also provide patch po\  
3575 ints either to access some of its functions (such as the individual waveform outputs\  
3576 of the oscillator) to send to other modules, or to override that pre-wiring. Many w\  
3577 ho are new to modular synthesis dip their toe in the water by getting a semi-modular\  
3578 synth, and then expanding it with additional modules."),  
3579 quiz::Quiz("Semitone","Also known as a half step or half tone, this is the smallest \  
3580 pitch division in most Western music – such as the difference between a C and a C#. \  
3581 With equal temperament (the most common way of tuning a Western scale), this pitch d\  
3582 ivision is 1/12 of an octave."),  
3583 quiz::Quiz("Send Level","A control determining the signal level sent to a send bus."\  
3584 ),  
3585 quiz::Quiz("Sensitivity","1) In audio settings, describes the amount of output that \  
3586 a microphone can produce from a standard level of sound, as compared to the output o\  
3587 f another microphone from the same sound level. 2) In music, describes the artistic \  
3588 persona in general."),  
3589 quiz::Quiz("Sequence","1) A pre-programmed set of musical events, such as pitches, s\  
3590 ounding of samples, and rests, to be played in order by a device. Also refers to the\  
3591 action of programming the device to play this set of musical events. 2) Loosely ref\  
3592 erring to a segment of music in general."),  
3593 quiz::Quiz("Sequencer","The most common type of sequencer you're going to see in a m\  
3594 odular synth contains a row of knobs (also known as steps or stages) that may each b\  
3595 e set to output a different voltage. A sequencer then goes through steps one at a ti\  
3596 me. This is most often used to create repetitive musical lines where each note has t\  
3597 he same duration, which is popular in trance-like forms of music as well as the clas\  
3598 sic Berlin School style (70s-era Tangerine Dream and Klaus Schulze; current Red Shif\  
3599 t and Node). A computerized device or software that can be programmed to play a step\  
3600 ped order of musical events, including playing of pitches, sounding of samples, and \  
3601 rests."),  
3602 quiz::Quiz("Sequential Switch","This module comes in a few different forms; in the m\  
3603 ost common, a few different inputs are routed to one output (although they are usual\  
3604 ly symmetrical – one input can be switched between several outputs). A pulse or gate\  
3605 input then steps through the inputs one at a time, switching which ones is routed t\  
3606 o the output. Fancyer sequential switches allow you to set the number of stages, to \  
3607 divide an input clock so it switches at a slower tempo than the master clock, or mig\  
3608 ht directly route a series of inputs to corresponding outputs (with usually a summed\  
3609 output as well)."),  
3610 quiz::Quiz("Serial Data","A digital data stream where individual bits are transmitte\  
3611 d one after another over a single connection (as opposed to "parallel data," in whic\  

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3612 h multiple bits can be sent at once). Most data connections in the recording studio \
3613 transmit serial data—for example, USB, Firewire and MIDI."),
3614 quiz::Quiz("Series Connection","Connecting devices (especially circuit elements) so \
3615 that the electrical signal flows from one thing to the next, to the next, etc."),
3616 quiz::Quiz("Set Up","The positioning of microphones, instruments, connections and mo\
3617 nitoring in the studio, as well as the controls and levels on consoles, DAWs, etc., \
3618 in preparation for recording."),
3619 quiz::Quiz("Shelf Filter","A name for the circuit in an equalizer used to obtain the\
3620 shelf."),
3621 quiz::Quiz("Shelf","A frequency response of an equalization circuit where the boost \
3622 or cut of frequencies forms a shelf on a frequency response graph. A high-frequency \
3623 shelf control affects signal levels at the set frequency and all frequencies above i\
3624 t; a low-frequency shelf does the same for signals at and below the set frequency.")\
3625 ,
3626 quiz::Quiz("Shield","The outer conductive wrapping around an inner wire or wires in \
3627 a cable, for the purpose of shielding the cable from picking up external electromagn\
3628 etic interference."),
3629 quiz::Quiz("Shielded Cable","Cable that has a shield around an inner conductor or in\
3630 ner conductors."),
3631 quiz::Quiz("Shock Mount","An elastic mount on microphone stand that reduces the impa\
3632 ct of unwanted vibrations that may affect the stand (for example, floor vibrations f\
3633 rom footsteps)."),
3634 quiz::Quiz("Short Circuit","A direct connection between two points in a circuit that\
3635 (usually) should not be connected."),
3636 quiz::Quiz("Short Delay","Delay times under 20 milliseconds."),
3637 quiz::Quiz("Shortest Path","A technique in recording that routes the signal through \
3638 the least amount of active (amplified) devices during recording."),
3639 quiz::Quiz("Shotgun Microphone","A microphone with a long line filter, a tube that a\
3640 coustically cancels sound arriving from the side, to make the microphone pick up muc\
3641 h better in one direction than in any other direction. This gives the shotgun mic a \
3642 tight, hypercardioid pickup pattern. Shotgun microphones are commonly used to record\
3643 dialogue in filming situations, usually held on a boom stand with a shock mount."),
3644 quiz::Quiz("Sibilance","Energy from a voice centered around 7 kHz, caused by pronoun\
3645 cing “s”, “sh” or “ch” sounds."),
3646 quiz::Quiz("Sidechain","An auxiliary input to a signal processor that allows control\
3647 of the processing to be triggered by an external source. A common use of sidechaini\
3648 ng is in compressors, particularly in ducking effects where the presence of a partic\
3649 ular audio signal triggers the compression of another audio signal. (See also “Ducki\
3650 ng.”)"),
3651 quiz::Quiz("Signal Flow","1) In the general sense, the path that an audio signal tra\
3652 vels from the sound source to the system output. (For example, from the vocalist’s v\
3653 oice into the microphone, through the cables, into the preamp, out of the preamp int\
3654 o the console, through all inserts and buses, and output into the DAW for recording.\
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3655 ) 2) Signal flow is often specifically meant to refer to the routing of an audio sig\
3656 nal through the console, from input to output."),
3657 quiz::Quiz("Signal Processing","The practice of altering the character or sound of a\
3658 n audio signal through a variety of devices or plug-ins, such as equalizers, compres\
3659 sors, reverb units, etc."),
3660 quiz::Quiz("Signal to Noise Ratio (SNR)","The comparison of the strength of a signal\
3661 level to the amount of noise emitted by the device, expressed in dB."),
3662 quiz::Quiz("Signal","1) In audio, an alternating current (or voltage) matching the w\
3663 aveform of, or being originally obtained from, a sound pressure wave. 2) Also in aud\
3664 io, an alternating current (or voltage) between 20 Hz and 20,000 Hz. 3) A digital au\
3665 dio bit stream."),
3666 quiz::Quiz("Sine Wave","1) In the general sense, the path that an audio signal trave\
3667 ls from the sound source to the system output. (For example, from the vocalist's voi\
3668 ce into the microphone, through the cables, into the preamp, out of the preamp into \
3669 the console, through all inserts and buses, and output into the DAW for recording.) \
3670 2) Signal flow is often specifically meant to refer to the routing of an audio signa\
3671 l through the console, from input to output."),
3672 quiz::Quiz("Sine","This is the purest waveform: It contains only the fundamental har\
3673 monic, and no higher harmonics. As a result, it's a great wave to use to create a su\
3674 b bass as well as a kick drum or other pure drum tone; it's also a great source wave\
3675 to use when exploring techniques such as frequency modulation (FM), amplitude modul\
3676 ation (AM), or wavefolding which add or shift harmonic content."),
3677 quiz::Quiz("Slap Echo (also called Slapback)","A single, distinct echo of a sound, w\
3678 hich can result naturally from higher frequencies reflecting off a non-absorbent wal\
3679 l, or artificially reproduced by a signal processing unit or plugin. Slap echo creat\
3680 es a "live" sounding effect similar to what you would hear in an arena."),
3681 quiz::Quiz("Slate","Slate (Slating) - 1) In video/film, the identification of a scen\
3682 e and take at the beginning of the clip for the purpose of video editing. This is do\
3683 ne by presenting the scene/take in written form in front of the camera on a clapboar\
3684 d, calling the scene/take verbally, then marking it audibly with the clapper for the\
3685 purpose of syncing audio to the video. 2) In audio recording, the similar practice \
3686 of identifying a take of music by an audible cue at the beginning of the recorded tr\
3687 ack. While some engineers still practice this, it was more necessary in the days of \
3688 analog tape recording because it helped editors keep track of the location of takes \
3689 on the recorder. Today, DAWs make it easier to keep track by identifying each take v\
3690 isually on the screen."),
3691 quiz::Quiz("Slave","1) In audio, any device which syncs to another device by reading\
3692 the clock information emitted by the master device. 2) In MIDI, any device or instr\
3693 ument that is being operated remotely by MIDI information sent from another device." \
3694 ),
3695 quiz::Quiz("Slew Limiter","This function smoothes out an incoming signal so that the\
3696 change in voltage level cannot exceed a certain number of volts per second. As a re\
3697 sult, it is sometimes called a lag generator or processor, or more technically as an
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3698     integrator."),
3699     quiz::Quiz("Sliding Rails","This is a common system for mounting modules into a case\
3700     where the rails that the modules attach to contain channels rather than holes. A nu\
3701     mber of nuts are inserted into these channels, which can then be slid to any positio\
3702     n to accommodate the mounting hole spacing of your modules. In a Eurorack case, thes\
3703     e nuts tend to have a 2.5mm or 3mm hole and corresponding thread."),
3704     quiz::Quiz("Slope Generator","A slope generator creates ramps: rising or falling vol\
3705     tages. It is essentially a gate generator and a slew limiter (see above) wired toget\
3706     her in the same module. A common example of a slope generator is an attack/decay (AD\
3707     ) or attack/release (AR) envelope generator. However, since it can be used for gener\
3708     alized control voltage functions – even creating a sawtooth or triangle wave oscilla\
3709     tor – some companies such as Buchla and Serge referred to by its elemental function \
3710     of generating sloping voltage changes."),
3711     quiz::Quiz("Slope","Most filters typically have a cutoff or corner frequency they ar\
3712     e tuned to. It then reduces (filters) the frequency spectrum of a signal going throu\
3713     gh it so that its harmonics get progressively quieter the further away they are from \
3714     this cutoff. The strength of this effect is referred to as its slope. Most filters h\
3715     ave slopes that are defined multiples of 6 decibels (dB) weaker for each octave furt\
3716     her away you get from the cutoff frequency. For example, a low-pass filter (LPF) wit\
3717     h a slope of 24 dB/octave would attenuate harmonics one octave above its cutoff freq\
3718     uency by 24 decibels."),
3719     quiz::Quiz("Smart FSK (Frequency-Shift Key)","Smart FSK – An updated form of Frequen\
3720     cy-Shift Key (FSK) sync that enables MIDI devices to sync to analog tape recorders a\
3721     nd/or other recording devices. A digital signal with MIDI Song Position Pointer (SPP\
3722     ) data is encoded onto a spare track, which identifies the exact bar, measure and be\
3723     at for MIDI sequencers/devices at any point in the recording. This enables the devic\
3724     e to start playing at exactly the right place and tempo no matter where you start th\
3725     e tape. (See also “Frequency-Shift Key.”)"),
3726     quiz::Quiz("SMPTE Time Code","(Abbreviated “SMPTE”) A standardized timing and sync s\
3727     ignal protocol created by the Society of Motion Picture and Television Engineers for\
3728     the purpose of syncing audio to video/film, which can also be used for syncing purp\
3729     oses in audio recording environments. Many audio professionals simply refer to this \
3730     time code as “SMPTE.”"),
3731     quiz::Quiz("SMPTE","1) Abbreviation for Society of Motion Picture and Television Eng\
3732     ineers. 2) See “SMPTE Time Code.”"),
3733     quiz::Quiz("Snare","1) Abbreviation for “snare drum.” 2) The metal strands stretched\
3734     across the bottom head of a snare drum, which help produce the piercing “cracking” \
3735     sound when the snare drum is struck."),
3736     quiz::Quiz("Sock Cymbal","A rarely used alternate term for “hi-hat,” left over from \
3737     the days when hi-hat cymbals were placed at “sock level.” (See also “Hi-Hat.”)"),
3738     quiz::Quiz("Soft Knee","In compression, refers to the gradual introduction of compre\
3739     ssion of the signal once the sound level crosses the threshold. (See also “Knee.”)"),
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3741 quiz::Quiz("Software Instrument (Virtual Instrument)", "One of a number of software-b\
3742 ased synthesizers, samplers or sound samples that are stored and accessed via comput\
3743 er and performed by an external MIDI controller, rather than in a standalone synthe\
3744 izer or module. Because of the wide versatility available from these instruments, a \
3745 growing number of composers and electronic musicians are working with virtual instru\
3746 ments that can be stored in hard drives, rather than purchasing stacks of keyboards \
3747 and modules."),
3748 quiz::Quiz("Soldering", "The action of making connections with solder, a soft metal a\
3749 lloy that is used to bond two metal surfaces by melting. In audio settings, solderin\
3750 g is used for a variety of purposes in building, modifying or repairing gear—perhaps\
3751 most often to repair or build audio cables as a cost-saving effort, as opposed to b\
3752 uying new ones or sending them off for repair."),
3753 quiz::Quiz("Solid State", "In electronics, refers to the use of transistors and semic\
3754 onductors (solid materials) in the building of electronic devices, as opposed to tub\
3755 es. In the recording studio, solid state amplifiers have different properties than t\
3756 ube amps, and each has its own advantages and disadvantages. A more recent applicati\
3757 on of solid state construction is in computer devices, particularly solid state hard\
3758 drives (SSD), which transfer data more quickly than conventional spinning disc driv\
3759 es, and are less prone to breakage."),
3760 quiz::Quiz("Solo Switch", "A switch that activates the solo function on a console or \
3761 DAW."),
3762 quiz::Quiz("Solo", "1) A circuit in a console or DAW that allows one or more selected\
3763 channels to be heard or to reach the output, while other channels are automatically\
3764 muted. 2) In music, a segment of a song in which a vocalist or instrument is featur\
3765 ed above other instruments."),
3766 quiz::Quiz("Song Position Pointer (SPP)", "A MIDI message that enables connected MIDI\
3767 devices to locate a given point in the song. Used in conjunction with MIDI clock as\
3768 a way of synchronizing devices or telling a connected device when to begin playing.\
3769 "),
3770 quiz::Quiz("Sound Blanket", "A thick blanket that can be put on floors or hung to add\
3771 sound absorption to the room, and help prevent sound reflections."),
3772 quiz::Quiz("Sound Effects (SFX)", "Sounds other than dialogue, narration or music tha\
3773 t are added to audio, usually in the context of film/video."),
3774 quiz::Quiz("Sound File", "A digital audio recording that can be stored in a computer \
3775 or on a digital storage medium (such as a hard disk)."),
3776 quiz::Quiz("Sound Modeling", "A technique that recreates a sound without directly mod\
3777 eling the physical device. An example is additive synthesis, which uses a combinatio\
3778 n of sine waves and noise to recreate sounds."),
3779 quiz::Quiz("Sound Module", "An electronic instrument (tone generator, synth or sample\
3780 r playback unit) that has no playable interface, but instead responds to incoming MI\
3781 DI message. Often sound modules were created as the “brains” of popular synthesizers\
3782 , cheaper versions of the product that could be added to an existing MIDI configurat\
3783 ion. Today, sound modules can also occur as software versions or plugins to be acces\
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3784 sed on a computer."),
3785 quiz::Quiz("Sound Pressure Level (SPL)","In scientific/technical terms, the measure \
3786 of the change in air pressure caused by a sound wave, measured in dB. We hear and pe\
3787 rceive SPL in terms of amplitude, volume or loudness of the sound."),
3788 quiz::Quiz("Sound Pressure Level","In scientific/technical terms, the measure of the\
3789 change in air pressure caused by a sound wave, measured in dB. We hear and perceive\
3790 SPL in terms of amplitude, volume or loudness of the sound."),
3791 quiz::Quiz("Sound Source","The origin of a sound, whose vibrations create sound wave\
3792 s."),
3793 quiz::Quiz("Sound Wave","(Also called "Sound Pressure Wave") A wave caused by a vibr\
3794 ation that results in slight variations in air pressure, which we hear as sound."),
3795 quiz::Quiz("Soundtrack","1) Broadly speaking, refers to any/all audio that accompani\
3796 es an instance of visual media, whether music, dialogue or SFX. 2) In more common te\
3797 rms, refers to the musical score and/or licensed music synced to a film, video, TV p\
3798 rogram or video game."),
3799 quiz::Quiz("Source of Uncertainty","This was the name for the Buchla 265 and 266 mod\
3800 ules that create random control voltages. Its name is often used for random source m\
3801 odules that follow or are inspired by the original Buchla template."),
3802 quiz::Quiz("Spaced Pair","(Also called "A/B Technique") A stereo microphone placemen\
3803 t technique in which two cardioid or omnidirectional microphones are spaced somewher\
3804 e between 3-10 feet apart from each other (depending on the size of the sound source\
3805 ) to create a left/right stereo image."),
3806 quiz::Quiz("Speaker","A device that converts electrical signals to sound; more techn\
3807 ically, a transducer that changes an electrical audio signal into sound pressure wav\
3808 es."),
3809 quiz::Quiz("Speed of Sound","Generally speaking, the time it takes for a sound wave \
3810 to travel through a medium. Sound travels at different speeds through solids, liquid\
3811 s and gases, and though we usually think of sound as traveling through the air, diff\
3812 erences in temperature, air pressure and humidity can also affect how fast sound tra\
3813 vels. For a starting frame of reference, the speed of sound is generally defined by\
3814 aerospace engineers as "Mach 1.0," translating to 340.29 meters per second (approx.\
3815 761.1 mph, or 1116 feet per second), which is how fast sound travels through the ai\
3816 r at sea level at a temperature of 15 degrees Celsius (59 degrees Fahrenheit). By co\
3817 ntrast, at 70 degrees Fahrenheit under standard atmospheric conditions, the speed of\
3818 sound is about 344 m/s, or 770 mph."),
3819 quiz::Quiz("Splicing","Historically, the act of attaching previously cut pieces of a\
3820 udio tape or film in precise locations by applying a special kind of adhesive tape o\
3821 n the back. This is/was done for the purpose of shortening sections of audio or edit\
3822 ing film. Today, splicing has become a very simple process by editing sections of au\
3823 dio or video digitally with a DAW or film editing software."),
3824 quiz::Quiz("Splitter","The short definition is something that can divide a signal in\
3825 to two or more copies, such as a splitter cable where two outputs are wired to one i\
3826 nput. For a deeper discussion, see the entry on multiple, as there are ways of going\
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3827 about this beyond simple wiring."),
3828 quiz::Quiz("Spread","A few oscillator modules can produce more than one tone at the \
3829 same time. Slightly detuning or “spreading” these tones from each other creates an o\
3830 ften pleasing chorusing sound. Depending on the module, you might even be able to sp\
3831 read these tones to form intervals, triads, and chords."),
3832 quiz::Quiz("Spring Reverb","A device that simulates reverberation by creating vibrat\
3833 ions within a metal spring by attaching it to a transducer and sending the audio sig\
3834 nal through it. A pickup at the other end converts those vibrations into an electric\
3835 al signal which is mixed with the original audio signal. While the physical spring r\
3836 everbs still exist, most studios emulate spring reverb with the use of plug-ins or h\
3837 ardware reverb units."),
3838 quiz::Quiz("Square wave","This is a common waveform produced by a synthesizer’s osci\
3839 llator. It alternates between a high and low voltage (typically +/-5 or 8 volts for \
3840 an audio oscillator; sometimes low frequency oscillators go between 0v and a positiv\
3841 e voltage). Aside from being a really easy waveshape to generate with analog circuit\
3842 ry, it has an interesting harmonic series: it has a strong fundamental, then gradual\
3843 ly weaker odd harmonics: a component at three times the fundamental frequency, one a\
3844 t fives time the fundamental, and so forth. The result is a more open, hollow sound,\
3845 especially when compared to a sawtooth (ramp) wave that has both odd and even harmo\
3846 nics present. A wave shape in which the voltage rises instantly to one level, stays \
3847 at that level for a time, instantly falls to another level and stays at that level, \
3848 and finally instantly rises to its original level to complete the wave cycle."),
3849 quiz::Quiz("Stackable Cable","Many banana style cables are constructed that each plu\
3850 g has a jack built into its back, allowing you to plug another cable directly in top\
3851 of the original plug. These are used by Buchla and Serge-compatible systems. TipTop\
3852 makes a similar cable using 3.5mm plugs and jacks for Eurorack format users called \
3853 Stackables."),
3854 quiz::Quiz("Stage Monitor","A speaker on the stage that enables performers to hear t\
3855 hemselves and to hear what the other musicians are playing on stage."),
3856 quiz::Quiz("Stage","1) The partially enclosed or raised area where live musicians pe\
3857 rform. 2) In reverberation effects devices, an echo added before the reverberation t\
3858 o simulate echoes that would come from a concert stage. In the most general terms, a\
3859 stage is the next change in voltage among a series of changes. In an 8-step sequenc\
3860 er, for example, each new note that it produces in order is a stage. In an envelope \
3861 generator such as an ADSR (Attack/Decay/Sustain/Release), each phase – such as attac\
3862 k, where the envelope generally rises from 0 volts to the highest voltage it can out\
3863 put – is a stage. You might also hear it used to describe the number of sample stage\
3864 s in a BBD (Bucket Brigade Delay), described elsewhere."),
3865 quiz::Quiz("Standard Operating Level","A reference voltage level or maximum average \
3866 level that should not be exceeded in normal operation."),
3867 quiz::Quiz("Standing Wave","An unwanted sound wave pattern that often occurs when th\
3868 e sound wave bounces between two reflective parallel surfaces in a room, and the ref\
3869 lected waves interfere with the initial wave coming from the sound source, in which \
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3870 the combined wavelength of the affected frequency is effectively the length of the r\
3871 oom. This creates the audible illusion that the wave is standing still, so the frequ\
3872 ency is amplified to an unwanted level in certain parts of the room while nearly abs\
3873 ent in others. Standing waves are most common in square or rectangular rooms with pa\
3874 rallel surfaces, so acoustic designers try to prevent these waves by installing abso\
3875 rptive materials or introducing other items to offset the parallel surfaces."),
3876 quiz::Quiz("Step Mode","A setting in a sequencer or DAW in which notes are input man\
3877 ually, one note or step at a time."),
3878 quiz::Quiz("Step Sequencer","This usually refers to a type of sequencer where you st\
3879 ep to and pause on a stage, enter the note (and possibly the duration) for that stag\
3880 e, move on to the next step, and so forth."),
3881 quiz::Quiz("Step","Step is often used interchangeably with stage (see above), espec\
3882 ally when talking about sequencers."),
3883 quiz::Quiz("Stereo Image","The audible perception of stereo, in which different soun\
3884 ds sources appear to be coming from far left, far right or any place in between."),
3885 quiz::Quiz("Stereo Micing","Placement of two or more mics so that their outputs comb\
3886 ine to create a stereo image."),
3887 quiz::Quiz("Stereo","A recording or reproduction of at least two channels where posi\
3888 tioning of instrument sounds left to right can be perceived."),
3889 quiz::Quiz("Strike","This term appears on several Make Noise modules, although it ha\
3890 s been creeping into the general lingo. Some filters, amplifiers, and low pass gates\
3891 (LPGs) that use or simulate vactrols (a light sensitive resistor placed next to a l\
3892 ight source such as an LED, allowing a voltage to be turned into a resistance to con\
3893 trol a parameter) may have a strike input. When you flash an LED at a light sensitiv\
3894 e resistor, it does not change the resistance instantaneously and stay there - inste\
3895 ad, there is some delay as it glides to the desired resistance. When you turn the LE\
3896 D off, the resistance may not go instantaneously to full; instead it might take a br\
3897 ief moment to decay. These characteristics are useful for creating percussive sounds\
3898 and attacks. The purpose of a strike input is either to pass just a short pulse, or\
3899 to allow you to re-attack while the LED is otherwise still on. To put away equipmen\
3900 t and clean up after a recording session."),
3901 quiz::Quiz("Subcode","Additional information bits that are recorded alongside digita\
3902 l audio, used for control and playback purposes."),
3903 quiz::Quiz("Subframe","A unit smaller than one frame in SMPTE time code."),
3904 quiz::Quiz("Subgroup","A number of input channels on a console that can be controlle\
3905 d and adjusted as a single set before sending the combined signal to the master outp\
3906 ut. Sometimes also called "Submix," "Bus" or just "Group.""),
3907 quiz::Quiz("Subharmonic","A circuit that divides the fundamental harmonic of the inc\
3908 oming sound to produce lower frequencies, and therefore subharmonics. The most commo\
3909 n is an octave divider or sub bass circuit that divides creates a subharmonic by div\
3910 iding the fundamental by 2 (some can also create a subharmonic two octaves below the\
3911 fundamental by dividing it by 4)."),
3912 quiz::Quiz("Submaster / Sub-Master","The fader which controls the combined level of \
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3913 sound from several channels during mixdown or recording."),
3914 quiz::Quiz("Submix","See "Subgroup.""),
3915 quiz::Quiz("Suboctave","A module that creates a new tone one or two octaves below th\
3916 e fundamental harmonic – the "pitch" – of the sound coming into it, to emphasize the\
3917 bass. (Subharmonics are discussed in detail elsewhere in this glossary.) This tone \
3918 is usually a square wave, although some clever modules may create something more sin\
3919 e-like, or that more closely resembles the original waveform."),
3920 quiz::Quiz("Subtractive Synthesis","The most common synthesis technique: You start w\
3921 ith one or more oscillators outputting waveforms with a large number of harmonics, a\
3922 nd then pass this mix through a filter that removes some of the harmonics to create \
3923 the desired sound or timbre. This modified tone is then sent to an amplifier that ad\
3924 ds articulation to the note by varying its loudness. An old-school method of sound s\
3925 ynthesis in which sounds are designed and created by generating harmonically rich wa\
3926 veforms, then filtering out unwanted harmonics to arrive at the desired sound."),
3927 quiz::Quiz("Sum","To sum is a fancy way of saying you added two (or more) things tog\
3928 ether; the sum is the result. It usually is used in the context of adding together c\
3929 ontrol voltages, although it can also be used for audio or even mixes of harmonics. \
3930 The opposite is difference, which subtracts one input from another. A signal that is\
3931 the mix of the two stereo channels at equal level and in phase."),
3932 quiz::Quiz("Summing","The process of blending two or more signals into one mixed sig\
3933 nal. In summing audio, each successive channel adds volume to the overall signal, so\
3934 channels must be mixed in order to prevent peaking the combined signal."),
3935 quiz::Quiz("Super-Cardioid Pattern","A very tight cardioid microphone pattern with m\
3936 aximum sensitivity on axis and the least amount of sensitivity approximately 150 deg\
3937 rees off-axis."),
3938 quiz::Quiz("Surround Sound","A technique of recording and playback in which the list\
3939 ener hears various aspects of the sound from front to back as well as side-to-side—a\
3940 360-degree audio image, as opposed to the standard stereo left-right image. Surroun\
3941 d sound can occur in various formats with different numbers of speakers arrayed thro\
3942 ugh the room. Surround sound today is most commonly used in film and TV production." \
3943 ),
3944 quiz::Quiz("Sustain","This is a common stage of an envelope generator where a voltag\
3945 e – usually being sent to a filter's cutoff frequency or an amplifier's level – is b\
3946 eing held a steady level while a note is still being held down. The knowledge that a\
3947 note is being held is usually provided by a gate signal, that stays high as long as\
3948 a note is held down, although some envelope generators may have a dedicated time co\
3949 ntrol for how long the sustain stage should last. Envelopes that contain sustain sta\
3950 ges include the ADSR (Attack/Decay/Sustain/Release) and AR (Attack/Release, which us\
3951 ually assumes a sustain stage)."),
3952 quiz::Quiz("SVF","A state variable filter (SVF) is a common design for synth filters\
3953 . This design lends itself to allowing low pass, high pass, and bandpass all being a\
3954 vailable simultaneously. Another side effect is that they are not prone to oscillati\
3955 ng at high feedback (resonance) settings, although some have certainly figured out h\
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3956 ow to make this happen. The Oberheim SEM (Synthesizer Expander Module) filter is per\  
3957 haps the most famous state variable design."),  
3958 quiz::Quiz("Sweetening", "A vague term referring to the fine-tuning of audio in the p\  
3959 ost-production stage of recording. Effectively, any small “tweaks” to to make the au\  
3960 dio sound better is considered sweetening."),  
3961 quiz::Quiz("Switch Trigger", "Some systems – such as the original Moog modular – use \  
3962 an s-trigger (switch or shorting trigger) instead of a normal gate, which was a wire\  
3963 that was shorted to 0 volts ground, like the closing of a switch wired to ground. Y\  
3964 ou cannot interconnect these two systems without some form of conversion between the\  
3965 two, which can be as simple as a special cable."),  
3966 quiz::Quiz("Switch", "A device that makes and/or breaks electrical connections."),  
3967 quiz::Quiz("Switchable Pattern Microphone", "A microphone having the capability of tw\  
3968 o or more pickup patterns, which can be toggled by use of a switch on the microphone\  
3969 ."),  
3970 quiz::Quiz("Switching Power Supply", "A switching power supply starts by directly con\  
3971 verting the incoming high-voltage AC signal into a high-voltage DC signal. They then\  
3972 rapidly switch that output on and off to average a lower output voltage. This switc\  
3973 hed voltage is then smoothed out to create a constant DC supply at the desired volta\  
3974 ge. Switching power supplies tend to be lighter, cooler, and less expensive, at the \  
3975 cost of often higher noise – both in the output voltage, and in radio frequencies (t\  
3976 his is why they are often surrounded by a shielding cage). Many are moving to a hybr\  
3977 id power supply that combines a switcher with a small linear supply or regulator to \  
3978 get the best of both worlds."),  
3979 quiz::Quiz("Sync Pop", "A short tone (usually a sine wave at 1 kHz, and the length of\  
3980 a frame of film) that is placed exactly two seconds before the start of a piece of \  
3981 film or music. The sync pop is used to make sure that all related audio and video tr\  
3982 acks stay in sync with each other through all stages of post-production."),  
3983 quiz::Quiz("Sync24", "Sync24 is an alternate name used for the Roland-created standar\  
3984 d DIN Sync, which sends a clock signal at the rate of 24 pulses per quarter note at \  
3985 the current tempo. Korg equipment used a variation of this running at 48 pulses per \  
3986 quarter note, also known as Sync48."),  
3987 quiz::Quiz("Sync", "Sync can have two different meanings, depending on whether we're \  
3988 talking about oscillators or about clock signals. Some oscillators support a mode wh\  
3989 ere they reset their waveshapes to the beginning when they receive a signal from ano\  
3990 ther oscillator. If there is not a precise octave relationship between the two oscil\  
3991 lators, the result is a modified waveform that has been reset prematurely, following\  
3992 the frequency of the second oscillator. You can create some very cool “ripping” sou\  
3993 nds by modulating the frequency of the slave oscillator; a simple AD envelope works \  
3994 well. In the context of timing, when you are synchronizing sequencers or drum patter\  
3995 ns, it is common to send a master timing or sync signal around the modular for all t\  
3996 he relevant modules to follow. This is typically a gate or trigger signal. Short for\  
3997 “Synchronization.” In audio/studio settings, sync refers to the correlating of two \  
3998 or more pieces of audio or video in relation to each other. This can include syncing\
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3999 two recording/playback devices timed to a sync signal like SMPTE Time Code, synchro\
4000 nizing audio with video in film or TV, and many other examples. Licensing a song or \
4001 piece of music for placement in film, TV or video is also referred to as "syncing.""\
4002 ),
4003 quiz::Quiz("Synthesizer Expander Module","The Oberheim SEM (Synthesizer Expander Mod\
4004 ule) was one of their earliest products. It was an entire synthesizer voice - two os\
4005 cillators, two simple envelopes, VCA, and a very popular two-pole state variable fil\
4006 ter design with a knob that crossfaded between low pass, notch, and high pass output\
4007 s plus a separate bandpass setting - in a cube-like case. Most often today, when a m\
4008 odular manufacturer uses the magic letters \"SEM\", they're referring to a filter me\
4009 ant to emulate that in the original Oberheim synth."),
4010 quiz::Quiz("Synthesizer","A musical instrument that uses electrical oscillators to g\
4011 enerate tones artificially, either to simulate the sounds of other instruments or to\
4012 create other sounds not possible with other instruments."),
4013 quiz::Quiz("System Exclusive","System Exclusive (SysEx) - A MIDI message that will o\
4014 nly be recognized by a unit of a particular manufacturer."),
4015 quiz::Quiz("Tach/Tachometer","In analog tape recording, a device on the recorder tha\
4016 t measures and regulates tape speed by emitting pulses as the tape moves across the \
4017 head."),
4018 quiz::Quiz("Tails Out","A method of winding audio tape so that the end of the last r\
4019 ecorded selection is at the outside of the reel."),
4020 quiz::Quiz("Take Notation","Writing down the takes of the tune being recorded on a t\
4021 ake sheet or on the track log with comments. Take notation was/is recommended for an\
4022 alog tape recording, but in most studios, this function is now accomplished on the D\
4023 AW."),
4024 quiz::Quiz("Take","The recording that is done between one start and stop of a tape r\
4025 ecorder or DAW."),
4026 quiz::Quiz("Talk Box","An effects unit that enables a musician to modulate the sound\
4027 of his/her instrument via a tube placed into the mouth. Historically, talk boxes ha\
4028 ve been used as an effect for guitars, but they can be used to modify other instrume\
4029 nts, as well."),
4030 quiz::Quiz("Talkback","A microphone in the control room carried on a separate circui\
4031 t from the recorded channels, allowing the engineer to communicate with the musician\
4032 s in the live room or sound booths through the monitoring system."),
4033 quiz::Quiz("Tape Delay","A signal processing technique for creating artificial delay\
4034 or echoes by manipulating time delays with analog tape machines. This technique beg\
4035 an by routing the signal to a separate tape recorder and mixing the delayed response\
4036 back in with the signal; it then evolved to the use of dedicated machines that coul\
4037 d adjust the length of the delay by adjusting the distance between the record and pl\
4038 ayback heads. Today, most tape delay effects in the studio are simulated digitally t\
4039 hrough plug-ins in a DAW."),
4040 quiz::Quiz("Tape Guide","Any stationary or rotating device which directs the tape pa\
4041 st the heads on a tape machine, or from one reel to the other."),
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4042 quiz::Quiz("Tape Hiss","The natural high-frequency noise that occurs on analog tape \
4043 due to the magnetic particles from which the tape is made. Tape hiss constitutes mos\
4044 t of the noise floor that occurs in analog recording, and can be reduced by using ta\
4045 pe constructed of finer magnetic particles. (See also "Noise Floor.")"),
4046 quiz::Quiz("Tape Loop","A length of tape with the ends spliced together so that the \
4047 recording will play continuously."),
4048 quiz::Quiz("Tape Recording Equalization","The increase in amplitude of signals, in a\
4049 tape machine's electronics, at the high frequencies as a tape is recorded to keep h\
4050 igh-frequency signals recorded above the tape hiss."),
4051 quiz::Quiz("Telephone Filter","A filter used to simulate the audio heard through a t\
4052 elephone receiver by removing signals at frequencies below 300 Hz and above 3500 Hz.\
4053 "),
4054 quiz::Quiz("Tempo Mapping","The act of programming a sequencer or DAW to follow the \
4055 tempo variations of a recorded performance. Unlike beat mapping or beatmatching, bot\
4056 h of which effectively adjust the recording to fit a set tempo, tempo mapping adjust\
4057 s the tempo of the project (especially the MIDI instruments) to match the natural te\
4058 mpo nuances of the recorded material. (See also "Beat Mapping," "Beatmatching.")"),
4059 quiz::Quiz("Tempo","The rate at which the music moves, measured in Beats Per Minute \
4060 (BPM)."),
4061 quiz::Quiz("Terminal","1) A point of connection between two wires, including the plu\
4062 g on the end of a cable, and the jack on a piece of equipment. 2) Refers to the keyb\
4063 oard and monitor of a computer that enable the user to enter information and to acce\
4064 ss data."),
4065 quiz::Quiz("Test Oscillator","A device that generates audio waveforms at various fre\
4066 quencies for testing purposes."),
4067 quiz::Quiz("Test Pressing","One of a few initial vinyl record copies pressed from th\
4068 e first stamper made, which is listened to and visually inspected to approve the qua\
4069 lity before more copies are pressed."),
4070 quiz::Quiz("Test Tones","1) A recording of several single-frequency tones at the beg\
4071 inning of a tape reel at the magnetic reference level that will be used to record th\
4072 e program. 2) Artificially generated tones that are used to calibrate an audio syste\
4073 m."),
4074 quiz::Quiz("Thin Sound","A vague term describing an audio signal that that is lackin\
4075 g in certain frequencies, especially on the low end. Over-filtering a signal with an\
4076 EQ can produce a thin sound, for example."),
4077 quiz::Quiz("Threaded Inserts","A common system for mounting modules into a case is c\
4078 alled sliding rails or nuts. A number of nuts are inserted into these channels, whic\
4079 h can then be slid to any position to accommodate the mounting whole spacing of your\
4080 modules. Some don't like this system, so they replace the nuts with strip of metal \
4081 inserted into the channel that have been pre-drilled for the standard Eurorack mount\
4082 ing hole spacing. They may be drilled for 2.5 or 3 mm screws; pay attention when buy\
4083 ing the rails or a case that has them pre-installed."),
4084 quiz::Quiz("Three-To-One Rule","A principle of microphone placement that says when m\
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4085  ultiple mics are used at once, the distance between microphones should be at least t\  
4086 hree times the distance between each microphone and its respective sound source. The\  
4087 three-to-one rule is used to prevent phasing issues between the audio signals."),  
4088 quiz::Quiz("Three-Way Speaker","A speaker system that has separate speakers to repro\  
4089 duce the bass, mid-range and treble frequencies."),  
4090 quiz::Quiz("Threshold of Hearing","Described as the sound pressure level at which pe\  
4091 ople can hear only 50 percent of the time."),  
4092 quiz::Quiz("Threshold","A threshold is generally a voltage level a signal needs to c\  
4093 ross before a module takes an action. For example, when the output of an envelope fo\  
4094 llower (a module that creates a voltage that corresponds to the current level of an \  
4095 audio signal) rises above a threshold level, then its gate signal will go high indic\  
4096 ating a note has started. When the output of the envelope follower falls before a th\  
4097 reshhold (which may be the same or different than the note-on threshold), then the ga\  
4098 te goes low, indicating the note should be finishing. The level at which a dynamics \  
4099 processing unit will begin to change the gain of the incoming signal."),  
4100 quiz::Quiz("Throat","In a speaker, the small opening in a horn or in a driver throug\  
4101 h which the sound wave passes from the driver to the horn."),  
4102 quiz::Quiz("Through-Zero Frequency Modulation","TZFM is the abbreviation for Throug\  
4103 -Zero Frequency Modulation. Think of a patch where you feed the output of one oscill\  
4104 ator (the modulator) into the frequency control voltage input of a second oscillator\  
4105 (the carrier). As the waveform output of the modulator rises above zero volts, it i\  
4106 s added to the normal pitch control voltage for the carrier, and the pitch of the ca\  
4107 rrier goes up. As the waveform output of the modulator goes below zero, it is subtra\  
4108 cted from the normal pitch control voltage, and the pitch goes down. But what happen\  
4109 s if the result of subtracting the modulator from the pitch control goes below zero \  
4110 volts? In an oscillator that explicitly says it implements through-zero frequency mo\  
4111 dulation, the carrier will start playing backwards – in essence, a negative frequenc\  
4112 y. This generally produces a more pleasing result, and is a desirable characteristic\  
4113 for an oscillator."),  
4114 quiz::Quiz("Throw","1) In speakers and in microphones, describes the amount of unres\  
4115 tricted movement that the diaphragm can make. In microphone, this affects the mic's \  
4116 sensitivity; in speakers, it affects the distance of sound projection. (A speaker de\  
4117 signed for smaller spaces has a "short throw," while one designed for a farther proj\  
4118 ection has a "long throw." 2) In speakers, "throw" may also be used to describe the \  
4119 speaker's directional output, often based on the frequencies it emits. A horn, for e\  
4120 xample, emits high frequencies in a limited angle of direction, so it has a "long th\  
4121 row," while a subwoofer emits low frequencies in all directions and has a "short thr\  
4122 ow." 3) Something a producer, engineer or musician might do with whatever is in his\  
4123 her hand during a moment of intense frustration."),  
4124 quiz::Quiz("Tie Lines","Tie Lines – Cables with connectors at both ends, which are u\  
4125 sually run through walls or floors in the studio, for the purpose of sending signals\  
4126 between rooms. Tie lines provide a great semi-permanent way to route and configure \  
4127 signal paths quickly through various parts of the studio and help the engineer keep \  

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4128 track of signal flow."),
4129 quiz::Quiz("Timbre","This word is often used to describe the unique tonal characteri\
4130 stic of a sound you are creating, separate from its pitch or loudness. Different sou\
4131 nds, by definition, have different timbres. When you change a parameter of a sound t\
4132 hat changes its tonal characteristic – such as changing the filter cutoff, pulse wid\
4133 th, amount of wavfolding, etc. – you are changing its timbre. The timbre often chan\
4134 ges during life of a note. The sound quality that makes one instrument sound differe\
4135 nt from other instruments, even while playing the same pitch. The timbre of a trumpe\
4136 t, for example, is what makes it sound like a trumpet and not like a flute. Timbre i\
4137 s largely shaped through the presence, absence and complexity of harmonics when the \
4138 instrument is played."),
4139 quiz::Quiz("Time Code","A standardized timing signal used to help devices sync with \
4140 one another, or to sync audio to video. Common time codes used in the studio are MID\
4141 I Time Code (MTC) and SMPTE time code."),
4142 quiz::Quiz("Time Compression / Expansion","(Also called “Time Stretching” or “Time S\
4143 hifting“) The process of speeding up or slowing down an audio recording without chan\
4144 ging the pitch of the sounds."),
4145 quiz::Quiz("Time Constant","A complex mathematical ides that basically describes the\
4146 time delay between when an electrical voltage is applied to a circuit and when the \
4147 circuit responds to it."),
4148 quiz::Quiz("Tini-Jax","This is a special design of jack made by Switchcraft that is \
4149 used by Buchla (and many of their clones) to carry audio signals. They are 3.5mm in \
4150 diameter, but differ slightly physically from a common 3.5 mm jack. 1/8” plugs would\
4151 be loose in when plugged into a Tini-Jax jack; a Tini-Jax plug might not fit into o\
4152 r might even damage a 1/8” jack."),
4153 quiz::Quiz("Toms","The small drums (as little as 10 inch diameter) that mount on rac\
4154 ks above the kick drum and the large drums in a drum set."),
4155 quiz::Quiz("Tone Generator","1) A device that puts out test tones at various frequen\
4156 cies to align a tape machine or for other testing purposes. 2) The circuits in a syn\
4157 thesizer that create the audio signals put out by the unit, usually to emulate the s\
4158 ound of another instrument."),
4159 quiz::Quiz("Tone","1) Any single-frequency signal or sound. 2) The sound quality of \
4160 an instrument’s sound relative to the amount of energy present at different frequenc\
4161 ies."),
4162 quiz::Quiz("Tonguing","The technique of controlling the start of a note in a brass o\
4163 r woodwind instrument with the tongue."),
4164 quiz::Quiz("Total Harmonic Distortion (THD)","The measure of the difference between \
4165 the level of harmonic frequencies at the output stage of an amplifier as compared wi\
4166 th the input stage, a ratio expressed as a percentage. It’s a fine-tuning specificat\
4167 ion barely perceptible to many ears, but the lower the THD, the more accurately the \
4168 amplifier/speaker is reproducing the sound."),
4169 quiz::Quiz("Touch Sensitive","See “Velocity Sensitive.”"),
4170 quiz::Quiz("Track & Hold","This is a variation of a Sample & Hold. Both have two inp\
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4171 uts - a gate signal, and a voltage reference signal - and a voltage output. When a S\  
4172 ample & Hold receives a gate high signal, it freezes and outputs the voltage referen\  
4173 ce coming into the reference input. This voltage is maintained until a new gate high\  
4174 signal; gate low signals are ignored. With a Track & Hold, when the gate is high, t\  
4175 he reference input it passed along to the voltage output (this is the "tracking" pha\  
4176 se); when the gate goes low, the input voltage at that instant is frozen and maintai\  
4177 ned at the voltage output until a new gate high signal is received."),  
4178 quiz::Quiz("Track Log / Track Assignment Sheet", "Track Log/Track Assignment Sheet - \  
4179 A sheet of paper kept with a multitrack tape which tells which instrument was record\  
4180 ed on each track."),  
4181 quiz::Quiz("Track", "1) One audio recording made on a portion of the width of a multi\  
4182 track tape, or created as a digital representation using a DAW. 2) One set of contro\  
4183 l commands in a sequencer or DAW that is used to control one instrument over one MID\  
4184 I channel. 3) See "Band Track.""),  
4185 quiz::Quiz("Tracking", "Tracking usually refers to how well an oscillator follows the\  
4186 pitch control voltage (CV) sent to it. As the voltage rises, the oscillator "tracks\  
4187 " it and produces a higher pitch. Most (but not all!) synths follow a 1 volt per oct\  
4188 ave system where a rise of 1.00 volts on the pitch input should produce exactly a do\  
4189 ubling (one octave rise) in the oscillator's pitch. If this is indeed what happens, \  
4190 the oscillator has good tracking. If the oscillator goes slightly out of tune, it is\  
4191 considered a tracking error, or to have poor tracking. Sometimes you will find volt\  
4192 age-controlled filters have a "tracking" switch for a CV input where the pitch of th\  
4193 e filter's corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding chan\  
4194 ge of the pitch input. This can prevent high notes from sounding too bright without \  
4195 the bass notes sounding too dull. Sometimes you will find voltage-controlled filters\  
4196 have a "tracking" switch for a CV input where the pitch of the filter's corner freq\  
4197 uency only rises at 1/3, 1/2, or 2/3 of the corresponding change of the pitch input.\  
4198 The act of recording the individual tracks of a multitrack recording."),  
4199 quiz::Quiz("Transducer", "A device that converts energy from one medium to another. T\  
4200 ransducers are prevalent throughout the equipment in a recording studio."),  
4201 quiz::Quiz("Transient", "The initial high-energy peak at the beginning of a waveform,\  
4202 such as one caused by the percussive action of a pick or hammer hitting a string, o\  
4203 r the strike of a drum."),  
4204 quiz::Quiz("Transistor Ladder Filter", "This term is often used to describe the desig\  
4205 n of the much-loved Moog low-pass filter, which is still held up by many as being th\  
4206 e gold standard in low pass filter sound. Moog actually received a patent for this d\  
4207 esign (it has since expired); many of their competitors either outright copied it or\  
4208 did their best to emulate it."),  
4209 quiz::Quiz("Transport", "1) The portion of a tape machine that moves the tape from th\  
4210 e supply reel, past the heads, to the take-up reel. 2) The set of controls found on \  
4211 a DAW or sequencer for starting, stopping pausing, fast-forward and rewind, emulatin\  
4212 g the functions of a tape machine transport."),  
4213 quiz::Quiz("Transpose", "In the simplest terms, to transpose the pitch of a musical l\  

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4214 ine is to shift it up or down by a fixed number of semitones or octaves. This is som\
4215 etimes referred to as “chromatic” transposition. A more sophisticated variation is “\
4216 scalar” transposition where each note is shifted by a set number of scale steps; thi\
4217 s differs from chromatic transposition because some scales may have differing number\
4218 s of semitones between steps than other scales. To shift a set of musical notes by a\
4219 fixed interval. This can happen in a number of ways—for example: 1) by rewriting an\
4220 entire piece of music in a new key; 2) by shifting the tuning of an instrument so t\
4221 hat it plays at a lower or higher interval than the note played (either artificially\
4222 , as with an electronic keyboard, or by the natural tuning of a transposed instrumen\
4223 t, like a trumpet); or 3) Transposing on-the-fly, playing at a set interval above or\
4224 below what is written (also known as transposing by sight).”),
4225 quiz::Quiz("Trap","1) A filter designed to reject audio signals at certain frequenci\
4226 es. 2) An object designed with acoustically absorptive material, placed into walls t\
4227 o reduce low frequency reflections in the room (also called “bass trap”). 3) Another\
4228 word for a drum set (as in “trap set”).”),
4229 quiz::Quiz("Tremolo","This is the effect of varying the amplitude (loudness) of a no\
4230 te. A way to create this effect on a modular synth is to patch a low frequency oscil\
4231 lator (LFO) to one of the control voltage inputs on an amplifier. Tremolo is differe\
4232 nt than vibrato; the latter is a warbling in pitch rather than loudness. A wavering \
4233 or “shaking” musical effect, created either by quick reiterations of the notes (as i\
4234 n a violin tremolo) or by rapid shifts in amplitude.”),
4235 quiz::Quiz("Triangle","The triangle is a common synthesizer waveform. When selected \
4236 for the output of an oscillator, it was a more mellow sound than the standard square\
4237 or sawtooth waves, with fewer and weaker higher harmonics. It is also a popular out\
4238 put for low frequency oscillators (LFOs), as it produces a relatively smooth up and \
4239 down variation in whatever it controls, while being easier to create than the even s\
4240 moother sine wave.”),
4241 quiz::Quiz("Triangular Wave","A harmonically rich waveform that appears triangular i\
4242 n shape when depicted graphically, due to a combination of the presence of odd harmo\
4243 nics and rapid rolloff.”),
4244 quiz::Quiz("Trigger","A trigger is a very short electrical pulse signal, rising from\
4245 0 volts to a standard level such as 5 or 10 volts for a few milliseconds before fal\
4246 ling back to 0 volts. It is often used to start or “trigger” the playback of a percus\
4247 sion sound, including starting an envelope generator. They can also be used to pass\
4248 clock signals around a synth so connected modules all know when a note (or finer su\
4249 bdivision of a note) starts. A trigger usually has a fixed duration, compared to a g\
4250 ate signal which also rises from 0 volts to a higher voltage and falls back to zero \
4251 again, but which stays “high” a variable length of time depending on the length of a\
4252 note. The signal or the action of sending a signal to control the start of an event\
4253 .”),
4254 quiz::Quiz("Trim / Trim Control","A device that reduces or increases the signal stre\
4255 ngth in an amplifier, often over a restricted range. Often used interchangeably with\
4256 gain, but usually referring to fine-tuning signal strength, rather than merely ampl\
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4257 ifying it."),
4258 quiz::Quiz("Truncation","1) The shortening of an audio signal, sample or song, typic\
4259 ally by cutting off the end. 2) The dropping of bits of data when the bit resolution\
4260 is reduced (for example, from 24-bit to 16-bit), causing digital distortion unless \
4261 dithering is applied."),
4262 quiz::Quiz("Tune","The act of adjusting the pitch of a synthesizer's oscillator (the\
4263 main pitch-generating element) to match another oscillator, instrument, or referenc\
4264 e is known as tuning it."),
4265 quiz::Quiz("Tuning Fork","A metal fork with two prongs that vibrate with a fairly pu\
4266 re tone of one frequency when the fork is struck."),
4267 quiz::Quiz("Turntable","A device to support and rotate a phonograph record during pl\
4268 ayback."),
4269 quiz::Quiz("Tweeter","A speaker designed to reproduce only the higher frequencies of\
4270 the sound."),
4271 quiz::Quiz("Two Quadrant Multiplier","A two-quadrant multiplier performs a simple ve\
4272 rsion of amplitude modulation (AM), where that varies the amplitude or loudness of o\
4273 ne signal known as the carrier (typically an audio signal, swinging both above and b\
4274 elow 0 volts) with a second signal called the modulator. In the typical amplitude mo\
4275 dulation (AM) scenario, a low frequency oscillator with a positive voltage (say, bet\
4276 ween 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into th\
4277 e control input of a voltage controlled amplifier to add vibrato to an audio signal \
4278 passing through it. Any negative swings in the modulation signal are ignored; when p\
4279 atching tremolo, you may need to make sure an offset voltage is being added to your \
4280 LFO to make sure the sound doesn't cut out on the lower excursions of the LFO's wave\
4281 form. (The case where the modulator's negative as well as positive excursions are us\
4282 ed is referred to as a four quadrant multiplier.) "),
4283 quiz::Quiz("Two-Way Speaker","A speaker system with separate speakers to reproduce t\
4284 he lower frequencies (woofer) and the higher frequencies (tweeter)."),
4285 quiz::Quiz("TZFM","TZFM is the abbreviation for Through-Zero Frequency Modulation. T\
4286 hink of a patch where you feed the output of one oscillator (the modulator) into the\
4287 frequency control voltage input of a second oscillator (the carrier). As the wavefo\
4288 rm output of the modulator rises above zero volts, it is added to the normal pitch c\
4289 ontrol voltage for the carrier, and the pitch of the carrier goes up. As the wavefor\
4290 m output of the modulator goes below zero, it is subtracted from the normal pitch co\
4291 ntrol voltage, and the pitch goes down. But what happens if the result of subtractin\
4292 g the modulator from the pitch control goes below zero volts? In an oscillator that \
4293 explicitly says it implements through-zero frequency modulation, the carrier will st\
4294 art playing backwards - in essence, a negative frequency. This generally produces a \
4295 more pleasing result, and is a desirable characteristic for an oscillator."),
4296 quiz::Quiz("U","Rack-mounted equipment usually follows a standard set of dimensions,\
4297 including 19" (48.3 cm)for width, and a "rack unit" (or U for short) for height equ\
4298 aling 1.75" (4.4 cm) per U. Many common modular synthesizer formats follow the rack \
4299 unit system for standardizing module height - such as 3U (3 x 1.75 = 5.25" or 13.3 c\
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4300 m) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometime\
4301 s referred to as MU for Moog Unit)."),
4302 quiz::Quiz("Unbalanced Audio","Most audio signals are passed around on cables with t\
4303 wo wires: one for the voltage that represents the audio vibrations, and one for grou\
4304 nd. This arrangement is often referred to as unbalanced audio."),
4305 quiz::Quiz("Unbalanced Cable","A cable with two conductors (a signal wire and a grou\
4306 nd wire) and connectors on each end. Unbalanced cables are often susceptible to elec\
4307 tromagnetic interference and noise. Examples of unbalanced cables are guitar/instrum\
4308 ent cables (also called tip-sleeve or TS cables) and RCA cables."),
4309 quiz::Quiz("Unidirectional Pattern","A microphone pick-up pattern which is more sens\
4310 itive to sound arriving from one direction than from any other."),
4311 quiz::Quiz("Unipolar","Many voltages in a modular synth – including the output of an\
4312 audio oscillator, and most low frequency oscillators – fluctuates between positive \
4313 and negative voltages. This is known as a bipolar voltage. Some voltages – such as t\
4314 he output of an envelope generator – only vary between 0 volts and some maximum posi\
4315 tive voltage; this is referred to as unipolar."),
4316 quiz::Quiz("Unison","Several performers, instruments or sound sources that are sound\
4317 ing at the same time and with the same pitch."),
4318 quiz::Quiz("Unity Gain","The scenario in which there is no increase or decrease in s\
4319 ignal strength at the output of an amplifier or device compared to the signal streng\
4320 th at the input (typically described as 0 dB)."),
4321 quiz::Quiz("Unity","Usually used in the phrase “unity gain” this mean a signal keeps\
4322 the exact same level from input to output."),
4323 quiz::Quiz("Vacuum Tube","A diode, a glass tube with the gases removed, through whic\
4324 h electrical current can flow. In audio, vacuum tubes are used in amplifiers, oscill\
4325 ators, and other analog devices."),
4326 quiz::Quiz("Vamp and Fade","A method of ending the recording of a song where the mus\
4327 ic has a repeating part and the engineer reduces volume until the music fades out.")\
4328 ,
4329 quiz::Quiz("Vamp","A part of a song or chord progression that is repeated, usually a\
4330 t the end of the song, and usually the chorus or part of the chorus."),
4331 quiz::Quiz("Vari-Speed","A control on a tape machine that changes the play speed."),
4332 quiz::Quiz("Variable-D","A trademarked, patented technology of ElectroVoice in its m\
4333 icrophone designs to vary the proximity effect in its microphones. Variable-D places\
4334 several ports along the microphone body, each of which has a reduced level of sensi\
4335 tivity to higher frequencies the further they are placed from the microphone’s diaph\
4336 ragm."),
4337 quiz::Quiz("VCA Automation","A system of mix automation in some mixing consoles in w\
4338 hich sound levels or other functions are altered through the use of voltage controll\
4339 ed amplifiers."),
4340 quiz::Quiz("VCA Group","Several VCA faders that are fed control voltages from a grou\
4341 p master slide. A feature in higher-end mixing boards that enables the engineer to c\
4342 ontrol groupings of independent signals by a single fader that uses VCA to adjust th\
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4343 e voltage sent to each channel."),
4344 quiz::Quiz("Velocity Message","In synthesizers and keyboard controllers, a MIDI mess\
4345 age that transmits data on how hard the key was struck. Velocity messages can be use\
4346 d to transmit volume information, as well as triggering different samples on a multi\
4347 -sampled instrument patch."),
4348 quiz::Quiz("Velocity Microphone","See “Pressure-Gradient Microphone.””),
4349 quiz::Quiz("Velocity Sensitive","(Also called “Touch Sensitive”) A feature on a MIDI\
4350 instrument such as a keyboard that transmits a MIDI velocity message depending on h\
4351 ow hard the key is struck."),
4352 quiz::Quiz("Vibrato","A smooth and repeated changing of the pitch up and down from t\
4353 he regular musical pitch, often done by singers or performed by string and wind play\
4354 ers."),
4355 quiz::Quiz("Virtual Instrument","(Also called Software Instrument) One of a number o\
4356 f software-based synthesizers, samplers or sound samples that are stored and accesse\
4357 d via computer and performed by an external MIDI controller, rather than in a standa\
4358 lone synthesizer or module. Because of the wide versatility available from these ins\
4359 truments, a growing number of composers and electronic musicians are working with vi\
4360 rtual instruments that can be stored in hard drives, rather than purchasing stacks o\
4361 f keyboards and modules."),
4362 quiz::Quiz("Vocal Booth","A room in the recording studio that is used for recording \
4363 vocals in isolation. This practice prevents bleed-through of the sounds of other ins\
4364 truments into the vocal microphone, and also reduces natural ambience and reverberat\
4365 ion in the vocal recording."),
4366 quiz::Quiz("Vocoder","An audio processing device effects device or plug-in that anal\
4367 yzes the characteristics of an audio signal and uses them to affect another synthesi\
4368 zed signal. Primarily developed for the purpose of producing synthesized voice effec\
4369 ts from human speech, a vocoder creates the characteristic robotic vocal sound or th\
4370 e “human synthesizer” effect that makes it sound like the synth is speaking or singi\
4371 ng words."),
4372 quiz::Quiz("Voice Over","The recording of vocal announcements or narration over a be\
4373 d of music in video, film or commercials."),
4374 quiz::Quiz("Voice","1) Besides the obvious definition of the sound humans make from \
4375 their mouths...in synthesizers, a voice refers to one of a number of sounds/pitches th\
4376 at may be played at the same time. “Monophonic” means only one voice plays at a time\
4377 , while “polyphonic” means multiple voices can sound at once. (See also “Polyphonic”\
4378 , “Monophonic.”) 2) In some synthesizers, like Yamaha, “voice” may also refer to a s\
4379 pecific sound patch available on the synth."),
4380 quiz::Quiz("Volatile Memory","Computer memory whose data will will be lost when the \
4381 computer is turned off. RAM (Random Access Memory) is the most common form of volati\
4382 le memory."),
4383 quiz::Quiz("Voltage Controlled Amplifier (VCA)","An amplifier whose gain level is af\
4384 fected by an external voltage being sent to it. VCAs are commonly used in synthesize\
4385 rs, signal processors, and as a means of automation for some mixing consoles."),
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4386 quiz::Quiz("Voltage Controlled Filter","A filter (especially a low-pass filter) that\  
4387 will change its cutoff frequency according to a control voltage fed to its control \  
4388 input."),  
4389 quiz::Quiz("Voltage Controlled Oscillator (VCO)","An oscillator whose frequencies ar\  
4390 e modified by voltage input. Most commonly found in synthesizers."),  
4391 quiz::Quiz("Voltage","The difference in electrical force or pressure (“potential”) b\  
4392 etween two objects, causing a flow of electric current between them."),  
4393 quiz::Quiz("Volume Unit (VU)","A unit to measure perceived loudness changes in audio\  
4394 . The unit is basically the decibel change of the average level as read by a VU Mete\  
4395 r. (See also “VU Meter.”)"),  
4396 quiz::Quiz("Volume","A common, non-technical term that either refers to sound pressu\  
4397 re level (which we hear as loudness), or to audio voltage level."),  
4398 quiz::Quiz("Vox","A Latin word meaning “voice,” often used as an abbreviation for tr\  
4399 ack logs in the studio."),  
4400 quiz::Quiz("VU Meter","A meter that reads audio voltage levels in or out of a piece \  
4401 of equipment and is designed to match the ear’s response to sudden changes in level.\  
4402 "),  
4403 quiz::Quiz("Watt","Unit of electrical power."),  
4404 quiz::Quiz("Wave","This is the pattern of vibrations – up and down fluctuations in v\  
4405 oltage – output by an oscillator. Different patterns generate different sounds."),  
4406 quiz::Quiz("Wavefolder","A wavefolder is a very specific design of waveshaper that u\  
4407 ses a comparator and some other circuitry. What they do is look to see if the wave g\  
4408 oes above (or below) a specific threshold. When it does, instead of clipping off the\  
4409 top and bottom of the wave, they create a mirror image of it and reflect that porti\  
4410 on of the wave back upon itself, creating more high harmonics and interesting spectr\  
4411 a in the process."),  
4412 quiz::Quiz("Waveform","This is the pattern of vibrations – up and down fluctuations \  
4413 in voltage – output by an oscillator. Different patterns generate different sounds. \  
4414 A visual representation or graphic of a sound wave, audio signal or other type of wa\  
4415 ve, showing the wave’s oscillations above and below the zero line."),  
4416 quiz::Quiz("Wavelength","The physical length of one cycle of a wave, measured in fee\  
4417 t, inches, etc. The longer the wavelength of a sound wave, the lower its frequency; \  
4418 the shorter the wavelength, the higher the frequency."),  
4419 quiz::Quiz("Waveshaper","It would be a bit obvious to say “a circuit that changes th\  
4420 e shape of the waveform going through it”, but that is the point. Waveshapers often \  
4421 have specific goals in mind, such as converting an incoming triangle wave into an ou\  
4422 tgoing sine wave, or to add tube-like soft clipping to the peaks and transients of w\  
4423 aves. Many waveshapers are simply intended to mangle (er, add higher harmonics to) w\  
4424 aveforms in interesting ways, creating noisier (er, more complex and bright) harmoni\  
4425 c spectra to create new sounds."),  
4426 quiz::Quiz("Wavetable","This term can have two related but slightly different meanin\  
4427 gs. A digital oscillator often produces sound by reading a table of numbers in order\  
4428 , jumping from the level described by one number to the next. This table of numbers \  

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4429 describes one cycle of a wave, and therefore is often called a wavetable. Many digital\
4430 al oscillators have multiple wave tables lined up, and can move between these tables\
4431 - either by jumping suddenly (which the original PPG Wave synths did), or by crossf\
4432 ading between them (what most digital wavetable oscillators today do). Some people r\
4433 efer to each table as a “wave” and a set of individual waves as a wavetable.”),
4434 quiz::Quiz("Weighting","An equalization curve used in audio tests that compensates f\
4435 or the Fletcher Munson Curve at various levels. (See also “Fletcher-Munson Curves.”)\
4436 "),
4437 quiz::Quiz("West Coast Synthesis","The so-called \"West Coast\" approach to synthesi\
4438 s - traditionally associated with companies such as Buchla and Serge - is often base\
4439 d around adding harmonics to simple waveforms, rather than removing (filtering) them\
4440 from complex waveforms. This is often accomplished by using a pair of oscillators (\
4441 sometimes combined into what's called a \"complex oscillator\") where one modulates \
4442 the frequency (FM) or amplitude (AM) of the other; another common West Coast module \
4443 is a waveshaper or a wavefolder. You may also find two-stage envelope generators suc\
4444 h as an AD or AR (often called slope generators) rather than four-stage ADSRs, as we\
4445 ll as more of an emphasis on control voltage manipulation, A common feature is also \
4446 voltage controlled amplifiers that have low-pass filters built into them, creating w\
4447 hat's known as a Low Pass Gate (LPG). The West Coast approach also embraces non-trad\
4448 itional controllers, such as touch plates and the such. Today it's common to mix bot\
4449 h East Coast and West Coast approaches in the same system."),
4450 quiz::Quiz("wet sound","Sometimes people will say a filter has a “wet” sound. This u\
4451 sually refers to a fewer-than-4-pole filter sound - often low or bandpass - with res\
4452 onance turned up a bit, but not to the point of self-oscillation. It’s a sound that \
4453 is popular in acid house and other similar techno styles."),
4454 quiz::Quiz("Wet","A sound with effects (such as reverb) mixed is referred to as \"we\
4455 t\"; a sound with no effects is referred to as \"dry.\" Effects units or mixers ofte\
4456 n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
4457 d and the fully-effected sound. Refers to a signal that has the full amount of an ef\
4458 fect (like reverb) applied to it, as opposed to “dry,” which refers to the un-effect\
4459 ed sound. Many times, the preferred sound in mixing will be a blend of wet and dry s\
4460 ignals. (See also “Dry.”)"),
4461 quiz::Quiz("White Noise","Noise is a random signal that does not have a distinct pit\
4462 ch, such as hissing, breath noise, or the sound of wind or the surf. Noise is often \
4463 described by different “colors” such as white, pink, red, or blue which have differe\
4464 nt frequency distributions. White noise has equal power per unit of frequency (such \
4465 as every 1000 hertz), resulting in a brighter, hissier sound. A noise signal contain\
4466 ing an equal spread of energy across all audible frequencies. Like pink noise, engin\
4467 eers often send a white noise signal through audio equipment for tuning and calibrat\
4468 ion purposes, or in EQ-ing a live audio space. (See also “Pink Noise.”)"),
4469 quiz::Quiz("Whole Step","A change in pitch equivalent to two half steps, or the diff\
4470 erence in pitch between two piano keys."),
4471 quiz::Quiz("Wild Sound","In film and video, audio that is recorded separately from t\
```

```
4472 he visual that may be added to the audio track later, and does not need to be synchr\
4473 onized with the picture."),
4474 quiz::Quiz("Wind Controller","A device that is played like a wind instrument to cont\
4475 rol a synthesizer, module or virtual instrument via MIDI signals, as opposed to a ke\
4476 yboard controller."),
4477 quiz::Quiz("Windscreen","A covering that fits over a microphone to reduce the excess\
4478 ive noise resulting from wind blowing into the mic. Typically used for recording in \
4479 outdoor locations."),
4480 quiz::Quiz("Wireless Microphone","A microphone that transmits its signal over an FM \
4481 frequency to a receiver offstage, rather than traveling over an audio cable."),
4482 quiz::Quiz("Woofers","A speaker that is designed to reproduce bass frequencies only."),
4483 ),
4484 quiz::Quiz("Write Mode","A mode of operation in an automated console where the engin\
4485 eer is in control of channel gain and the computer is recording the gain changes ove\
4486 r time."),
4487 quiz::Quiz("XLR Cable","A balanced microphone cable utilizing XLR connectors. (See a\
4488 lso "XLR Connector.")"),
4489 quiz::Quiz("XLR Connector","A balanced cable connector consisting of 3 or 7 pins, mo\
4490 st commonly used in microphone cables."),
4491 quiz::Quiz("XY Miking","A coincident stereo microphone placement technique in which \
4492 two cardioid microphones are placed with their heads toward each other at a 90-degre\
4493 e angle, and as close together as possible. (See also "Coincident Miking.")"),
4494 quiz::Quiz("Y-Cord","A cable with three connectors so that one output may be sent to\
4495 two inputs. Basically, a signal splitter done with spliced wires rather than compo\
4496 nents."),
4497 quiz::Quiz("Zenith","In analog tape recording, refers to the tilt of the tape head i\
4498 n the direction perpendicular to the tape travel."),
4499 quiz::Quiz("Zero-Order Hold (ZOH)","Refers to the mathematical expression of the sig\
4500 nal processing done by a conventional digital-to-analog converter (DAC)."),
4501 };
4502
4503
4504 int main()
4505 {
4506     std::random_device rd;
4507     std::mt19937 gen(rd());
4508     std::uniform_int_distribution<> distria(1, 4);
4509     std::uniform_int_distribution<> distrib(0, game.size()-1);
4510     std::shuffle(std::begin(game), std::end(game), std::default_random_engine());
4511     std::vector<std::string> answers;
4512     std::string question;
4513     uint32_t n;
4514     uint8_t correct;
```

```

4515     uint32_t score=0;
4516     uint32_t tq=0;
4517
4518     for (uint32_t ctr=0;ctr<game.size();++ctr) {
4519         answers.clear();
4520         correct=distria(gen);
4521         for (uint8_t i=1;i<=4;++i) {
4522             if (i == correct) {
4523                 answers.push_back(game[ctr].getQ());
4524                 question=game[ctr].getA();
4525             } else {
4526                 answers.push_back(game[distrib(gen)].getQ());
4527             }
4528         }
4529         std::cout << "\33c\e[3J";
4530         if (tq != 0) {
4531             std::cout << "[QUESTIONS: " << tq << " / " << game.size() << " SCORE: " <<
4532 << "]\n";
4533         }
4534         std::cout << "Question #" << tq+1 << ": " << question << "\n\n";
4535         std::cout << "Answer #1.\n" << answers[0] << "\n\n";
4536         std::cout << "Answer #2.\n" << answers[1] << "\n\n";
4537         std::cout << "Answer #3.\n" << answers[2] << "\n\n";
4538         std::cout << "Answer #4.\n" << answers[3] << "\n\n";
4539         std::cout << "What answer is correct (q=quit)? ";
4540         std::cin >> n;
4541         if (n == 0) {
4542             break;
4543         } else if (n == correct) {
4544             score++;
4545         }
4546         tq++;
4547         std::cout << n << " is the answer you gave. And the correct answer is: " << correc\
4548 t << '\n';
4549     }
4550
4551     std::cout << "\33c\e[3J";
4552     if (tq != 0) {
4553         std::cout << "[QUESTIONS: " << tq << " / " << game.size() << " SCORE: " << score\
4554 << "]\n";
4555         std::cout << "[" << ((double(score))/double(tq))*100.0 << "% correct answers.]\n\
4556 ";
4557     }

```

```

4558         return 0;
4559     }

```

---

**quiz2.cpp - 338681 bytes.**

---

```

1  // compile: clang++ -std=c++20 quiz2.cpp -o quiz2
2  #include <iostream>
3  #include <fstream>
4  #include <vector>
5  #include <algorithm>
6  #include <random>
7
8  namespace quiz
9  {
10     class Quiz
11     {
12     public:
13         Quiz(const std::string &q, const std::string &a) {_a=a;_q=q;}
14         virtual ~Quiz() {}
15         std::string getQ() {return _q;}
16         std::string getA() {return _a;}
17     private:
18         std::string _q;
19         std::string _a;
20     };
21 }
22
23 std::vector<quiz::Quiz> game{
24     quiz::Quiz("0-5v","Denotes a range of 0 to 5 volts, which is common for gates, trigg\
25     ers, and modulation control voltages in modular synthesizers. Gates and triggers - w\
26     high initiate events such as new notes - typically rise from 0v to 5v (0 to 10v is a\
27     lso common), with roughly the middle of that onset starting the event. Gates are con\
28     sidered high when held at 5v (or 10v), and then low when they return to 0v."),
29     quiz::Quiz("1 pole","This format of numbers and abbreviations (dB/oct = decibels per\
30     octave) is often used to refer to the frequency response behavior of a filter. A fi\
31     lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
32     ters) the frequency spectrum of a signal going through it so that its loudness is mu\
33     ltiples of 6 decibels weaker for each octave further away you get from the cutoff fr\
34     equency. A 6dB/octave filter is often referred to as a "one pole" filter (as each po\
35     le of a filter's design results in 6dB of attenuation), and has a relatively weak ef\
36     fect on the signal going through it. Low Pass Gates (LPGs) typically - but not alway\
37     s - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 deci\
38     bels for every octave above its cutoff frequency."),

```

```
39 quiz::Quiz("1 ppqn","The most common sequencer clock division forwards it one step (\
40 pulse) per quarter note. This is often the core sync pulse that is distributed in a \
41 modular system, and is either multiplied or divided to create other musical division\
42 s."),
43 quiz::Quiz("1 v/oct","The most common standard for controlling pitch in a modular sy\
44 nthesizer. Under the system, increasing the voltage going into a VCO (Voltage Contro\
45 lled Oscillator) 1 volt – say, from 0.5v to 1.5v – would raise its pitch by one octa\
46 ve."),
47 quiz::Quiz("1.2 v/oct","Buchla compatible synths have standardized on the 1.2 volt p\
48 er octave system, instead of the more common 1 v/oct. With 12 semitones to an octave\
49 in Western music, an equally tempered scale would work out to precisely 0.1 volts f\
50 or a change in pitch of 1 semitone."),
51 quiz::Quiz("1/4\"", "The most common connector size used for 5U (Moog format) modular \
52 synthesizers. These are TS (tip/sleeve) jacks and plugs, similar to guitar and other\
53 instrument cables."),
54 quiz::Quiz("1/8\"", "Often used to incorrectly describe the connector size commonly us\
55 ed in Eurorack format modules, as well as Buchla audio signals. In fact, Eurorack mo\
56 dules use 3.5mm jacks and plugs (slightly larger than 1/8"); Buchla uses Switchcraft\
57 Tini-Jax connectors. Tini-Jax are 3.5mm in diameter, but are slightly different phy\
58 sically from a common 3.5 mm jack. 1/8" plugs would be loose in both of these jacks,\
59 so make sure you get 3.5mm connectors ordering parts or cables for these formats.")\
60 ,
61 quiz::Quiz("10 vpp","An abbreviation for \"10 volts peak to peak\" with peak to peak\
62 being the difference between the lowest and highest voltage reached during a signal\
63 's travels. This is a common voltage range for both audio and modulation signals in \
64 a modular synthesizer. The actual range is between -5 and +5 volts. The precise rang\
65 e may be varied to change the depth of their effect, so don't get too hung up on spe\
66 cific voltage ranges. Pay more attention to whether they vary between 0v and some va\
67 lue, or swing in roughly equal amounts both above and below 0v (as 10vpp does)."),
68 quiz::Quiz("12 dB/oct","This format of numbers and abbreviations (dB/oct = decibels \
69 per octave) is often used to refer to the frequency response behavior of a filter. A\
70 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\
71 filters) the frequency spectrum of a signal going through it so that its loudness is\
72 multiples of 12 decibels weaker for each octave further away you get from the cutoff\
73 frequency. A 12dB/octave filter is often referred to as a "two pole" filter (as ea\
74 ch pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and \
75 Oberheim instruments often featured 2-pole filters, often resulting in brighter soun\
76 ds when compared to those with 4-pole instruments."),
77 quiz::Quiz("16'", "Sometimes seen on octave selector switches on oscillators. It refe\
78 rs to the length of an organ pipe. Longer pipes = lower pitches; 16' is in the mid-b\
79 ass range. A pipe or setting half as long (8') is one octave higher; a pipe half as \
80 long again (4') is two octaves higher; etc."),
81 quiz::Quiz("18 dB/oct","This format of numbers and abbreviations (dB/oct = decibels \
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82 per octave) is often used to refer to the frequency response behavior of a filter. A\  
83 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\  
84 filters) the frequency spectrum of a signal going through it so that its loudness is\  
85 multiples of 18 decibels weaker for each octave further away you get from the cutoff\  
86 f frequency. It is often used a coded shorthand for when someone wants to refer to a\  
87 cid-type bass lines from a TB-303 without mentioning the instrument by name."),  
88 quiz::Quiz("2 Pole","This format of numbers and abbreviations (dB/oct = decibels per\  
89 octave) is often used to refer to the frequency response behavior of a filter. A fi\  
90 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\  
91 ters) the frequency spectrum of a signal going through it so that its loudness is mu\  
92 ltiples of 12 decibels weaker for each octave further away you get from the cutoff f\  
93 requency. A 12dB/octave filter is often referred to as a "two pole" filter (as each \  
94 pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and Obe\  
95 rheim instruments often featured 2-pole filters, often resulting in brighter sounds \  
96 when compared to those with 4-pole instruments."),  
97 quiz::Quiz("2.5 mm","A common screw thread size used to mount Eurorack modules. This\  
98 size is most common when using a system of loose nuts that slide along the rails th\  
99 at the modules are attached to."),  
100 quiz::Quiz("24 dB/oct","This format of numbers and abbreviations (dB/oct = decibels \  
101 per octave) is often used to refer to the frequency response behavior of a filter. A\  
102 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (\  
103 filters) the frequency spectrum of a signal going through it so that its loudness is\  
104 multiples of 24 decibels weaker for each octave further away you get from the cutoff\  
105 f frequency. This design is often used in vintage Moog and Roland synths. 4-pole fil\  
106 ters are often associated with subjectively fatter, more "round" sounds than 2-pole \  
107 filters - but generalizations are always dangerous."),  
108 quiz::Quiz("24 ppqn","A common master clock division used in MIDI, DIN sync, and oth\  
109 er systems common to electronic music and synthesizers. It means internally, 24 subd\  
110 ivisions of time are counted for every quarter note at the current tempo. This fast \  
111 internal clock can then be divided down to create sixteenth notes ( $\div 6$ ), eighth notes\  
112 ( $\div 12$ ), eight note triplets ( $\div 8$ ), etc."),  
113 quiz::Quiz("2'", "Sometimes seen on octave selector switches for oscillators. It refe\  
114 rs to the length of an organ pipe. Shorter pipes = higher pitches; 2' is rarely seen\  
115 on modular oscillators as it's rather high in pitch - two octaves above middle C as\  
116 a starting point. A pipe or setting twice as long (4') is one octave lower; a pipe \  
117 twice as long again (8') is two octaves lower; etc."),  
118 quiz::Quiz("3 mm","A common screw thread size used to mount Eurorack modules. This s\  
119 ize is most common when using module mounting rails that have been pre-drilled."),  
120 quiz::Quiz("3 Pole","This format of numbers and abbreviations (dB/oct = decibels per\  
121 octave) is often used to refer to the frequency response behavior of a filter. A fi\  
122 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\  
123 ters) the frequency spectrum of a signal going through it so that its loudness is mu\  
124 ltiples of 18 decibels weaker for each octave further away you get from the cutoff f\
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125 reQUENCY. It is often used a coded shorthand for when someone wants to refer to acid\
126 -type bass lines from a TB-303 without mentioning the instrument by name."),
127 quiz::Quiz("3.5 mm", "The standard connector size used for jacks and cables in Eurora\
128 ck format modular synthesizers. Note that this is slightly larger than 1/8")."),
129 quiz::Quiz("303", "The TB-303 Bass Line by Roland became a cult favorite in Acid Hous\
130 e and other flavors of EDM (Electronic Dance Music) for its rubbery, slithery synth \
131 bass sound. Many attribute the sound of the 303 to its filter design;"),
132 quiz::Quiz("32'", "Sometimes seen on octave selector switches on oscillators. It refe\
133 rs to the length of an organ pipe. Longer pipes = lower pitches; 32' is the lowest s\
134 etting you will see and is getting into earthquake territory. A pipe or setting half\
135 as long (16') is one octave higher; a pipe half as long again (8') is two octaves h\
136 igher; etc."),
137 quiz::Quiz("3U", "Refers to modules that are 3 rack units (U) high – the Eurorack sta\
138 ndard, which is by far the most common modular format today, even though it's one of\
139 the youngest formats."),
140 quiz::Quiz("4 Pole", "This format of numbers and abbreviations (dB/oct = decibels per\
141 octave) is often used to refer to the frequency response behavior of a filter. A fi\
142 lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
143 ters) the frequency spectrum of a signal going through it so that its loudness is mu\
144 ltiples of 24 decibels weaker for each octave further away you get from the cutoff f\
145 reQUENCY. This design is often used in vintage Moog and Roland synths. 4-pole filter\
146 s are often associated with subjectively fatter, more “round” sounds than 2-pole fil\
147 ters – but generalizations are always dangerous."),
148 quiz::Quiz("4-40", "A screw thread size occasionally used to mount Eurorack modules. \
149 This size is used by Pittsburgh Modular for their cases, for example."),
150 quiz::Quiz("4U", "Refers to modules that are 4U (rack units) high – namely, Buchla an\
151 d Serge systems, as well as do-it-yourself clones of these modules. Both Buchla and \
152 Serge lean toward a more experimental approach to synthesis and music, so some users\
153 wear “4U” as a badge of honor that they're non-conformist and cool. (And they are.)\
154 "),
155 quiz::Quiz("4'", "Sometimes seen on octave selector switches on oscillators. It refer\
156 s to the length of an organ pipe. Shorter pipes = higher pitches; 4' is the highest \
157 octave setting you will see on most oscillators. A pipe or setting twice as long (8'\
158 ) is one octave lower; a pipe twice as long again (16') is two octaves lower; etc.")\
159 ,
160 quiz::Quiz("5U", "Refers to modules that are 5U (rack units) or 8.75” (22.2 cm) high,\
161 which is most often associated with the vintage Moog standard and those who have fo\
162 llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
163 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
164 fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
165 d is both historical and physically large, some users “5U” as a badge of honor that \
166 they're traditional and cool. (And they are.) There was also a briefly popular 5U for\
167 mat from MOTM that used a different width and power connection. It has since been di\
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168 scontinued, but there are still diehard MOTM format users today."),
169 quiz::Quiz("6 dB/oct","This format of numbers and abbreviations (dB/oct = decibels p\
170 er octave) is often used to refer to the frequency response behavior of a filter. A \
171 filter typically has a cutoff or corner frequency it is tuned to. It then reduces (f\
172 ilters) the frequency spectrum of a signal going through it so that its loudness is \
173 multiples of 6 decibels weaker for each octave further away you get from the cutoff \
174 frequency. A 6dB/octave filter is often referred to as a "one pole" filter (as each \
175 pole of a filter's design results in 6dB of attenuation), and has a relatively weak \
176 effect on the signal going through it. Low Pass Gates (LPGs) typically - but not alw\
177 ays - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 de\
178 cibels for every octave above its cutoff frequency."),
179 quiz::Quiz("808","The TR-808 Rhythm Composer by Roland created all of its sounds usi\
180 ng analog circuitry. When it first came out, it was not well loved, as the analog so\
181 unds weren't realistic enough. But later, music styles such as House and Hip-Hop ado\
182 pted its big, booming synthetic sounds. When a module says it recreates "808" drums,\
183 this is the instrument they are trying to emulate. Most copied is the 808 kick drum\
184 sound, which tends to be a low-pitched, long-decaying sine-like wave often with a s\
185 nappy attack."),
186 quiz::Quiz("8'", "Sometimes seen on octave selector switches on oscillators. It refer\
187 s to the length of an organ pipe. Shorter pipes = higher pitches; 8' is typically as\
188 sociated with middle C. A pipe or setting half as long (4') is one octave higher; a \
189 pipe or setting twice as long (16') is one octave lower."),
190 quiz::Quiz("909","The TR-909 Rhythm Composer was the follow-up to Roland's now-rever\
191 ed TR-808. It combined digital samples for the hi-hat and cymbal along with the 808'\
192 s analog sounds, and has also become popular. When a module says it produces 909-lik\
193 e sounds, this is the instrument it is referencing."),
194 quiz::Quiz("A-440","This is the frequency in hertz (cycles per second) of the A abov\
195 e Middle C. It is often used as a tuning reference."),
196 quiz::Quiz("A/B Technique","A stereo microphone placement technique in which two car\
197 dioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart fr\
198 om each other (depending on the size of the sound source) to create a left/right ste\
199 reo image. Also known as Spaced Pair."),
200 quiz::Quiz("A/D","Abbreviation of Analog-to-Digital Conversion, the conversion of a \
201 quantity that has continuous changes (like electrical signals) into numbers that app\
202 roximate those changes (i.e., computer data)."),
203 quiz::Quiz("Absolute Phase","This term describes a perfect polarity between an origi\
204 nal signal (into the microphone) and the reproduced signal (through the speaker). Wh\
205 en positive pressure exerted upon the microphone is translated as positive pressure \
206 to the loudspeaker, the two are in "absolute phase."."),
207 quiz::Quiz("Absorption","In acoustics, absorption is what happens when sound waves a\
208 re absorbed by a surface, as opposed to bouncing off the surface (reflection). Absor\
209 ptive materials in a control room, for example, tend to "deaden" the sound of the ro\
210 om because the sound energy is absorbed rather than reflected. (See also "Reflection\
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211 .”)”),
212 quiz::Quiz("AC Coupled","An AC coupled input attempts to remove any constant DC volt\
213 age going through it. This is useful if have an audio signal (such as the output of \
214 an oscillator) which is AC in nature, and you want to remove any accidental DC offse\
215 t that might have crept into it. These offsets can cause one half of the AC waveform\
216 to clip prematurely, or can cause clicks at the start and end of envelopes or mutes\
217 . However, this coupling can mildly distort a wave going through it, as in essence A\
218 C coupling is a high pass filter that is attempting to remove very low frequency com\
219 ponents."),
220 quiz::Quiz("AC","Alternating Current - The type of electrical current found in stand\
221 ard electrical outlets and studio signals running through audio lines. In AC, the cu\
222 rrent “alternates” directions, flowing back and forth through the circuit. In modula\
223 r terms, AC refers to a voltage that alternates between positive and negative values\
224 - such as the output of an oscillator."),
225 quiz::Quiz("Accelerometer","A device that measures the acceleration to which it is s\
226 ubjected and creates an electric signal to match it. In music and audio, acceleromet\
227 ers are found in such things as microphones and guitar pickups."),
228 quiz::Quiz("Acorn Tube","Named for its acorn-like shape, an acorn tube is a small va\
229 cuum tube used in ultra high frequency (UHF) electronics such as tube amplifiers."),
230 quiz::Quiz("Acoustic Amplifier","The part of a musical instrument that vibrates in r\
231 esponse to the initial vibration of the instrument, causing the surrounding air to m\
232 ove more efficiently and making the sound louder. For example: the body of an acoust\
233 ic guitar, the bell of a horn, a drum’s shell, and the wooden soundboard of a piano.\
234 "),
235 quiz::Quiz("Acoustic Echo Chamber","A room designed with hard, non-parallel surfaces\
236 to create reverberation. In recording studios, they are used to add natural reverb \
237 to a dry signal."),
238 quiz::Quiz("Acoustics","The science of the sound—more specifically, the science of t\
239 he properties and behavior of sound waves. A good understanding of acoustics is esse\
240 ntial to audio engineering and studio design."),
241 quiz::Quiz("Active Device","A component that is designed with the ability to control\
242 electrical current (as opposed to a “Passive Device”). In the recording studio, act\
243 ive devices are generally components that include an amplifier. (See also “Passive D\
244 evice.”)”),
245 quiz::Quiz("Active Multiple","Quite often you need to split or copy a signal to send\
246 to more than one destination. This is commonly done with a multiple, where you plug\
247 one source in, and then plug in additional patch cables to go off to multiple desti\
248 nations. An active or buffered multiple is one that includes a buffer circuit betwee\
249 n the input and output, making sure the signal does not lose its strength or integri\
250 ty by being split too many times, and that no funny business happening on one of the\
251 outputs affects any of the other connections. Some modules have good buffering buil\
252 t into their outputs, and can drive multiple modules without issue. But if you try t\
253 o use a passive mult to connect to, say, three oscillators, and you realize the trac\
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254 king isn't very good (they quickly go out of tune as you go up and down the scale), \
255 then you need a buffered mult instead."),
256 quiz::Quiz("Actuator","The part of a switch that causes change of the contact connec\
257 tions (e.g., toggle, pushbutton, or rocker)."),
258 quiz::Quiz("AD","Shorthand for a two-stage Attack/Decay envelope. This simple envelo\
259 pe shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 vol\
260 ts) at a speed defined by its Attack parameter, and then immediately falls back to 0\
261 volts at a rate defined by its Decay parameter. A variation on this is the AHD envel\
262 lope: After finishing the Attack stage, it holds at the maximum level for a specifie\
263 d amount of time (in contrast to an AR envelope, which holds at the maximum level fo\
264 r as long as the note on gate is high), and then decays back to zero. I have heard t\
265 here are some envelopes that a hybrid of AHD and AR in that they hold the maximum le\
266 vel for either the defined Hold time or the as long as the incoming gate is high;"),
267 quiz::Quiz("Additive Synthesis","One of the main properties that make a sound unique\
268 is the mixture of harmonics - pure component frequencies - that it is built from. A\
269 dditive synthesis is a technique that gives you direct control over each of those co\
270 mponent harmonics, allowing you to directly dial in the mix you want. As immediate a\
271 nd intuitive as that sounds on paper (or on screen), in reality it takes a lot of wo\
272 rk to craft the correct mixture to recreate another sound, especially since the stre\
273 ngth of each harmonic usually varies over time. Additive synthesis oscillators are r\
274 elatively rare in modular synths; two examples are the Verbos Harmonic Oscillator an\
275 d the Make Noise TELHARMONIC."),
276 quiz::Quiz("ADSR","An envelope generator with four stages: Attack, Decay, Sustain, a\
277 nd Release. When this envelope generator receives a gate input, it typically starts \
278 at 0 volts (which is the equivalent of silence when connected to a Voltage Controlle\
279 d Amplifier, or the lowest frequency when connected to a voltage controlled filter o\
280 r oscillator) and raises to the maximum voltage it can output (typically 5 to 10 vol\
281 ts depending on system; it can often be set with an output level control) over a tim\
282 e set by the Attack control. Once it reaches that level, the output voltage immediat\
283 ely starts dropping to speed set by the Decay control it until it reaches the voltag\
284 e set by the Sustain control. If the input gate is still active, this level is maint\
285 ained until the gate goes back to 0 volts (usually because you released the key on a\
286 controlling keyboard, etc.). At that time, the output voltage then starts dropping \
287 back to 0 volts at the rate set by the Release control."),
288 quiz::Quiz("AES","Audio Engineering Society."),
289 quiz::Quiz("AES3","(sometimes called AES/EBU) A digital audio transfer standard deve\
290 loped by the Audio Engineering Society and the European Broadcasting Union for carry\
291 ing dual-channel digital audio data between devices. AES3 is the protocol behind XLR\
292 cables, as well as RCA and S/PDIF cables."),
293 quiz::Quiz("AFG","The AFG (Audio Frequency Generator) is a very full-featured analog\
294 oscillator released by Livewire Electronics. It has since been discontinued, but re\
295 furnished B-stock units come up for sale every now and then. The expansion modules w\
296 ere, to the best of my knowledge, never released (at least not widely)."),
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297 quiz::Quiz("Aftertouch", "(Also called “Pressure Sensitivity“) some keyboards measure\  
298 how hard you press down on the keys, and convert this to a voltage (or other contro\  
299 l signal such as MIDI, which can then be converted into a control voltage) that you \  
300 can use to add expression to a note, such as adding vibrato or opening the filter wi\  
301 der. Monophonic aftertouch measures one pressure value for the entire keyboard, rega\  
302 rdless of which key(s) you are pressing; polyphonic aftertouch produces a signal for\  
303 each individual key. Important trivia: Touch plate keyboards actually measure the s\  
304 urface area of the skin touching them rather than pressure or force – so you can inc\  
305 rease or decrease the aftertouch amount by rolling between the tip and length of you\  
306 r finger."),  
307 quiz::Quiz("AHDSR", "Attack, Hold, Decay, Sustain, and Release. This is a slightly fa\  
308 ncier ADSR envelope that holds the voltage typically at its maximum value for a spec\  
309 ified time after the attack is done rising and before the decay starts falling."),  
310 quiz::Quiz("Aliasing", "A type of digital signal distortion that occurs in a sampler \  
311 when the incoming signal frequency exceeds the Nyquist frequency for that unit. The \  
312 sampler reproduces it at an incorrect frequency, or an “alias,” causing a distortion\  
313 or artifact in the sound. If you play back a digital audio file where half of the s\  
314 ample rate is an audible pitch, you will also hear a mirror image of the sound’s har\  
315 monic content reproduced started at that half-sample-rate pivot (unless some excelle\  
316 nt filtering has taken place). (See also “Nyquist Frequency.”)."),  
317 quiz::Quiz("Alternating Current (or AC)", "The type of electrical current found in st\  
318 andard electrical outlets and studio signals running through audio lines. In AC, the\  
319 current “alternates” directions, flowing back and forth through the circuit."),  
320 quiz::Quiz("AM", "Amplitude Modulation (AM) is the name given the to the technique of\  
321 varying the amplitude or loudness of one signal known as the carrier (typically an \  
322 audio signal, swinging both above and below 0 volts) with a second signal called the\  
323 modulator. In the typical amplitude modulation (AM) scenario, a low frequency oscil\  
324 lator with a positive voltage (say, between 0v and 5v, or maybe something smaller su\  
325 ch as between 1v and 2v) is fed into the control input of a voltage controlled ampli\  
326 fier to add vibrato to an audio signal passing through it. Technically, this is know\  
327 n as a two-quadrant multiplier or modulator, as any negative swings in the modulatio\  
328 n signal are ignored; when patching tremolo, you may need to make sure an offset vol\  
329 tage is being added to your LFO to make sure the sound doesn’t cut out on the lower \  
330 excursions of the LFO’s waveform."),  
331 quiz::Quiz("Ambience", "In most cases, this refers to the “atmosphere” of a certain p\  
332 lace, like a restaurant. But in recording, it refers to the part of the sound that c\  
333 omes from the surrounding environment rather than directly from the sound source. Fo\  
334 r example, the sound waves coming into your ears from a cello being played are comin\  
335 g directly from the source, but the sound of the same cello coming to you after boun\  
336 cing off the back wall is ambient sound."),  
337 quiz::Quiz("Ambient Field", "The area away from the sound source where the reverberat\  
338 ion is louder than the direct sound."),  
339 quiz::Quiz("Ambient Miking", "This refers to placing a microphone in the ambient fiel\
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340 d of a room to record the ambient reverberations of the sound. The recording engineer\
341 r often does this in addition to direct mic'ing of the instrument(s) to create a blend\
342 d or mix of direct and reverberant sound in the recording."),
343 quiz::Quiz("Amp", "An abbreviation for "Amplifier," "Amplitude" or "Ampere," dependin\
344 g on context."),
345 quiz::Quiz("Ampere", "The unit of measure for electrical current, abbreviated Amp."),
346 quiz::Quiz("Amplifier", "A device that increases the level or amplitude of an electri\
347 cal signal, making the resulting sound louder."),
348 quiz::Quiz("Amplitude Modulation", "Amplitude Modulation (AM) is the name given the t\
349 o the technique of varying the amplitude or loudness of one signal known as the carr\
350 ier (typically an audio signal, swinging both above and below 0 volts) with a second\
351 signal called the modulator. In the typical amplitude modulation (AM) scenario, a l\
352 ow frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe so\
353 mething smaller such as between 1v and 2v) is fed into the control input of a voltag\
354 e controlled amplifier to add vibrato to an audio signal passing through it. Technic\
355 ally, this is known as a two-quadrant multiplier or modulator, as any negative swing\
356 s in the modulation signal are ignored; when patching tremolo, you may need to make \
357 sure an offset voltage is being added to your LFO to make sure the sound doesn't cut\
358 out on the lower excursions of the LFO's waveform."),
359 quiz::Quiz("Amplitude", "The height of a waveform above or below the zero line. In au\
360 dio, this usually translates to the signal strength or the volume of the sound."),
361 quiz::Quiz("Analog Recording", "A recording of the continuous changes of an audio wav\
362 eform. The most common example of analog recording in a recording studio is recordin\
363 g on reel-to-reel magnetic tape."),
364 quiz::Quiz("Analog Shift Register", "An Analog Shift Register (ASR) is a cross betwee\
365 n a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how t\
366 hose work). When initially triggered, it samples the incoming voltage, and presents \
367 that at its first output. On the second trigger, the incoming voltage is sampled aga\
368 in with this new voltage presented at the first output, while the original voltage i\
369 s now moved to a second output. This game of "telephone" is passed along for as ma\
370 ny stages as the ASR has - traditionally three or four."),
371 quiz::Quiz("Analog To Digital Converter (A/D; or ADC)", "A device that translates a c\
372 ontinuously changing signal (analog) into numeric values that approximate those chan\
373 ges (digital). In audio recording, this refers to converting recorded sound from ele\
374 ctrical voltages to computerized data."),
375 quiz::Quiz("Analog", "The term analog implies a signal is continuously variable, comp\
376 ared to digital where a signal has been converted into discrete numbers. In the land\
377 of modular synthesizers, analog refers to a circuit design that has no digital (or \
378 at least, computer-based) components - instead, it does all of its processing using \
379 transistors, diodes, capacitors, and the such rather than CPUs and DSPs."),
380 quiz::Quiz("AND function", "One of the most common Boolean or binary logic functions,\
381 AND says only output a gate on signal if all of the inputs see "high" gate signals \
382 (i.e. input 1 and input 2 etc. all have gate ons). A NAND function has an inverted o\
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383 utput: The output would be low if both inputs were high, but otherwise would be high\
384 ."),
385 quiz::Quiz("AR", "The two-stage Attack/Release envelope raises from 0 volts to its ma\
386 ximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack param\
387 eter, and then stays at that value for as long as the gate signal fed into the envel\
388 ope generator stays high. Then when the gate signal goes back to zero, the envelope'\
389 s output also falls back to zero at a rate set by its Release parameter. (There is a\
390 separate type of envelope known as an AHD – Attack/Hold/Decay – where you specify a\
391 fixed time for the level to stay at its maximum, rather than pay attention to the g\
392 ate signal.)"),
393 quiz::Quiz("Arpeggiator", "Putting on our music theory hat for a second, an arpeggio \
394 is a type of “broken chord” where the notes are played individually rather than all \
395 at once. An arpeggiator – usually built into a keyboard, or a device inserted betwee\
396 n your keyboard and sound module – makes it easier for you to play arpeggios: You ju\
397 st hold down the notes of the chord, and it automatically plays the notes one at a t\
398 ime, over and over again, like a step sequencer you can program on the fly just by h\
399 olding down a chord. Good arpeggiators have options for different patterns (up, down\
400 , back and forth, random, etc.), and even a latch or hold where it will keep doing t\
401 his even after you’ve released the keys."),
402 quiz::Quiz("ASR", "An Analog Shift Register (ASR) is a cross between a Sample & Hold \
403 module and a Bucket Brigade Delay (assuming you already know how those work). When i\
404 nitially triggered, it samples the incoming voltage, and presents that at its first \
405 output. On the second trigger, the incoming voltage is sampled again with this new v\
406 oltage presented at the first output, while the original voltage is now moved to a s\
407 econd output. This game of \"telephone\" is passed along for as many stages as the A\
408 SR has – traditionally three or four."),
409 quiz::Quiz("Attack/Decay/Sustain/Release", "An envelope generator with four stages: A\
410 ttack, Decay, Sustain, and Release. When this envelope generator receives a gate inp\
411 ut, it typically starts at 0 volts (which is the equivalent of silence when connecte\
412 d to a Voltage Controlled Amplifier, or the lowest frequency when connected to a vol\
413 tage controlled filter or oscillator) and raises to the maximum voltage it can output\
414 t (typically 5 to 10 volts depending on system; it can often be set with an output l\
415 evel control) over a time set by the Attack control. Once it reaches that level, the\
416 output voltage immediately starts dropping to speed set by the Decay control it unt\
417 il it reaches the voltage set by the Sustain control. If the input gate is still act\
418 ive, this level is maintained until the gate goes back to 0 volts (usually because y\
419 ou released the key on a controlling keyboard, etc.). At that time, the output volta\
420 ge then starts dropping back to 0 volts at the rate set by the Release control."),
421 quiz::Quiz("Attack/Decay", "Shorthand for a two-stage Attack/Decay envelope. This sim\
422 ple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perh\
423 aps 10 volts) at a speed defined by its Attack parameter, and then immediately falls\
424 back to 0 volts at a rate defined by its Decay parameter. A variation on this is th\
425 e AHD envelope: After finishing the Attack stage, it holds at the maximum level for \
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426 a specified amount of time (in contrast to an AR envelope, which holds at the maximum level for as long as the note on gate is high), and then decays back to zero. I have heard there are some envelopes that are a hybrid of AHD and AR in that they hold the maximum level for either the defined Hold time or the as long as the incoming gate is high;"),
431 quiz::Quiz("Attack/Hold/Decay/Sustain/Release", "This is a slightly fancier ADSR envelope that holds the voltage typically at its maximum value for a specified time after the attack is done rising and before the decay starts falling."),
432 quiz::Quiz("Attack/Release", "The two-stage Attack/Release envelope raises from 0 volts to its maximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack parameter, and then stays at that value for as long as the gate signal fed into the envelope generator stays high. Then when the gate signal goes back to zero, the envelope's output also falls back to zero at a rate set by its Release parameter. (There is a separate type of envelope known as an AHD - Attack/Hold/Decay - where you specify a fixed time for the level to stay at its maximum, rather than pay attention to the gate signal.)"),
442 quiz::Quiz("Attack", "This usually refers to the first stage of an envelope that occurs at the onset of a note, as it rises from 0 volts (silence when controlling an amplifier module) to typically the value of maximum loudness. Percussive and plucked sounds have very fast attacks; slow, languid wind or string instrument phrases may have long attacks."),
447 quiz::Quiz("Attenuation", "The reduction of electrical or acoustic signal strength. In audio, attenuation is measured in decibels (dB) and is typically heard as a reduction in volume. Sound waves traveling through the air naturally attenuate as they travel away from the source of the sound. Engineers also purposefully attenuate signals in the studio through gain controls or pads to prevent overload."),
452 quiz::Quiz("Attenuator", "A control that can reduce the strength of a signal or voltage going through it."),
454 quiz::Quiz("Attenuverter", "A special version of an attenuator that can also invert the polarity of the signal or voltage going through it. Most attenuverters use pass through no signal at their center position; as you turn them clockwise, you turn up the normal version of the signal; as you turn them counterclockwise, they turn up an inverted version of the signal. Some attenuverters are a normal attenuator with a polarity switch added on."),
460 quiz::Quiz("Audio Frequency Generator", "The AFG (Audio Frequency Generator) is a very full-featured analog oscillator released by Livewire Electronics. It has since been discontinued, but refurbished B-stock units come up for sale every now and then. The expansion modules were, to the best of my knowledge, never released (at least not widely)."),
465 quiz::Quiz("Audio", "In its broadest sense, audio is the range of frequencies we humans can hear with our ears. In the technical sense, audio refers to the transmission, recording or reproduction of sound, whether digitally, electrically or acoustically.."),
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469 quiz::Quiz("Automatic Dialogue Replacement (ADR)", "The process of re-recording dialo\
470 gue for film in a controlled environment after the film is shot, for the purpose of \
471 replacing poorly recorded dialogue."),
472 quiz::Quiz("Automatic Gain Control", "A compressor with a long release time, which is\
473 used to keep the volume of the audio at a consistent level."),
474 quiz::Quiz("Automation", "Programming certain changes to occur automatically during r\
475 ecording and/or playback. In the studio, engineers use automation on their consoles \
476 or computers so various parameters will change automatically at different times duri\
477 ng multitrack recording and playback. This pre-programming feature makes it easier t\
478 o create those changes than attempting to perform them all manually in real time."),
479 quiz::Quiz("Auxiliary Equipment", "External signal processing devices that work along\
480 side the mixing console to modify the signal."),
481 quiz::Quiz("Auxiliary Return", "(Abbreviated Aux Return or Return) The input on a con\
482 sole or DAW that returns the effected signal sent through the auxiliary send back in\
483 to the channel mix."),
484 quiz::Quiz("Auxiliary Send", "(Abbreviated Aux Send or Send) A control to adjust the \
485 signal level being sent from the input channel on a console or DAW to auxiliary equi\
486 pment or plug-ins through the auxiliary bus. This is typically used for creating an \
487 effects loop that processes a portion of the signal, then returns it into the mix th\
488 rough the auxiliary return."),
489 quiz::Quiz("Axis", "An imaginary line around which a device operates. For example: in\
490 microphone use, the axis is an imaginary line coming out from the front of the micr\
491 ophone in the direction of motion of the diaphragm, delineating the optimum locati\
492 on for the mic to pick up the sound. Sounds that occur "off-axis" from the microphone \
493 will not be picked up as clearly."),
494 quiz::Quiz("Background Noise", "Refers to either 1) The ambient noise in a room unrel\
495 ated to the instrument(s) or vocal(s) being recorded; or 2) The system noise unrelat\
496 ed to the recorded signal. (All electronics emit a level of noise.)"),
497 quiz::Quiz("Baffles", "Sound absorbing panels that are used to prevent sound waves fr\
498 om entering or leaving a space."),
499 quiz::Quiz("Balance", "1) The relative level of two or more instruments in a mix, or \
500 the relative level of audio signals in the channels of a stereo recording. 2) To eve\
501 n out the relative levels of audio signals in the channels of stereo recording."),
502 quiz::Quiz("Balanced Audio", "This refers to a system where three wires are used to c\
503 arry an audio signal: one is the ground (the 0 volt reference), the second carries t\
504 he audio signal as it varies above and below 0v, and the third carries an inverted c\
505 opy of the audio signal that goes negative while the original is going positive. Bal\
506 anced audio usually implies a reference signal level of +4dB (higher than line level\
507 ; still lower than most modular synths), although microphone signals - much weaker b\
508 y comparison, and therefore more susceptible to outside noise - are almost always ba\
509 lanced as well. Modular synths tend to use unbalanced audio for their internal signa\
510 ls. If you require a balanced output (or input), you need a special module that conv\
511 erts between balanced and unbalanced audio, plus does any necessary level matching.")\
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512 ),
513 quiz::Quiz("Balanced Cable","A cable consisting of three wires (two signal wires and\
514 a ground wire) and two connectors. The two signal wires carry the same signal in op\
515 posite polarities, providing protection against interference and noise in a balanced\
516 system. Examples of balanced cables include tip-ring-sleeve (TRS) stereo cables and\
517 XLR cables."),
518 quiz::Quiz("Balanced Mixer","A circuit or device that generates the sum and differen\
519 ce frequencies of two input signals."),
520 quiz::Quiz("Balanced Modulator","Balanced or ring modulation is a special type of am\
521 plitude modulation, where one bipolar (swinging both above and below 0 volts) signal\
522 - the modulator - is used to vary the amplitude of a second bipolar signal, known a\
523 s the carrier. The modulator's frequency is both added to and subtracted from the ca\
524 rrier's frequency; the resulting harmonics replace the original carrier and modulato\
525 r."),
526 quiz::Quiz("Banana","An alternate type of connector (https://en.wikipedia.org/wiki/B\
527 anana\_connector) used by 4U systems such as Buchla (control voltages) and Serge (bot\
528 h control and audio). These cables have only one wire, so they carry only the signal\
529 , relying on the module panels and chassis of the system to provide the ground refer\
530 ence. Banana connectors have an advantage in that they are usually "stackable" meani\
531 ng you can plug a one jack into the back of another, providing a passive multiple.")\
532 ,
533 quiz::Quiz("Band Pass Filter","A device, circuit or plug-in that allows a narrow ban\
534 d of frequencies to pass through the circuit, rejecting or attenuating frequencies t\
535 hat are either higher or lower than the specified range."),
536 quiz::Quiz("Band Stop Filter","A device, circuit or plug-in that attenuates a narrow\
537 band of frequencies in the signal, allowing frequencies outside the band to pass. T\
538 he exact opposite of a band pass filter."),
539 quiz::Quiz("Band Track","(Sometimes abbreviated "Track") A mixdown of a song minus t\
540 he lead vocal and/or background vocals. In other words, a mixed track containing onl\
541 y the instrumental parts of the song."),
542 quiz::Quiz("Band","1) A range of frequencies, often identified by the center frequen\
543 cy of the range. 2) A group of musicians playing together."),
544 quiz::Quiz("Bandpass Filter","A bandpass filter (BPF) leaves the harmonics around th\
545 e center, corner or cutoff frequency untouched, and attenuates those above and below\
546 the center frequency. The further away you get from the center, the more they are a\
547 ttenuated, based on the number of poles in the filter, with each pole equalling 6 de\
548 cibels of attenuation for each octave you get away from that center."),
549 quiz::Quiz("Bandwidth","In signal processing, bandwidth refers to the usable frequen\
550 cy range of a communication channel, measured by the difference between the device's\
551 highest and lowest usable frequencies."),
552 quiz::Quiz("Bank","1) A collection of sound patches, sequencer data and/or operating\
553 parameters of a synthesizer's generators and modifiers in memory. 2) A group of sou\
554 nd modules as a unit."),
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555 quiz::Quiz("Bar","In music notation, bar is another term for measure a specified per\
556 iod of time containing a certain number of beats, and marked by bar lines on each si\
557 de of the written measure."),
558 quiz::Quiz("Bark Scale","The human auditory (hearing) system can be thought of as co\
559 nsisting of a series of bandpass filters. Interestingly, the spacing of these filter\
560 s do not strictly follow either a linear frequency scale or a logarithmic musical sc\
561 ale. The Bark Scale is an attempt to determine what the center frequency and bandwid\
562 th of those \"hearing filters\" are (known as critical bands)."),
563 quiz::Quiz("Barrier Miking","A microphone placement technique in which a microphone \
564 is placed close to a reflective surface. When done correctly, barrier miking ensures\
565 that both the direct and reflected sounds reach the microphone simultaneously, prev\
566 enting phase cancellation between the two."),
567 quiz::Quiz("Basic Session","The first audio recording session for recording the basi\
568 c tracks that serve as the song's foundation (for example, the drums and bass)."),
569 quiz::Quiz("Bass Reflex","A type of loudspeaker cabinet design in which a port (open\
570 ing) in the speaker cabinet enhances bass frequencies. The principle is that the sou\
571 nd pressure generated by the back of the speaker cone inside the cabinet is routed o\
572 ut the port at the front of the cabinet, mixed with the sound coming from the front \
573 of the woofer. Changing the port size and position will greatly change the character\
574 of the low frequencies."),
575 quiz::Quiz("Bass","The lower range of audio frequencies up to approximately 250 Hz. \
576 A reference value."),
577 quiz::Quiz("BBD","An early design for an echo or delay effect where the input audio \
578 would be sampled as an analog voltage, and held for a brief moment. Then at the next\
579 above-audio sample rate clock pulse, this voltage would get passed to the next samp\
580 le and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade d\
581 elays (BBDs) usually have numbers of stages or buckets that are powers of two (256, \
582 512, 1024, 2048, etc.); the delay length is determined by the number of stages multi\
583 plied by the time interval between samples."),
584 quiz::Quiz("Beaming","A phenomenon found in loudspeakers in which higher frequencies\
585 are projected straight out of the loudspeaker, rather than dispersing along with th\
586 e lower frequencies. When you stand on-axis in front of the speaker, it sounds as th\
587 ough it is only reproducing the high frequencies, rather than the mids or lows. This\
588 phenomenon is alleviated by routing the high frequncies through horns in the loudsp\
589 eaker."),
590 quiz::Quiz("Beat Mapping","The process of adjusting the tempo variations in a record\
591 ed piece of music to fit the set tempo of the project. In a DAW, this is done using \
592 time stretching tools and cuts to synchronize the transients to the appropriate temp\
593 o markers. This technique is often used, for example, to reconcile a drum or bass pe\
594 rformance that was recorded without a click track."),
595 quiz::Quiz("Beat","1) The steady, even pulse in music. 2) The action of two sounds o\
596 r audio signals of slightly different frequency interfering with one another and cau\
597 sing periodic increases and decreases in volume, heard to the ear as "beats.""),
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598 quiz::Quiz("Beating","When two oscillators are tuned to very nearly - but not quite \
599 - the same frequency, the difference between them causes an interference pattern kno\
600 wn as beating. When the difference in frequency is below the audio rate, this can so\
601 und like a tremolo applied to the loudness of the combined sound."),
602 quiz::Quiz("Beatmatching","A technique predominantly used by DJs to synchronize the \
603 tempos of two recorded tracks, generally through the use of time stretching and pitc\
604 h shifting tools, to create a seamless transition from one song into another."),
605 quiz::Quiz("Beats Per Minute (B.P.M.)","BPM (beats per minute) is the most common wa\
606 y of stating tempo: How many beats (typically, quarter notes) should be counted ever\
607 y minute. A tempo of 120 beats per minute means there would be two beats every secon\
608 d (120 beats/minute x 1 minute/60 seconds = 2). The number of steady even pulses in \
609 music occurring in one minute, defining the tempo of the song."),
610 quiz::Quiz("Berlin School","A particular style of electronic music popularized by th\
611 e likes of Tangerine Dream and Klaus Schulze based on analog synthesizers, heavy on \
612 repetitive sequences and floating chords or drones with solos played on top. More re\
613 cent versions of Berlin School music can be heard from Node and Red Shift."),
614 quiz::Quiz("Bi-amplification","A technique in which high and low frequencies in a sp\
615 eaker or speaker system are driven by two separate amplifiers."),
616 quiz::Quiz("Bi-Directional Pattern","A microphone pickup pattern which is most sensi\
617 tive to picking up sounds directly in front and back of the mic, effectively rejecti\
618 ng sounds coming from the sides. Also called a "figure-8 pattern.""),
619 quiz::Quiz("Binary","A cornerstone of digital systems is the binary counting method,\
620 where each digit can have only two different values: 0 or 1; off or on; low or high\
621 . A binary signal can only have one of these two states. Therefore, a gate or trigge\
622 r signal in a modular synth - even if generated by analog circuitry - could be refer\
623 red to as a binary type signal. See the entry for Boolean for things you can do with\
624 binary signals like gates and divided clocks."),
625 quiz::Quiz("Bipolar","A voltage that can range both above and below zero is referred\
626 to as bipolar. Some modulation signals inside a modular synth - such as vibrato (va\
627 rying the pitch of an oscillator both above and below the note it is supposed to be \
628 playing) - are bipolar in nature."),
629 quiz::Quiz("Bit","The smallest unit of digital information representing a single "0"\
630 or "1.""),
631 quiz::Quiz("Bitrate (or Bit Depth)","In digital recording, the number of computer bi\
632 ts used to describe each sample. The greater the bitrate, the greater the dynamic ra\
633 nge of the sampled sound. The quality and resolution of an audio sample are describe\
634 d as a combination of sample rate and bitrate. (See also "Sample Rate.")"),
635 quiz::Quiz("Blending","The mixing of multiple sounds or channels together to form on\
636 e sound, or mixing the left and right signals together."),
637 quiz::Quiz("Blue Noise","Technically, a type of noise whose power density (spectral \
638 loudness) increases 3 dB per octave with increasing frequency. It has a very "hissy"\
639 characteristic, lacking in bass."),
640 quiz::Quiz("Boolean","Boolean logic only can have two states: high or low; 1 or 0; o\
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641 n or off."),
642 quiz::Quiz("Boom Stand","A microphone stand equipped with a telescoping support arm \
643 to hold the microphone."),
644 quiz::Quiz("Boom","A telescoping support arm attached to a microphone stand holding \
645 the microphone."),
646 quiz::Quiz("Boost","To increase gain at specific frequencies with an equalizer."),
647 quiz::Quiz("Bouncing","(also called "Ping-Ponging" or "Ponging") The technique of co\
648 mbining and mixing multiple tracks onto one or two tracks (mono or stereo). This can\
649 be done in real-time or analog by playing the tracks through the console and record\
650 ing them onto separate tracks, or digitally through a digital audio workstation. Bou\
651 ncing was once used frequently by engineers to free up additional tracks for recordi\
652 ng, but in digital workstations where tracks are virtually unlimited, this practice \
653 is basically obsolete. Today, engineers typically bounce tracks for the purpose of c\
654 reating a preliminary or final mix of a song."),
655 quiz::Quiz("Boundary Microphone","An omnidirectional microphone designed to be place\
656 d flush against a flat surface (or boundary), effectively creating a "half-Omni" pic\
657 kup pattern while eliminating the danger of phase issues from reflected sounds. A po\
658 pular type of boundary microphone is Crown Audio's trademark Pressure Zone Microphon\
659 e (PZM)."),
660 quiz::Quiz("BPF","A bandpass filter (BPF) leaves the harmonics around the center, co\
661 rner or cutoff frequency untouched, and attenuates those above and below the center \
662 frequency. The further away you get from the center, the more they are attenuated, b\
663 ased on the number of poles in the filter, with each pole equalling 6 decibels of at\
664 tenuation for each octave you get away from that center."),
665 quiz::Quiz("BPM","BPM (beats per minute) is the most common way of stating tempo: Ho\
666 w many beats (typically, quarter notes) should be counted every minute. A tempo of 1\
667 20 beats per minute means there would be two beats every second (120 beats/minute x \
668 1 minute/60 seconds = 2)."),
669 quiz::Quiz("Breathing","Pumping and Breathing - In studio jargon, an effect created \
670 when a compressor is rapidly compressing and releasing the sound, creating audible c\
671 hanges in the signal level. "Pumping" generally refers to the audible increase of so\
672 und levels after compression has taken place; "breathing" refers to a similar effect\
673 with vocals, raising the signal volume just as the vocalist is inhaling. Pumping an\
674 d breathing is a sign of cheap compression or over-compression, and is usually undes\
675 irable, although some engineers and musicians use it on purpose occasionally to crea\
676 te a particular effect."),
677 quiz::Quiz("Brickwall Filter","A certain type of low-pass filter exhibiting a steep \
678 cutoff slope which resembles a "brick wall." While these filters are often found in \
679 A/D converters to prevent aliasing, their steep cutoff can introduce unwanted side-e\
680 ffects to the audio signal, such as phase shift."),
681 quiz::Quiz("Bridging","A technique of feeding a single input to both channels of an \
682 amplifier, then summing them into one, thereby effectively doubling the amplifier po\
683 wer supplied to the signal."),
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684 quiz::Quiz("Brownian Noise", "Also referred to as brown noise, technically it's a typ\
685 e of noise whose power density (spectral loudness) decreases 6 dB per octave with in\
686 creasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a di\
687 stance. It can also be used a slowly changing random control voltage or modulation s\
688 ignal, instead of as an audio source."),
689 quiz::Quiz("Buchla Bongos", "This is a classic patch where a complex sound source - s\
690 uch as one oscillator frequency modulating another - is sent through a Low Pass Gate\
691 with either just a trigger to "strike" the vactrol inside or otherwise an instant a\
692 ttack/fast decay envelope to create a nice percussive sound. The fact that the low p\
693 ass gate reduces the higher harmonics as its volume dies away helps tame the harmoni\
694 cs coming from the complex source, and give it a decay similar to a struck percussiv\
695 e instrument."),
696 quiz::Quiz("Bucket Brigade Delay", "An early design for an echo or delay effect where\
697 the input audio would be sampled as an analog voltage, and held for a brief moment.\
698 Then at the next above-audio sample rate clock pulse, this voltage would get passed\
699 to the next sample and hold (bucket) in the circuit, while a new level was sampled.\
700 Bucket brigade delays (BBDs) usually have numbers of stages or buckets that are pow\
701 ers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the numbe\
702 r of stages multiplied by the time interval between samples."),
703 quiz::Quiz("Bucking", "A type of phase cancellation in which two identical signals or\
704 frequencies, having the same amplitude but opposite polarity, cancel one another ou\
705 t. Most commonly used in the context of musical instrument frequencies. Example: a "\
706 Humbucker" guitar pickup is designed to remove or "buck" hum frequencies from the si\
707 gnal using this principle."),
708 quiz::Quiz("Buffered Multiple", "Quite often you need to split or copy a signal to se\
709 nd to more than one destination. This is commonly done with a multiple, where you pl\
710 ug one source in, and then plug in additional patch cables to go off to multiple des\
711 tinations. An active or buffered multiple is one that includes a buffer circuit betw\
712 een the input and output, making sure the signal does not lose its strength or integ\
713 rity by being split too many times, and that no funny business happening on one of t\
714 he outputs affects any of the other connections. Some modules have good buffering bu\
715 ilt into their outputs, and can drive multiple modules without issue. But if you try\
716 to use a passive mult to connect to, say, three oscillators, and you realize the tr\
717 acking isn't very good (they quickly go out of tune as you go up and down the scale)\
718 , then you need a buffered mult instead."),
719 quiz::Quiz("Bulk Dump", "Short for System Exclusive Bulk Dump, a method of transmitti\
720 ng data such as the internal parameters between MIDI devices."),
721 quiz::Quiz("Burst Generator", "When you send this module a trigger, it outputs a stre\
722 am or "burst" of triggers in response. You usually have control over the number of t\
723 riggers, the spacing between them, and often the probability that individual trigger\
724 output will be sent or skipped (for random patterns). At its most tame, it can be u\
725 se to create "double pluck" triggers in response to a normal note on; and its most e\
726 xtreme, it is used to trigger a high-energy, chaotic stream of drum hits that may or\
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727     may not be in time with the music."),
728     quiz::Quiz("Bus Board","This simple circuit board takes the output of your modular s\
729 ystem's power supply and creates multiple copies of it, routed to connectors that go\
730   to your individual modules."),
731     quiz::Quiz("Bus","An audio pathway by which one or more signals, usually from differ\
732 ent sources, are routed to a designated place. Because busses are highly connected t\
733 o signal flow, they serve a broad range of purposes in audio applications. 2) A shor\
734 thand term for the signals themselves that are routed through the bus (see also "Sub\
735 group")."),
736     quiz::Quiz("Byte","Information (data) bits in a grouping of eight. One byte = eight \
737 bits."),
738     quiz::Quiz("Cable Assembly","Cable that is ready for installation in specific applic\
739 ations and usually terminated with connectors."),
740     quiz::Quiz("Cable Harness","A grouping of cables or wires used to interconnect elect\
741 ronic systems."),
742     quiz::Quiz("Cable Sheath","Conductive protective cover that is applied to cables."),
743     quiz::Quiz("Cable","A group of one or more insulated conductors, optical fibers, or \
744 a combination of both within an enveloping jacket, typically for transmitting electr\
745 ical signals of different types."),
746     quiz::Quiz("Capacitor","An electronic device made of two plates separated by an insu\
747 lator, designed to store electrostatic energy. The capacitor is a key component in c\
748 ondenser microphones, for example."),
749     quiz::Quiz("Capstan","A mechanical part of a magnetic tape recorder that controls th\
750 e speed of the tape as it passes across the tape heads."),
751     quiz::Quiz("Capsule","Space-travel definitions aside, this is the name given to the \
752 part of a microphone that contains the diaphragm and active element, the mechanical \
753 structure that converts acoustic sound waves into electrical current."),
754     quiz::Quiz("Carbon Microphone","A microphone that uses carbon granules to convert so\
755 und waves to electrical impulses. The carbon element sits between two plates; as sou\
756 nd waves hit the carbon granules, it generates changes in resistance between the pla\
757 tes, affecting the electrical signal."),
758     quiz::Quiz("Cardioid Pattern","A microphone pickup pattern which is most sensitive t\
759 o sound coming from the front, less from the sides, and least from the back of the d\
760 iaphragm. So named because the pickup pattern is in the shape of a heart (cardio.)"\
761   ,
762     quiz::Quiz("Carrier","There are a few different synthesis techniques where one usual\
763 ly audio-rate signal varies another audio signal. For example, in frequency modulati\
764 on, a second signal (called the modulator) varies the frequency (pitch) of the main \
765 signal, called the carrier. More specifics are described in the entries on frequency\
766 modulation and amplitude modulation."),
767     quiz::Quiz("Cascade","To connect or "daisy chain" two mixers so that the stereo mixi\
768 ng busses of the first mixer feed into the stereo busses of the second."),
769     quiz::Quiz("CCW","Counter-clockwise, usually in the context of rotating a control th\
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770 e left (in the opposite direction of how a clock's hands move)."),
771 quiz::Quiz("CD","An abbreviation for Compact Disc, or a small optical disk with digi\
772 tal audio recorded on it."),
773 quiz::Quiz("Cent","When tuning instruments, a semitone is divided into 100 units cal\
774 led cents; there are 1200 cents per octave (100 x 12 semitones). When one oscillator\
775 is detuned compared to another, the difference in their frequencies is sometimes me\
776 asured in cents."),
777 quiz::Quiz("Center Frequency","The frequency of an audio signal that is most affecte\
778 d by an equalizer, either boosting or attenuating the frequency. Drawn graphically, \
779 this is the very top or bottom (the "peak") of the frequency bell-shaped curve."),
780 quiz::Quiz("Channel Path","The complete signal path from the sound source to the mul\
781 titrack recorder (or DAW). For example, an audio signal that travels from the microp\
782 hone to the preamplifier, then into a channel strip on the mixing console, then is s\
783 ent through the outputs into the recorder. This is different from the monitor path, \
784 which feeds a mix of signals into monitor speakers or headphones without affecting t\
785 he recorded signals. (See also "Monitor Path.")"),
786 quiz::Quiz("Channel","1) An audio recording made on a portion of the width of a mult\
787 itrack tape, or isolated within a digital audio workstation, usually for the purpose\
788 of combining with other channels. 2) A single path that an audio signal travels or \
789 can travel through a device from an input to an output."),
790 quiz::Quiz("Chaotic","Believe it or not, chaotic does not mean completely random to \
791 mathematicians. Chaos theory deals with systems that are random within certain bound\
792 aries – such as the path of a wobbling wheel or the frequency of a dripping faucet. \
793 Although they are not out of control, neither are they completely predictable. In sy\
794 nthesis, a chaotic system usually refers to a modulation generator that is similar t\
795 o a low frequency oscillator, but which has unpredictable wobbles or glitches in an \
796 otherwise loosely or occasionally repetitive pattern. It can also refer to bursts of\
797 triggers that do not follow musical divisions."),
798 quiz::Quiz("Chase","The automatic adjusting of the speed of a recorder (or sequencer\
799 ) to keep time with another recorder."),
800 quiz::Quiz("Chord Chart","A shorthand form of musical notation that provides the bas\
801 ic chord changes and essential rhythmic information of a song. Most commonly used by\
802 studio session players, rhythm sections or jazz bands to provide the skeletal struc\
803 ture of the song while allowing players room to create their own parts and improvise\
804 . While lead sheets typically focus on melody line and chord structure, chord charts\
805 display mainly chord changes and rhythm. (See also "Lead Sheet.")"),
806 quiz::Quiz("Chord","Three or more musical pitches sung or played together."),
807 quiz::Quiz("Chorus","1) The part of a song that is repeated with the same music and \
808 lyrics each time, often containing the main point or hook of the song. 2) A musical \
809 singing group with many singers. 3) A delay effect that simulates a vocal chorus by \
810 adding several delays with a mild amount of feedback and a medium amount of depth.")\
811 ,
812 quiz::Quiz("Circuit","1) One complete path of electric current. 2) Similar to defini\
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813 tion 1, but including all audio signal paths and components to accomplish a particul\
814 ar audio function."),
815 quiz::Quiz("Class Compliant","This refers to a device that is \"plug and play\" - it\
816 can be plugged directly into a computer or other host and immediately be recognized\
817 without additional drivers needing to be installed. This comes up in the modular wo\
818 rld with MIDI to CV/Gate interfaces that use USB: If your converter is a USB Host, a\
819 nd you plug a class compliant USB Device such as a controller keyboard or fader pane\
820 l into it, the converter will recognize it."),
821 quiz::Quiz("Click Track","A metronome “click” fed into headphone monitors for the pu\
822 rpose of helping the musicians play in time with the song."),
823 quiz::Quiz("Clip","All active electronic circuits have a limit on how strong of a si\
824 gnal can pass through them. These limits are often associated with the positive and \
825 negative power supply levels. If the signal attempts to go beyond these limits, they\
826 instead get chopped or clipped off at that limit. For example, an input voltage of \
827 +12 volts may get through without alteration, but +13 volts at the input would come \
828 out as 12 volts. This clipping causes distortion in the waveform, usually adding hig\
829 her harmonics (such as a harsh buzz). Different circuits enter clipping in different\
830 ways - some may have a bit of rounding off before they reach that flat threshold; t\
831 his is referred to as soft clipping and is often desirable as it can be less harsh. \
832 Clipping is so named because the resulting graphic waveform looks like the edges of \
833 the waveform have been “clipped.”),
834 quiz::Quiz("Clock Signal","A signal sent by a device within the circuit that generat\
835 es steady pulses or codes to keep other devices in sync with each other. An example \
836 in the music world is sequencing via MIDI. The sequencer sends a clock signal so con\
837 nected devices will play in time."),
838 quiz::Quiz("Clock","Usually refers to the main rhythmic pulse in a system. Often, th\
839 e clock pulse is much faster than anything it might drive, such as a sequencer or LF\
840 O. The most common clock rate is 24 ppqn (pulses per quarter note), as is the case w\
841 ith MIDI clocks and DIN Sync. However, a trigger that drives a sequencer forward one\
842 note at a time may also be called the “clock” in a system. Indeed, there are module\
843 s that create divisions and multiplications of the main clock to generate new clock \
844 signals with a relationship to the main clock."),
845 quiz::Quiz("Clockwise","Clockwise, as in rotating a control the the right - in the s\
846 ame direction as a clock's hands move."),
847 quiz::Quiz("Close Miking","A microphone placement technique that places the mic clos\
848 e to the sound source to pick up the direct sound and reject ambient sound."),
849 quiz::Quiz("Coaxial Cable","(abbreviated “Coax”) A two-conductor cable that consists\
850 of one conductor surrounded by a shield."),
851 quiz::Quiz("Coincident Miking","A stereo miking technique in which two microphones a\
852 re placed with their heads as close to each other as possible. This prevents phase c\
853 ancillation problems in the mix because the distance from the sound to either microp\
854 hone is the same."),
855 quiz::Quiz("Compander","A signal processor serving as a combination compressor and e\
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856 xpander, primarily used for noise reduction purposes in analog systems. The audio s\  
857 ignal is compressed prior to recording, then expanded at the reproduction stage. Com\  
858 panding is the principle behind Dolby noise reduction systems."),  
859 quiz::Quiz("Comparator","An electrical device that compares the level of one voltage\  
860 to a second. That second voltage may be a second input on a comparator synth module\  
861 , or may be set with a knob or internal reference voltage. Most often, a comparator \  
862 outputs a gate signal that goes high when the first signal is higher than the second\  
863 (or vice versa), and which goes low when the first signal is lower than the second.\  
864 At audio rates, it converts an input waveform into a square or pulse wave, with the\  
865 second signal setting when the new waveform goes high or low in voltage."),  
866 quiz::Quiz("Comping","1) In digital audio workstations (DAWs), the process of blendi\  
867 ng portions of multiple recorded takes to create a “compilation” track. (See also “T\  
868 ake,” “Playlist.”) 2) In jazz music performance, an abbreviation for “accompanying.”\  
869 "),  
870 quiz::Quiz("Complex Oscillator","This module typically has a pair of oscillators beh\  
871 ind one panel that is prewired where one oscillator modulates the other’s frequency \  
872 (known as Frequency Modulation or FM synthesis); some also allow you to quickly swit\  
873 ch them so that the first modulates the amplitude of the second, or some other varia\  
874 tion. They may also have waveshapers built in. They are based on a popular module cr\  
875 eated by Buchla, which is a standard of the “West Coast” approach to synthesis."),  
876 quiz::Quiz("Compression Driver","A diaphragm that feeds a sound pressure wave into a\  
877 horn loudspeaker."),  
878 quiz::Quiz("Compression Ratio","The rate by which a compressor attenuates an incomin\  
879 g signal, measured in decibels. For example, a compression ratio of 4:1 means the co\  
880 mpressor will only allow a 1 dB increase in the signal for every 4 dB increase in th\  
881 e signal above the threshold."),  
882 quiz::Quiz("Compression","1) In signal processing, the action performed by a compres\  
883 sor (see also “Compressor”). 2) In acoustics, the increased air pressure caused by t\  
884 he peak of a sound pressure wave, used in the context of “compression and rarefactio\  
885 n” (see also “Rarefaction”)."),  
886 quiz::Quiz("Compressor","A signal processor or plug-in that reduces the dynamic rang\  
887 e of an audio signal by amplifying its quieter sections and attenuating its louder o\  
888 nes."),  
889 quiz::Quiz("Condenser Microphone","A microphone in which sound is converted into ele\  
890 ctrical current through changes in a capacitor. The sound pressure waves move the di\  
891 aaphragm, producing changes in capacitance which are then changed into electrical vol\  
892 tage."),  
893 quiz::Quiz("Contact Microphone","A microphone designed to pick up vibrations from so\  
894 lid objects (as opposed to vibrations in the air). Also known as a “pickup” or “piez\  
895 o,” this microphone is often used as an acoustic guitar pickup to pick up the vibrat\  
896 ions from the soundboard, or by experimental musicians creating “noise music” from a\  
897 variety of objects."),  
898 quiz::Quiz("Control Voltage Processor","CVP is the abbreviation for a module that al\  

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899 lows processing of the voltage going through it – such as amplifying or attenuating \
900 it, offsetting it in a positive or negative direction, introducing slew (slurring of\
901 changes in voltage), and possibly other functions such as deriving a gate signal fr\
902 om an incoming voltage by running it through a comparator. Make Noise’s Maths is per\
903 haps the most well known control voltage processor out there; you will also find som\
904 e modules with CVP specifically in their name. Regardless, it’s good to have one or \
905 more of this type of module in your system to help massage voltages to get them to d\
906 o what you want (or to teach them new tricks)."),
907 quiz::Quiz("Control Voltage","The concept of control voltage (CV) is at the very roo\
908 t of modular synthesizer. The general idea is that analog voltage levels are used co\
909 ntrol functions and parameters of a module. For example, one control voltage may det\
910 ermine the pitch played by an oscillator; a second control voltage may determine how\
911 loud that signal is after it’s passed through a voltage-controlled amplifier. CV is\
912 the most common shorthand to refer to control voltage – for example, when a synthes\
913 izer module says it features “CV over the filter’s resonance,” that means there is a\
914 control voltage input to control the amount of resonance (feedback) – not just the \
915 customary knob on the front panel."),
916 quiz::Quiz("Controller","In the broadest sense, a controller is any device that is u\
917 sed to control another device. Most commonly used in the context of MIDI controllers\
918 , which send out MIDI signals to control other connected MIDI instruments and device\
919 s. Other examples of controllers in the recording studio can include monitor control\
920 lers, DAW controllers and DJ controllers."),
921 quiz::Quiz("Corner Frequency","The cutoff or corner frequency of a filter is the poi\
922 nt at which it starts filtering. For example, if a low-pass filter has a corner freq\
923 uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
924 00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
925 tered” – reduced in loudness – the further above 500Hz you go."),
926 quiz::Quiz("Counter Clockwise","Counter-clockwise, usually in the context of rotatin\
927 g a control knob the left (in the opposite direction of how a clock’s hands move)."),
928 quiz::Quiz("CPU","Abbreviation for Central Processing Unit, the main “brain” chip in\
929 a computer (also known simply as “Processor”)."),
930 quiz::Quiz("Critical Distance","The distance from the sound source at which the dire\
931 ct sound and the reverberant sound are at equal volume. Critical distance varies acc\
932 ording to the space; in a room with absorbent walls, the critical distance will be f\
933 urther from the source, and in a reverberant room, the distance will be closer to th\
934 e source."),
935 quiz::Quiz("Crossfade","An audio editing technique in which one sound is faded out a\
936 s another sound is faded in, to create a seamless transition between the two. Audio \
937 engineers use crossfading, for example, to blend two takes or more “takes” of a reco\
938 rded track into a composite take. Club DJs also use crossfading to transition from o\
939 ne song to the next with no stops."),
940 quiz::Quiz("Crossover Frequency","The frequency at which the crossover stops sending\
941 the signal to one speaker and starts sending it to another."),
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942 quiz::Quiz("Crossover", "An audio filter component that splits an audio signal into t\  
943 wo or more bands or signals, usually to be fed into different components of a loudsp\  
944 eaker system according to frequency range. (Also called a “crossover network.”)”),  
945 quiz::Quiz("Crosstalk", "The unwanted leakage of an audio signal between two audio ch\  
946 annels—for example, overlapping signals between channels on a mixing console, or ove\  
947 rlapping audio between two tracks of audiotape.”),  
948 quiz::Quiz("Cue", "In general terms, a cue is the starting point for a piece of music\  
949 or section of music. Depending on the context, the word “cue” may describe: 1) The \  
950 point at which a musician or vocalist is supposed to start playing or singing; 2) Th\  
951 e audio fed to the musicians through headphones so they can determine when to start \  
952 playing/singing; 3) A specific location point on the music timeline within a DAW or \  
953 on the tape; or 4) To set the tape or disc to a certain starting point in the song (\  
954 “cueing” the tape). A cue can even refer to an entire section of music being used fo\  
955 r video production.”),  
956 quiz::Quiz("Cutoff Frequency", "The cutoff or corner frequency of a filter is the poi\  
957 nt at which is starts filtering. For example, if a low-pass filter has a corner freq\  
958 uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\  
959 00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\  
960 tered” - reduced in loudness - the further above 500Hz you go.”),  
961 quiz::Quiz("Cutoff Slope", "The rate of reduction of the frequencies beyond the passb\  
962 and of a filter. The slope is described as the number of dB the filter reduces the s\  
963 ignal for each octave past the cutoff frequency.”),  
964 quiz::Quiz("CV/Gate", "This is the shorthand to say a synthesizer may be controlled b\  
965 y voltages - usually for pitch - and gate signals to indicate when a note is “on.” A\  
966 n increasing number of controller keyboards are including CV/Gate output in addition\  
967 to the customary MIDI (Musical Instrument Digital Interface), making them much easi\  
968 er to connect to a modular synthesizer, as no additional MIDI to CV interface is req\  
969 uired.”),  
970 quiz::Quiz("CV", "The concept of control voltage (CV) is at the very root of modular \  
971 synthesizer. The general idea is that analog voltage levels are used control functio\  
972 ns and parameters of a module. For example, one control voltage may determine the pi\  
973 tch played by an oscillator; a second control voltage may determine how loud that si\  
974 gnal is after it’s passed through a voltage-controlled amplifier. CV is the most com\  
975 mon shorthand to refer to control voltage - for example, when a synthesizer module s\  
976 ays it features “CV over the filter’s resonance,” that means there is a control volt\  
977 age input to control the amount of resonance (feedback) - not just the customary kno\  
978 b on the front panel.”),  
979 quiz::Quiz("CVP", "CVP is the abbreviation for a module that allows processing of the\  
980 voltage going through it - such as amplifying or attenuating it, offsetting it in a\  
981 positive or negative direction, introducing slew (slurring of changes in voltage), \  
982 and possibly other functions such as deriving a gate signal from an incoming voltage\  
983 by running it through a comparator. Make Noise’s Maths is perhaps the most well kno\  
984 wn control voltage processor out there; you will also find some modules with CVP spe
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985   cifically in their name. Regardless, it's good to have one or more of this type of m\
986   odule in your system to help massage voltages to get them to do what you want (or to\
987   teach them new tricks)."),
988   quiz::Quiz("CW","Clockwise, as in rotating a control the the right - in the same dir\
989   ection as a clock's hands move."),
990   quiz::Quiz("Cycle","One complete expression of a waveform beginning at a certain poi\
991   nt, progressing through the zero line to the wave's highest and lowest points, and r\
992   eturning to the same value as the starting point. One complete vibration or sound wa\
993   ve."),
994   quiz::Quiz("D-Sub Connector","Abbreviation for "D-subminiature connector," a D-sub i\
995   s a multipin connector that is most often used to connect a computer to a VGA monito\
996   r, but also used occasionally in digital audio applications in the recording studio.\
997   "),
998   quiz::Quiz("D/A","Abbreviation for Digital to Analog conversion, which changes digit\
999   al data numbers (digital audio signal) into discrete voltage level. The reverse proc\
1000  ess of A/D. Also known as DAC."),
1001  quiz::Quiz("DADSR","This is a slightly fancier take on the standard ADSR envelope ge\
1002  nerator that introduces an initial timed delay before the initial attack stage (risi\
1003  ng from 0 to a peak level) begins. One patch idea is to route this type of envelope \
1004  to a low pass filter cutoff, so there's initially a muted, filtered sound when the n\
1005  ote starts, and then after a pause it starts to swell into a brighter, fuller sound.\
1006  "),
1007  quiz::Quiz("Daisy Chain","The connection of three or more devices in a series, where\
1008  the audio signal passes through one device to reach a second, and through the secon\
1009  d to reach the third, etc."),
1010  quiz::Quiz("Damping Factor","Describes an amplifier's ability to restrain the pushba\
1011  ck motion (back-EMF) of the loudspeaker cone when the audio signal stops."),
1012  quiz::Quiz("Damping","The reduction of energy in a vibrating system, through frictio\
1013  n. Can refer to the reduced amplitude in an electrical signal, or the stifled vibrat\
1014  ions of a musical instrument (for example, the damper pedal on an acoustic piano.)\
1015  ,
1016  quiz::Quiz("DAW","An abbreviation for Digital Audio Workstation, a device or softwar\
1017  e program designed for recording and mixing audio digitally."),
1018  quiz::Quiz("dB","An abbreviation for decibel, a measurement ratio that compares sign\
1019  al strengths (usually audio levels)."),
1020  quiz::Quiz("DBX","A series of noise reduction systems, named for the company that de\
1021  veloped them. DBX noise reduction has been less commercially successful than the mor\
1022  e widely known Dolby systems, but is still found on occasion in recording studios.")\
1023  ,
1024  quiz::Quiz("DC Coupled","When a module says its inputs are DC Coupled, that means it\
1025  can accept DC voltages (constant or slowly changing voltages) and pass them through\
1026  unaltered. This is important if, for example, you want to use a VCA to control the \
1027  amplitude of an envelope going through it: You would need one that was DC coupled, a\
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1028 s an AC coupled input would try to remove the DC component of the signal (such as it\  
1029 s sustain level) and return it to 0v."),  
1030 quiz::Quiz("DC", "Electrical current that flows in a single direction, as opposed to \  
1031 Alternating Current (AC), which flows in alternating directions. Many electronic dev\  
1032 ices run on DC, which is usually provided by battery power, USB power or an AC adapt\  
1033 er plugged into the wall. In modular terms, DC refers to a voltage that tends to sta\  
1034 y at one steady level for awhile, such as a gate output that switches between 0v whe\  
1035 n a note is off and 5 or 10v when a note is on. It can also refer to a slowly changi\  
1036 ng voltage, such as an envelope."),  
1037 quiz::Quiz("DCO", "A DCO (Digitally Controlled Oscillator) is a hybrid design for an \  
1038 analog oscillator that – instead of using a voltage level to determine the pitch of \  
1039 the oscillator – uses a digital device such as a counter to determine the length of \  
1040 each waveform cycle and therefore the pitch. On the plus side, tuning is very stable\  
1041 , unlike some all-analog designs. On the minus side, there are no imperfections in p\  
1042 itch that cause subtle detuning (and therefore the perception of “fatness”) when usi\  
1043 ng more than oscillator per voice."),  
1044 quiz::Quiz("De-esser", "An audio compressor designed to reduce the volume of sibilant\  
1045 sounds and frequencies, especially those produced by pronouncing the letter “s.”),  
1046 quiz::Quiz("Decay", "In general, decay refers to a voltage or overall level dropping \  
1047 down from some high point, such as the decay stage of an envelope generator. A real-\  
1048 world analogy is that after you initially strike a drum or pluck a string, it decays\  
1049 in volume from its initial loudness eventually all the way to silence. It can also \  
1050 refer to the tail of a reverb or echo effect where the sound dies away over time."),  
1051 quiz::Quiz("Decca Tree", "A stereo microphone placement technique involving three mic\  
1052 rophones (usually omnidirectional) placed in a “T” pattern. Commonly used in miking \  
1053 choirs, orchestras and other large ensembles, but variations of the Decca tree techn\  
1054 ique are also being used today in surround sound situations."),  
1055 quiz::Quiz("Decibel", "(abbreviated “dB”) The ratio measurement of two levels accordi\  
1056 ng to a scale where a certain percentage change comprises one unit. Most often used \  
1057 to describe audio levels."),  
1058 quiz::Quiz("Degaussing", "The process of demagnetizing an object. In the context of a\  
1059 udio, degaussing essentially erases the recording on magnetic tape."),  
1060 quiz::Quiz("Delay/Attack/Decay/Sustain/Release", "This is a slightly fancier take on \  
1061 the standard ADSR envelope generator that introduces an initial timed delay before t\  
1062 he initial attack stage (rising from 0 to a peak level) begins. One patch idea is to\  
1063 route this type of envelope to a low pass filter cutoff, so there’s initially a mut\  
1064 ed, filtered sound when the note starts, and then after a pause it starts to swell i\  
1065 nto a brighter, fuller sound."),  
1066 quiz::Quiz("Delay", "You all know what the word delay means in the normal world; it c\  
1067 an appear in different forms inside a modular synth. For example, it can refer to th\  
1068 e spacing between repeats in an echo; that’s why an echo device is often known as a \  
1069 “delay” effect. It can also refer to a programmable amount of time you delay a signa\  
1070 l, such as a gate, trigger, or initial stage of an envelope so a note would start la
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1071 ter than it was actually played. Also, 1) An process by which an audio signal is rec\
1072 orded to a medium or device, reproduced at a time delay, then mixed with the origina\
1073 l, non-delayed signal to create a variety of effects such as a fuller sound, echo, c\
1074 horusing, flanging, etc. 2) A signal processor that creates delay effects."),
1075 quiz::Quiz("Demo","A preliminary recording that is intended to give the listener an \
1076 idea of how a song could sound in a final production. A demo usually involves minima\
1077 l tracking or production, almost like a “rough draft” of a recording."),
1078 quiz::Quiz("Detune","If you have two oscillators tuned to exactly the same frequency\
1079 - and I mean, exactly the same frequency - there’s not much point in having more th\
1080 an one oscillator. However, when you change the tuning of one ever so slightly - in \
1081 other words, detune it - you will start to hear interesting interactions between the\
1082 two, often referred to as chorusing or beating. The result tends to be more interes\
1083 ting and “full” - and a bit more natural, as two singers or instruments can rarely h\
1084 it exactly the same note. To purposely cause an instrument or signal to play out of \
1085 tune (usually slightly). This effect can be used for a number of purposes in the stu\
1086 dio, but is often used in “double-tracking,” blending the detuned instrument/track w\
1087 ith the original to create a fuller sound."),
1088 quiz::Quiz("DI","The process of sending an electrical audio signal directly from an \
1089 instrument to the mixing console through the use of electric pickups or direct boxes\
1090 , as opposed to using a microphone."),
1091 quiz::Quiz("Dialogue","The spoken word recorded in film/video sound, commercials and\
1092 instructional recordings."),
1093 quiz::Quiz("Diaphragm","The part of a microphone that moves in response to sound wav\
1094 es, converting them to electrical signals."),
1095 quiz::Quiz("Difference","A fancy way of saying you subtracted on control voltage fro\
1096 m another. It can also be applied to audio or harmonics."),
1097 quiz::Quiz("Digital Audio Workstation","abbreviated DAW) A device or computer softwa\
1098 re that records and mixes audio digitally and creates digital audio files. A DAW can\
1099 be a standalone unit or an integrated set of components, but today they are most co\
1100 mmonly found as “in-the-box” software programs run from a computer. The most common \
1101 DAW program found in recording studios is Pro Tools; other commonly used programs in\
1102 clude Reason, Ableton and Logic."),
1103 quiz::Quiz("Digital Multimeter","A small device that tests electrical voltage, curre\
1104 nt, and resistance. Multimeters are useful in recording studios for calibrating elec\
1105 trical systems and troubleshooting problems."),
1106 quiz::Quiz("Digital Recording","The process of converting audio signals into numbers\
1107 that represent the waveform, then storing these numbers as data."),
1108 quiz::Quiz("Digital Signal Processing","(abbreviated “DSP”) Any signal processing do\
1109 ne after an analog audio signal has been converted into digital audio."),
1110 quiz::Quiz("Digital to Analog Converter","(abbreviated D/A) A device that converts t\
1111 he digital data of digital audio into voltage levels that approximate the original a\
1112 nalog audio."),
1113 quiz::Quiz("Digital","There was a time when digital (referring to circuitry based ar\

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1114 ound binary logic, computers, and the such compared to the old-fashioned transistors\  
1115 , op amps, capacitors, and other bits that make up analog circuitry) was a dirty wor\  
1116 d among synthesists. The assumption was digital techniques created sounds that were \  
1117 more sterile, brittle, and abrasive – and just not as “authentic.” Today, digital ci\  
1118 rcuitry is embraced in synthesizers, including modular systems. Although analog will\  
1119 always hold a special place in our hearts, a well-implemented digital circuit can s\  
1120 ound just as good as an analog one, while digital signal processing and programming \  
1121 can create a wider range of sounds than most analog circuitry.”),  
1122 quiz::Quiz("Digitally Controlled Oscillator","A DCO (Digitally Controlled Oscillator\  
1123 ) is a hybrid design for an analog oscillator that – instead of using a voltage leve\  
1124 l to determine the pitch of the oscillator – uses a digital device such as a counter\  
1125 to determine the length of each waveform cycle and therefore the pitch. On the plus\  
1126 side, tuning is very stable, unlike some all-analog designs. On the minus side, the\  
1127 re are no imperfections in pitch that cause subtle detuning (and therefore the perce\  
1128 ption of “fatness”) when using more than oscillator per voice.”),  
1129 quiz::Quiz("DIN Stereo","A stereo microphone placement technique that places two car\  
1130 dioid microphones about 20cm apart and set outward from each other at a 90-degree an\  
1131 gle to create a stereo image. Particularly for stereo miking at close ranges. (See \  
1132 also “Near-Coincident Miking.””)),  
1133 quiz::Quiz("DIN Sync","A clock signal for controlling the tempo of sequencers, arpeg\  
1134 giators, and drum machines, distributed using cables with DIN-style connectors (yes,\  
1135 just like old-fashioned MIDI connectors, but DIN Sync is even older). Roland pionee\  
1136 red this standard, which included sending 24 pulses per quarter note (PPQN), giving \  
1137 rise to the alternate name Sync24. Korg equipment used a variation of this running a\  
1138 t 48 pulses per quarter note, also known as Sync48. DIN Sync is still a popular way \  
1139 of sending a clock signal to a modular synth today, especially when interfacing with\  
1140 other vintage synthesizers, sequencers, and drum machines.”),  
1141 quiz::Quiz("Diode Ladder Filter","This is a filter design most often associated with\  
1142 the Roland TB-303 Bass Line, which is known for its rubbery sound with eager resona\  
1143 nce.”),  
1144 quiz::Quiz("Diode","An electrical component that enables easy electrical current flo\  
1145 w in one direction but not the other. In the recording studio, these are commonly fo\  
1146 und in the vacuum tubes of tube amplifiers.”),  
1147 quiz::Quiz("Direct Box","A small device that to converts an unbalanced, high-impedan\  
1148 ce speaker or instrument-level output to a balanced, low-impedance mic-level output.\  
1149 Frequently used in the signal path connecting electric instruments “directly” to th\  
1150 e mixing console, as opposed to miking them acoustically. Also called “direct inject\  
1151 ion box” or “DI box.””)),  
1152 quiz::Quiz("Direct Current","In modular terms, DC refers to a voltage that tends to \  
1153 stay at one steady level for awhile, such as a gate output that switches between 0v \  
1154 when a note is off and 5 or 10v when a note is on. It can also refer to a slowly cha\  
1155 nging voltage, such as an envelope. (abbreviated “DC”) Electrical current that flows\  
1156 in a single direction, as opposed to Alternating Current (AC), which flows in alter\  

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1157 nating directions. Many electronic devices run on DC, which is usually provided by b\  
1158 attery power, USB power or an AC adapter plugged into the wall."),  
1159 quiz::Quiz("Direct Injection","(abbreviated “DI”) The process of sending an electric\  
1160 al audio signal directly from an instrument to the mixing console through the use of\  
1161 electric pickups or direct boxes, as opposed to using a microphone."),  
1162 quiz::Quiz("Direct Out","An output available on some consoles which is fed directly \  
1163 from the preamplifier stage of the input, bypassing the channel strips and faders. T\  
1164 his feature is often used to send a “dry” signal to a monitor mix or a recording dev\  
1165 ice."),  
1166 quiz::Quiz("Direct Sound","The sound that reaches a microphone or a listener’s ear w\  
1167 ithout hitting or bouncing off any obstacles (as opposed to reflected or ambient sou\  
1168 nd)."),  
1169 quiz::Quiz("Directional Pattern","1) In microphones, a term meaning the same thing a\  
1170 s “Pick Up Pattern,” a description of the area in which a microphone is most sensiti\  
1171 ve to sounds. 2) In loudspeakers, it is the pattern of dispersion, the area that the\  
1172 sound from a speaker will evenly cover in a listening area."),  
1173 quiz::Quiz("Dispersion (also Dispersion Angle)","The area that is effectively covere\  
1174 d by the sound coming from a loudspeaker; specifically, the imaginary boundaries on \  
1175 either side of the speaker at which the sound level is 6 dB lower than if you were s\  
1176 tanding directly in front of the speaker. Each speaker has both a horizontal and ver\  
1177 tical dispersion angle."),  
1178 quiz::Quiz("Distant Miking","The technique of placing a microphone far from the soun\  
1179 d source in order to pick up a combination of the direct and reflected sounds."),  
1180 quiz::Quiz("Distortion","Refers to the deforming of a waveform at the output of a de\  
1181 vice as compared with the input, usually due to overload, creating a distorted or “d\  
1182 irtly” signal. While electrical or audio distortion is typically unwanted and avoided\  
1183 , it is frequently used in controlled situations in audio to create certain desirabl\  
1184 e effects, particularly with electric guitars and amplifiers."),  
1185 quiz::Quiz("Diversity","1) In audio settings: the use of two or more antennas in a w\  
1186 ireless receiver system to prevent dropouts in the audio from a wireless microphone.\  
1187 2) In other settings: the embracing of the uniqueness of all individuals."),  
1188 quiz::Quiz("Dolby","The brand name of a manufacturer of noise reduction systems and \  
1189 other audio systems, to improve performance and fidelity of audio recording, playbac\  
1190 k, and transmission."),  
1191 quiz::Quiz("Doppler Effect","The phenomenon in which the human ear perceives a chang\  
1192 e in the frequency (pitch) of a sound while the sound source is in motion. As the so\  
1193 und source approaches, the sound waves travel a shorter distance to the ear, increas\  
1194 ing the frequency of the waves and the pitch of the sound; as the sound source moves\  
1195 away, the sound waves must travel farther and farther, resulting in lower frequenci\  
1196 es. A common example of this effect is an approaching emergency vehicle whose siren \  
1197 sounds higher as it approaches and lower after it passes. The Doppler Effect can be \  
1198 utilized in audio settings, for example, in the Leslie speaker in which an electric\  
1199 motor rotates the speakers inside the cabinet, constantly changing the distance bet\  

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1200 ween the sound source and the listener (or microphone) and creating its signature wa\
1201 rbling vibrato effect."),
1202 quiz::Quiz("Double","1) To record a second performance closely matching the first pe\
1203 rformance, for the purpose of blending the two tracks. 2) To use a delay line with m\
1204 edium delay to simulate double tracking."),
1205 quiz::Quiz("Driver","1) A transducer in a loudspeaker that converts electrical signa\
1206 ls into sound pressure waves. 2) A computer program that controls an attached device\
1207 or piece of hardware."),
1208 quiz::Quiz("Dropout","A brief loss of audio signal on tape, or a brief loss of data \
1209 in a digital audio file (often due to a dropped sample), that can result in an unwan\
1210 ted dip in audio, a crackle or a pop."),
1211 quiz::Quiz("Drum Machine","An electronic device containing synthesized and/or sample\
1212 d drum sounds in its memory, along with an internal sequencer that can be programmed\
1213 to play drum patterns or loops."),
1214 quiz::Quiz("Drum Pattern","A specific sequence of drum sounds played by a drummer or\
1215 sequenced into a drum machine for use in a song."),
1216 quiz::Quiz("Dry","A sound with no effects is referred to as \"dry\"; a sound with ef\
1217 fects (such as reverb) mixed is referred to as \"wet.\" Effects units or mixers ofte\
1218 n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
1219 d and the fully-effected sound."),
1220 quiz::Quiz("DSP","Any signal processing done after an analog audio signal has been c\
1221 onverted into digital audio."),
1222 quiz::Quiz("Dub (or Dubbing)","1) To copy a recording. 2) To record in real time wit\
1223 h another recording with the intent of mixing the two recordings (see also “Overdub/\
1224 Overdubbing”). 3) “Dub” is an abbreviation for “dubstep,” a style or subgenre of ele\
1225 ctronic music."),
1226 quiz::Quiz("Ducking","A compression-based audio effect in which an audio signal is r\
1227 educed proportionately by the presence of another audio signal, sometimes accomplish\
1228 ed through a “sidechain” connection with the signal processor. A notable example is \
1229 a spoken-word voice-over track recorded over a musical track, where the music drops \
1230 in volume when the speaker begins to speak. A more subtle example is when an audio e\
1231 ngineer “ducks” specific sounds to make room for others in the track; for example, w\
1232 hen a bass guitar signal triggers a slight reduction in the level of drums or guitar\
1233 s. (See also “Sidechain.”)"),
1234 quiz::Quiz("Duophonic","Duophonic means two \"voices.\" Most early synths (including\
1235 modular systems) are monophonic, which means they can play only one note at a time;\
1236 some instruments have enough oscillators, filters, envelopes, and amplifiers that t\
1237 hey could play two separate notes as once. Some MIDI interfaces for modular synths i\
1238 nclude duophonic modes so you can patch up and control two separate voices from your\
1239 keyboard. Some users play fast and loose with terms such as duophonic, monophonic, \
1240 and polyphonic;"),
1241 quiz::Quiz("Duration","Duration is another way of saying length. A clock pulse or a \
1242 gate signal that is “high” for a certain amount of time – say, 100 msec – is said to\
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1243     have a duration of 100 msec. The length of time you hold a note down, or the length\  
1244     of a step in a sequence, is also called its duration."),  
1245     quiz::Quiz("Dynamic Microphone","(Also called Moving Coil Microphone) A microphone i\  
1246     n which sound pressure waves are converted to an electrical audio signal by an induc\  
1247     tion coil moving within a magnetic field—a process often compared to a loudspeaker w\  
1248     orking in reverse. Dynamic microphones are less sensitive than condenser microphones\  
1249     , but can be effective for miking louder sound sources or for close-miking applicati\  
1250     ons."),  
1251     quiz::Quiz("Dynamic Processing/Dynamic Signal Processing","The process of automatica\  
1252     lly changing the level (or gain) to alter the level relationship of the loudest audi\  
1253     o to the softest audio. Dynamic processors include compressors, limiters, expanders \  
1254     and gates."),  
1255     quiz::Quiz("Dynamic Range","1) The ratio (in dB) between the loudest peak and the so\  
1256     ftest level of a song or recording. 2) The ratio (in dB) between the softest and lou\  
1257     dest possible levels a device or system can provide without distortion."),  
1258     quiz::Quiz("Early Reflections","The first sound waves that reach a listener's ear af\  
1259     ter bouncing off a surface in the room, usually heard almost immediately after the i\  
1260     nitial sound. The first stage of reverberation."),  
1261     quiz::Quiz("East Coast Synthesis","This blanket term is applied to most common synth\  
1262     esizer configuration pioneered by East Coast based companies such as Moog, Arp, and \  
1263     EML (as well as "Far East" companies such as Roland and Korg) where one or more osci\  
1264     llators producing waveforms with rich harmonic content (such as a sawtooth or square\  
1265     wave) are fed into a filter that removes some of those harmonics, and then onto an \  
1266     amplifier to shape the loudness of a note. This approach is also often known as subt\  
1267     ractive synthesis, as the filter reduces (subtracts) harmonics that came from the os\  
1268     cillators. East Coast synthesizers also regularly have organ-style black & white key\  
1269     boards, and four stage ADSR type envelopes. Today it's common to mix both East Coast\  
1270     and West Coast approaches in the same system."),  
1271     quiz::Quiz("Echo Chamber","An enclosed room designed with reflective, non-parallel s\  
1272     urfaces for the purpose of creating acoustic echoes (reverberation)."),  
1273     quiz::Quiz("Echo","The distinct repetition of an initial sound, caused by the reflec\  
1274     tion of the sound waves upon a surface. We recognize a sound as an echo when the dis\  
1275     tance between the source and the reflection is far enough apart that we can detect t\  
1276     he time delay between one and the other. Essentially, reverberation is the combinati\  
1277     on of many echoes occurring too rapidly to hear each individually. In the studio, ec\  
1278     hoes can be reproduced acoustically or simulated by a digital signal processor."),  
1279     quiz::Quiz("Edit","To change one or more parameters of a recorded sound after the fa\  
1280     ct. This can take many forms, including "punching in" a section of the music that is\  
1281     re-recorded to replace the original version; altering the shape/size of waveforms g\  
1282     raphically; changing the sequence of playback; and many others. Analog editing would\  
1283     typically involve splicing the magnetic tape on which the audio signals were record\  
1284     ed. These days, almost all editing in the studio is done via computer using a digita\  
1285     l audio workstation (DAW)."),
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1286 quiz::Quiz("Effect Loop","Sometimes you might want to send a signal outside your mod\
1287 ular system, process it through an external effects device, and bring it back into y\
1288 our modular for more processing. This going out/coming back in is referred to as an \
1289 effect loop. The trick with modular synths is that their internal signal levels tend\
1290 to be much higher than those used by external effect equipment, so a modular effect\
1291 loop will usually have level matching circuitry as well."),
1292 quiz::Quiz("Effects Processor","(Also called Guitar Processor) A device that adds au\
1293 dio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, o\
1294 verdrive, amplifier simulation, etc. Effects processors can occur as individual effe\
1295 cts boxes or multi-sound pedal boards (see also "Foot Pedals," "Foot Switches") adde\
1296 d into the signal path between the guitar and the console. They can also be found as\
1297 presets in guitar amplifiers, or even as digital plug-ins within a DAW."),
1298 quiz::Quiz("Effects Track","1) In film production audio, a recording of the mixdown \
1299 of all the sound effects ready to be mixed with the dialogue and music. 2) In music \
1300 recording, one track with a recording of effects to be added to another track of a m\
1301 ultitrack recording."),
1302 quiz::Quiz("Effects","1) Various ways an audio signal can be modified by adding some\
1303 thing to the signal to change the sound. 2) Short for the term Sound Effects (sounds\
1304 other than dialogue, narration or music like door closings, wind, etc.) added to fi\
1305 lm or video."),
1306 quiz::Quiz("EG","The envelope generator (EG) module is used to shape the loudness or\
1307 dynamics of a note when connected to a VCA (Voltage Controlled Amplifier), as well \
1308 as how its frequency content or timbre changes over time when connected to a VCF (Vo\
1309 ltage Controlled Filter). To do this, and envelope generator creates a voltage that \
1310 typically rises from zero volts to some maximum level, and back down again. You cont\
1311 rol how long this takes, usually in various stages: an attack stage as it goes from \
1312 zero to max, a decay stage as it falls back down from maximum to either zero (in the\
1313 case of an AD, or Attack/Decay envelope) or an intermediate level known as the sust\
1314 ain, and then (usually after a key has been released and the corresponding gate sign\
1315 al has gone back to zero) from the sustain level back to zero over a duration known \
1316 as its release."),
1317 quiz::Quiz("Electret Microphone","A variation of condenser microphone that uses an e\
1318 lectret instead of a capacitor. (Also called "Electret Condenser Microphone.") Be cau\
1319 se the electret is permanently polarized, an electret microphone does not require an\
1320 external power source as a standard condenser microphone does."),
1321 quiz::Quiz("Electret","A dielectric plate that is designed with permanent polarity, \
1322 allowing it to function similarly to a magnet. ("Electret" comes from the words "ele\
1323 ctricity" and "magnet.") Used in some microphone types in place of a capacitor (cond\
1324 enser)."),
1325 quiz::Quiz("Electromagnetic Field","(Abbreviated EMF) A field of magnetic energy put\
1326 out because of current traveling through a conductor."),
1327 quiz::Quiz("Electromagnetic Interference (EMI)","The bane of audio professionals eve\
1328 rywhere, EMI is a type of interference caused by nearby electromagnetic activity, wh\
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1329 ich can be picked up by audio cables and equipment, causing unwanted noise, hum or b\  
1330 uzz in audio systems. Common causes of EMI in audio systems may include high-current\  
1331 power lines, fluorescent lighting, dimmer switches, computers, video monitors and r\  
1332 adio transmitters."),  
1333 quiz::Quiz("Electrons","Negatively charged particles revolving around the nucleus of\  
1334 an atom. Electrical current is generated by electrons moving along a conductor, lik\  
1335 e a metallic wire."),  
1336 quiz::Quiz("Emphasis","This word can have two meanings. In a normal audio context, i\  
1337 t usually means some form of high frequency boost, as emphasizing the higher harmoni\  
1338 cs can add clarity to a tone and help distinguish it from another. In synthesizers, \  
1339 emphasis usually means the Q or resonance setting on a filter, as increasing this se\  
1340 tting boosts (emphasizes) the harmonics at the cutoff or corner frequency."),  
1341 quiz::Quiz("Envelope Follower","This module follows the loudness contour of a sound,\  
1342 and outputs a voltage that corresponds to how that loudness changes. They tend to p\  
1343 erform some smoothing on this signal so that it's not too nervous or jumpy in nature\  
1344 . Envelope followers often also have a gate output that goes high when the loudness \  
1345 of the input signal went over a certain level, and low when it falls back below that\  
1346 level."),  
1347 quiz::Quiz("Envelope Generator","The envelope generator (EG) module is used to shape\  
1348 the loudness or dynamics of a note when connected to a VCA (Voltage Controlled Ampl\  
1349 ifier), as well as how its frequency content or timbre changes over time when connec\  
1350 ted to a VCF (Voltage Controlled Filter). To do this, and envelope generator creates\  
1351 a voltage that typically rises from zero volts to some maximum level, and back down\  
1352 again. You control how long this takes, usually in various stages: an attack stage \  
1353 as it goes from zero to max, a decay stage as if falls back down from maximum to eit\  
1354 her zero (in the case of an AD, or Attack/Decay envelope) or an intermediate level k\  
1355 nown as the sustain, and then (usually after a key has been released and the corresp\  
1356 onding gate signal has gone back to zero) from the sustain level back to zero over a\  
1357 duration known as its release."),  
1358 quiz::Quiz("Envelope Tracking","This describes the main action of an envelope follow\  
1359 er: a module or section of a module that follows the loudness of a signal and output\  
1360 s a voltage that corresponds to - tracks - that input."),  
1361 quiz::Quiz("Envelope","The collective term for the four elements of the lifespan of \  
1362 a sound: Attack, Decay, Sustain and Release (ASDR). The envelope of a sound describe\  
1363 s how a sound or audio signal varies in intensity over a period of time."),  
1364 quiz::Quiz("Equal Loudness Contours","A drawing of several curves showing how loud t\  
1365 he tones of different frequencies would have to be played for a person to say they w\  
1366 ere of equal loudness. (See also "Fletcher-Munson Curves.")),  
1367 quiz::Quiz("Equalizer","An audio signal processor that uses one or more filters to b\  
1368 oost or cut the amplitude (volume) of certain frequencies within the sound. The unde\  
1369 rlying principle is to balance or "equalize" the frequency response of the audio sys\  
1370 tem, or to create balance between multiple signals in a sonic space. However, audio \  
1371 engineers may use equalizers to alter or "color" the sound in many different ways.)\  

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1372 ,
1373 quiz::Quiz("Eurorack","Eurorack is arguably the most popular format of modular synth\
1374 esizer today, with over 100 manufacturers and over 1000 modules available. It was cr\
1375 eated by Doepfer Musikelektronik in 1995, basing its size off the Eurorack format fo\
1376 r lab equipment. Some users will try to tell you that Eurorack doesn't "sound" as go\
1377 od as other formats, but that's just based on a few substandard manufacturers or mod\
1378 ules; there's nothing inherent to the standard that makes a huge difference in the f\
1379 inal sound (no; the difference between 12 and 15 volt power supplies is not enough t\
1380 o most ears)."),
1381 quiz::Quiz("Expander","A signal processor (or plug-in) that performs the opposite fu\
1382 nction of a compressor, expanding the dynamic range of an audio signal rather than c\
1383 ompressing it. It accomplishes this by further reducing the amplitude of signals tha\
1384 t drop below a set threshold."),
1385 quiz::Quiz("Expansion Ratio","The rate by which an expander attenuates an incoming s\
1386 ignal, measured in decibels. For example, an expansion ratio of 2:1 means the expand\
1387 er will reduce the signal by 2dB for every 1dB it drops below the threshold. If the \
1388 signal falls 3dB below the threshold, the expander attenuates it by 6 dB, and so on.\
1389 "),
1390 quiz::Quiz("Exponential","In general terms, this is a mathematical curve that starts\
1391 out relatively flat and then bends to climb steeply. In synthesizer terms, it most \
1392 often refers to the control voltage scheme where a change of 1 volt corresponds to a\
1393 n increased pitch of one octave, which is doubling in cycles (vibrations) per second\
1394 . This is in contrast to a linear system where 1 volt increase would always result i\
1395 n the same increase of cycles per second."),
1396 quiz::Quiz("Fade","A gradual reduction of the level of the audio signal, or a gradua\
1397 l change of level from one pre-set level to another."),
1398 quiz::Quiz("Fader","A control which adjusts the level (gain or attenuation) of an in\
1399 coming signal to a channel or grouping of channels on a console."),
1400 quiz::Quiz("Far Field","The region away from a loudspeaker at which the sound drops \
1401 6dB for each doubling of the distance, up to the critical distance. The beginning of\
1402 the far field varies according to the size of the speaker, but in most cases the fa\
1403 r field begins around 3 feet from the sound source. Audio engineers often use both n\
1404 ear field and far field monitoring when fine-tuning a mix. (See also "Critical Dista\
1405 nce," "Near Field.")"),
1406 quiz::Quiz("Feed","To send an audio or control signal to."),
1407 quiz::Quiz("Feedback Control","The control on a delay line or delay effects device t\
1408 hat controls the amount of feedback into the system."),
1409 quiz::Quiz("Feedback","The return of a portion of the output signal back into the in\
1410 put of a system. This can be done in a controlled manner through a feedback circuit \
1411 to alter the sound of an instrument (most commonly electric guitars or analog synths\
1412 ). It can also describe the unwanted feedback loop created when an open microphone i\
1413 s picking up the sound from a nearby speaker, generating a loud, oscillating frequen\
1414 cy that increases in intensity until the feedback loop is broken by turning off the \
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1415 mic or speaker, or by use of an equalizer to attenuate the frequency."),
1416 quiz::Quiz("Fidelity","A term describing how accurately a sound is reproduced from i\
1417 ts original source."),
1418 quiz::Quiz("Figure-8 Pattern","A microphone pickup pattern which is most sensitive t\
1419 o picking up sounds directly in front and back of the mic, effectively rejecting sou\
1420 nds coming from the sides."),
1421 quiz::Quiz("Filter","A module that reduced or removes certain frequencies and harmon\
1422 ics from the sound that is passed through it. In a synthesizer, the most typical fil\
1423 ter types are low pass (passes all of the harmonics below its cutoff or corner frequ\
1424 ency untouched, and then reduces the level of higher harmonics the further you go ab\
1425 ove that cutoff frequency), high pass (passes all harmonics above its cutoff frequen\
1426 cy untouched, and reduces the level of progressively lower harmonics below the cutoff\
1427 f), bandpass (harmonics right around the cutoff are passed intact, and then reduced \
1428 more in level the further away they are above or below the cutoff frequency), and no\
1429 tch (harmonics right around the cutoff frequency are reduced or cut out entirely; ot\
1430 hers above or below are allowed to live)."),
1431 quiz::Quiz("Flanger","A signal processor often identified as the one that creates a \
1432 “jet taking off” whoosh. What’s going on behind the panel is that a copy of the inpu\
1433 t signal is delayed by a very small amount (longer than a chorus effect; shorter tha\
1434 n an echo effect) and mixed in with the original. When the delay is constant, the re\
1435 sult is a “comb filter” where certain harmonics are cancelled out as they are mixed \
1436 back on top of themselves out of phase. When the delay is varied over time, you get \
1437 swooshes and sweeps. The effect was originally created by playing two tape reels of \
1438 the same song, starting them in time with each other, and dragging your finger on th\
1439 e flange of one of the tape reels to delay it."),
1440 quiz::Quiz("Flanging","An audio effect caused by blending the signal with a copy of \
1441 that signal at a slight time delay, then modifying the delayed copy, creating a “swi\
1442 rling” sound. This was originally accomplished in analog tape recording by playing t\
1443 he original tape and the copy on two tape machines simultaneously, then physically p\
1444 ressing on the flange of one of the machines to alter the timing of the duplicate tr\
1445 ack. These days, most flanging is done through delay boxes or digital plug-ins."),
1446 quiz::Quiz("Flat","1) A term used to describe an even frequency response in a device\
1447 or speaker, meaning that the device/speaker treats all frequencies the same without\
1448 the need for EQ. When displayed graphically, the frequency response is shown as a “\
1449 flat” line with no peaks or valleys. 2) In music, describes a note or pitch that is \
1450 out of tune, sounding at a slightly lower frequency than it should. 3) In music nota\
1451 tion, an “accidental” mark that instructs the player to play/sing the note one-half \
1452 step lower."),
1453 quiz::Quiz("Fletcher-Munson Curves","Also known as “Equal Loudness Contours,” a set \
1454 of graphical curves plotted to illustrate how the human ear responds to different fr\
1455 equencies at different volume levels. Named after the two researchers who first plot\
1456 ted the curves. (See also “Equal Loudness Contours.”)"),
1457 quiz::Quiz("Flip-Flop","In binary logic terms, a flip-flop toggles between high and \
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1458 low every time it receives an input trigger (i.e. the first trigger would set the ou\
1459 tput high, the second trigger sets it low again, and so on). In clock or audio terms\
1460 , it divides the speed of an input clock or square wave by 2."),
1461 quiz::Quiz("Floating Unbalanced Line","A connection “workaround” in which an unbalan\
1462 ced output is connected to a balanced input by modifying the connections in the line\
1463 to resemble a balanced line, alleviating unwanted hum or buzz."),
1464 quiz::Quiz("Fly In","To add sounds into a mix or recording that have no synchronizat\
1465 ion."),
1466 quiz::Quiz("Flying Bus","This is a very simple type of power distribution or bus boa\
1467 rd that typically uses a ribbon cable with multiple connectors along its length to t\
1468 ake the output of your power supply and distribute it to your individual modules. Th\
1469 ey're cheap and easy to install and use, but in a few cases might be a cause of nois\
1470 e being shared between modules."),
1471 quiz::Quiz("FM","Frequency modulation (FM for short) refers to a synthesis technique\
1472 where the pitch of an oscillator is varied (modulated) very quickly – at audio rate\
1473 s – by another oscillator. The result is a complex side of harmonics that may either\
1474 be nicely in tune or clangorous and “out of tune” with the fundamental pitch of the\
1475 main oscillator."),
1476 quiz::Quiz("FOH","In live audio settings, the location in a venue opposite the stage\
1477 , where live audio for the show is controlled and mixed."),
1478 quiz::Quiz("Foldback","A stage monitoring system used in live audio. A set of on-sta\
1479 ge speakers called monitors or wedges (or “foldback speakers” in British countries) \
1480 are fed a special mix of audio signals for the onstage performers to hear in order t\
1481 o play. This mix is usually different from the FOH (front-of-house) mix that the aud\
1482 ience hears, and is sometimes controlled by a second engineer through amplifiers and\
1483 speakers separate from the main sound system. This type of stage monitoring is freq\
1484 uently susceptible to feedback from the microphones, and in certain venues can cause\
1485 unwanted reflective noise that makes it difficult for FOH engineers to create a goo\
1486 d mix for the audience. For this reason, many live audio systems now use in-ear moni\
1487 toring as an alternative to stage monitors to control the onstage noise and reduce t\
1488 he risk of feedback."),
1489 quiz::Quiz("Foot Pedal","An effects device controlled by a musician with his foot."),
1490 quiz::Quiz("Foot Switch","A switch placed on the floor and pressed by a musician to \
1491 do various functions."),
1492 quiz::Quiz("Force-Sensing Resistor","In modular systems, an FSR (Force-Sensing or -S\
1493 ensitive Resistor) usually takes the form of a circular pad that you press on to var\
1494 y a parameter. It acts as a resistor that decreases in resistance the harder you pre\
1495 ss."),
1496 quiz::Quiz("Formant","Many instruments based on vibrating tubes – including our own \
1497 vocal tract – have certain frequencies that they like to vibrate or “resonate” at. W\
1498 hen you send a sound down these tubes, they will accentuate the frequency of that so\
1499 und (or some of its harmonics) to match these resonate frequencies. Each of these re\
1500 sonant frequencies is known as a formant of that instrument. A common way of synthes\
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1501 izing vocal-like sounds is to pass an oscillator through a filter or equalizer that \
1502 has several formant peaks, spaced apart in ways that mimic certain vowels. Formant i\
1503 s an element in the sound of a voice or instrument that does not change frequency as\
1504 different pitches are sounded. Formants are essentially “fixed” frequencies or reso\
1505 nances that occur as a result of the physical structure of the sound source. These f\
1506 requencies are what create timbre, that element of sound that creates the specific s\
1507 ound of a guitar, a flute, a male or female voice, etc.”),
1508 quiz::Quiz("Format","1) One of many different media used to store and reproduce audi\
1509 o, whether in the recording studio or for listening purposes. Examples include curre\
1510 ntly used physical formats such as vinyl records and compact discs; obsolete formats\
1511 such as cassette tape, 8-track tape and DAT; analog recording staples such as reel-\
1512 to-reel multitrack tape; and many different digital audio file formats such as mp3, \
1513 WAV, WMA, AIFF and others. 2) Format can also describe specific parameters when reco\
1514 rding to analog tape, such as number of tracks, width, spacing and order. 3) To prep\
1515 are a hard drive or memory card for use, usually erasing all existing data in the pr\
1516 ocess.”),
1517 quiz::Quiz("Four Quadrant Multiplier","A Four-Quadrant Multiplier is a special case \
1518 of Amplitude Modulation (AM). It is also referred to as ring or balanced modulation.\
1519 One signal changes the level of - \"multiplies\" - the level of a second signal. A \
1520 typical use is two VCOs running at audio rates fed into a ring modulator (a four-qua\
1521 drant multiplier). The output is a complex set of component tones that don’t follow \
1522 typical “musical” spacing based on octaves above the fundamental that harmonics usua\
1523 lly follow. Namely, the modulation frequency is both added to and subtracted from th\
1524 e carrier’s frequency; the resulting harmonics replace the original carrier and modu\
1525 lator. Say the carrier was a sine wave (only the fundamental harmonic present) at 60\
1526 0Hz, and the modulator was a sine wave at 100Hz. The result would be a tone that had\
1527 frequency components at 500 and 700Hz.”),
1528 quiz::Quiz("FracRack","A less-common format of modular synthesizers put forward by P\
1529 AiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5” (3.8 cm) wi\
1530 de by 3U, or 5.25” (13.3 cm) high.”),
1531 quiz::Quiz("Fractional Rack","A less-common format of modular synthesizers put forwa\
1532 rd by PAiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5” (3.8\
1533 cm) wide by 3U, or 5.25” (13.3 cm) high.”),
1534 quiz::Quiz("Frequency Modulation (FM) Synthesis","A method of sound synthesis in whi\
1535 ch the frequencies generated by one oscillator (the carrier) are altered by the outp\
1536 ut of one or more additional oscillators (operators) to create a diversity of harmon\
1537 ically rich sounds.”),
1538 quiz::Quiz("Frequency Range","1) The range of frequencies over which an electronic d\
1539 evice puts out a useful signal (see also “Bandwidth”). 2) The range of frequencies t\
1540 hat can be substantially transmitted or received in relation to a sound source. Each\
1541 instrument has a certain frequency range in which it can play; the human ear can al\
1542 so hear within a certain frequency range.”),
1543 quiz::Quiz("Frequency Response","The range between high and low frequencies that a c\
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1544 component of an audio system can adequately handle, transmit or receive."),
1545 quiz::Quiz("Frequency-Agile","In wireless microphone systems, frequency-agile descri\
1546 bes the ability of the system to operate on a choice of different RF frequencies wit\
1547 hin a certain bandwidth. Frequency-agile systems are preferred for live touring and \
1548 in areas with high concentrations of radio signals (like large cities) because the R\
1549 F frequency of the device can be changed to avoid interference."),
1550 quiz::Quiz("Frequency-Shift Key (FSK)","A now out-of-date protocol in which a sync t\
1551 one is recorded onto a spare track of a multi-track tape recorder to enable electron\
1552 ic devices (mainly drum machines) to perform in sync with the tape. While some older\
1553 devices still read FSK, an updated protocol (Smart FSK) is now more commonly used. \
1554 (See also "Smart FSK.")"),
1555 quiz::Quiz("Frequency","The number of occurrences of a particular event within a cer\
1556 tain amount of time. In audio and acoustics, frequency specifically refers to the nu\
1557 mber of complete cycles a vibration or waveform makes in a second, measured in cycle\
1558 s per second, or Hertz (Hz). In sound, frequency determines what we hear as pitch. T\
1559 he longer the wavelength, the fewer the cycles per second, and the lower the pitch." \
1560 ),
1561 quiz::Quiz("Front-of-House","(Abbreviated FOH) In live audio settings, the location \
1562 in a venue opposite the stage, where live audio for the show is controlled and mixed\
1563 ."),
1564 quiz::Quiz("FSR","In modular systems, an FSR (Force-Sensing or -Sensitive Resistor) \
1565 usually takes the form of a circular pad that you press on to vary a parameter. It a\
1566 cts as a resistor that decreases in resistance the harder you press."),
1567 quiz::Quiz("Full-Normalled","Describes the configuration within a patch bay in which\
1568 the jacks form a connected pathway until a patch cord is inserted to change the pat\
1569 h. When a patch bay is "full-normalled," the connection is altered by inserting a co\
1570 rd into either the input or output side; when it is "half-normalled," the path chang\
1571 es only when a cord is plugged into the input. "Non-normalled" or "open" means there\
1572 are no internal connections, and each input sends the signal through its correspond\
1573 ing output."),
1574 quiz::Quiz("Full-Wave Rectifier","A full-wave rectifier takes any negative voltages \
1575 and inverts them so they become positive. This effectively doubles the frequency of \
1576 many simple waveforms, like the triangle and sine."),
1577 quiz::Quiz("Function Generator","The term function generator can have two meanings i\
1578 n the world of synthesis. One, test equipment that generates waveforms such as sine \
1579 or square waves are often called "function generators." Two, envelope generators are\
1580 sometimes referred to as "function generators." In both cases, "function" means to \
1581 execute an equation of some sort, such as creating a periodic waveform such as a sin\
1582 e or creating a rise & fall in response to a trigger."),
1583 quiz::Quiz("Fundamental","(Also called fundamental frequency or first harmonic) The \
1584 lowest frequency present in the sounding of a note by musical instrument or voice.")\
1585 ,
1586 quiz::Quiz("Gain Control","A device that changes the gain of an amplifier or circuit\
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1587 , often a knob (potentiometer) that can be turned. In a mixing console, each channel\  
1588 usually has its own gain control to regulate the gain of the signal coming into the\  
1589 board—not to be confused with the channel “fader,” which regulates the output of an\  
1590 already-amplified signal.”),  
1591 quiz::Quiz("Gain Reduction","The action of a compressor or limiter in regulating the\  
1592 amplitude of the audio signal.”),  
1593 quiz::Quiz("Gain Structure","A term that describes the interconnection of multiple c\  
1594 omponents in an audio system, and the amount of gain increase or reduction that occu\  
1595 rs at each point. A configuration with a good gain structure means that the componen\  
1596 ts are working properly together to provide optimal gain with minimal distortion or \  
1597 noise.”),  
1598 quiz::Quiz("Gain","The amount of increase in audio signal strength, often expressed \  
1599 in dB.”),  
1600 quiz::Quiz("Gate Detector","This is one of the main signal types that are passed aro\  
1601 und inside a modular synthesizer. It jumps to high level – typically 5 volts – when \  
1602 a new note is supposed to start (such as when you press a key on a keyboard controll\  
1603 er), or when a sequencer jumps to the next “stage” or note. A gate typically stays a\  
1604 t that level for the duration of the note (i.e. while the key is being held down), a\  
1605 nd suddenly drops or “goes low” to its resting level – typically 0 volts, but someti\  
1606 mes -5 volts or another number – when the note ends (i.e. when the key is released).\  
1607 In practice, when a gate signal is sent to a typical envelope generator, the start \  
1608 of the gate (when it “goes high”) tells the envelope to go through its Attack and De\  
1609 cay stages; while the gate remains high, the envelope stays at its Sustain level, an\  
1610 d when the gate goes low again, the envelope moves onto its Release stage.”),  
1611 quiz::Quiz("Generation Loss","The amount of clarity lost when recorded audio is copi\  
1612 ed, due to added noise and distortion.”),  
1613 quiz::Quiz("Generation","A term used to describe the number of times that the record\  
1614 ed audio signal has been copied.”),  
1615 quiz::Quiz("Glide","Refers to a note that glides from one pitch to another while it \  
1616 is still audible. The music term for this effect is portamento, which is a slurring \  
1617 between notes. In a synthesizer, this effect is created by causing the control volta\  
1618 ge for the pitch of a note to slide from the pitch of the previous note rather than \  
1619 make a discrete jump. The module that creates this effect is sometimes known as a sl\  
1620 ew generator, slew limiter, slope generator, or lag. Some use the terms glide, gliss\  
1621 ando, and portamento interchangeably, but if you want to split musical hairs, a glis\  
1622 sando (gliss) is a different effect where the intermediate notes are more distinct –\  
1623 such as played rapidly in order – rather than slurred through.”),  
1624 quiz::Quiz("Golden Section","(also called Golden Ratio) A ratio of height to width t\  
1625 o length, where the width is approximately 1.6 times the height, and the length appr\  
1626 oximately 2.6 times the height. First calculated by the ancient Greeks, this ratio (\  
1627 known mathematically as “phi”) is used as an optimal ratio in many applications, inc\  
1628 luding room dimensions and studio design (to achieve “optimal acoustics” in the room\  
1629 ), and even in the design of certain acoustic instruments.”),
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1630 quiz::Quiz("Granular Synthesis","Granular synthesis can be thought of as particle th\
1631 eory applied to sound. The concept is that a sound can be broken down into very smal\
1632 l “grains” – typically 1-50 or 100 msec in duration. These tiny snippets are then pl\
1633 ayed back to reproduce the original sound, or to create new sounds by changing the s\
1634 peed, pitch, volume, playback order, and direction of the individual grains. You can\
1635 crossfade between these modified grains, or layer more grains on top. The result ca\
1636 n range from audio processing tricks such as changing speed without changing pitch a\
1637 nd vice versa, to creating psychedelic “clouds” of sound (and indeed, there is a pop\
1638 ular module called Clouds)."),
1639 quiz::Quiz("Graphic Equalizer","A type of equalizer that can adjust various frequenc\
1640 ies of the incoming signal using sliders that are assigned to specific frequency ban\
1641 ds. (See also “Equalizer.”)"),
1642 quiz::Quiz("Ground Lift Plug","An adapter that enables a three-prong power cord to p\
1643 lug into two-prong outlet. Some engineers wrongly use this plug to interrupt the gro\
1644 und connection and prevent buzz, but it is a VERY unsafe practice to break the groun\
1645 d connection using this plug without grounding the unit by another means."),
1646 quiz::Quiz("Ground Lift Switch","A switch that breaks the connection between the gro\
1647 und point in one circuit and the ground point in another circuit, for the purpose of\
1648 eliminating hum or buzz caused by ground loops."),
1649 quiz::Quiz("Ground Loop","A situation caused when one or more electronic devices are\
1650 connected to the same ground at different points. The devices operate at different \
1651 ground potentials, which creates voltage along the ground, resulting in a low-freque\
1652 ncy hum that can be annoying at best and cause damage to gear at worst. The best res\
1653 olution for ground loops is to ground all devices at the same point using a central \
1654 power source. An alternative solution is to break the loop via ground lift switches \
1655 or plugs, but this should be avoided when possible as it is considered an unsafe man\
1656 agement of electricity."),
1657 quiz::Quiz("Group (or Grouping)","A number of input channels on a console that can b\
1658 e controlled and adjusted as a single set before sending the combined signal to the \
1659 master output. Sometimes also called “Submix,” “Bus” or just “Group.”"),
1660 quiz::Quiz("Group Delay","In audio, group delay is a phenomenon within all electroni\
1661 c audio devices (e.g., speakers, amplifiers) in which different frequencies in the s\
1662 ignal are output at slight delays from one another. In simpler terms, lower frequenc\
1663 ies are delivered slightly more slowly than higher ones. In all devices, there is an\
1664 inherent delay between input and output of the signal, but group delay specifically\
1665 deals with the time delays between specific frequencies of the sound. The goal in a\
1666 ny configuration is to keep the group delay as small as possible; in cases of extrem\
1667 ely poor configurations, the delays between highs and lows can be audible."),
1668 quiz::Quiz("Guitar Controller","An electric guitar (or device played like a guitar) \
1669 that transmits MIDI data that can be used to control synthesizers and sound modules.\
1670 "),
1671 quiz::Quiz("Guitar Processor","A device that adds audio effects to a direct guitar s\
1672 ignal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, \
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1673 etc. Effects processors can occur as individual effects boxes or multi-sound pedal b\
1674 oards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between t\
1675 he guitar and the console. They can also be found as presets in guitar amplifiers, o\
1676 r even as digital plug-ins within a DAW.”),
1677 quiz::Quiz("Haas Effect","(Also called Precedence Effect) Simply stated, a factor in\
1678 human hearing in which we perceive the source of a sound by its timing rather than \
1679 its sound level. In his research, Helmut Haas determined that the first sound waves \
1680 to reach our ears help our brains determine where the sound is coming from, rather t\
1681 han its reflection or reproduction from another source. The reflection of the sound \
1682 must be at least 10dB louder than the original source, or delayed by more than 30ms \
1683 (where we can perceive it as an echo), before it affects our perception of the direc\
1684 tion of the sound. This is what helps us distinguish the original sound source witho\
1685 ut being confused by reflections and reverberations off of nearby surfaces. Understa\
1686 nding the Haas effect is particularly useful in live audio settings, especially in l\
1687 arge venues where loudspeakers are time-delayed to match the initial sound waves com\
1688 ing from the source.”),
1689 quiz::Quiz("Half Step","A change in pitch equivalent to adjacent keys on a piano. Al\
1690 so known as a “semitone.””),
1691 quiz::Quiz("Half-Normalled","Describes the configuration within a patch bay in which\
1692 the jacks form a connected pathway until a patch cord is inserted to change the pat\
1693 h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\
1694 rd into either the input or output side; when it is “half-normalled,” the path chang\
1695 es only when a cord is plugged into the input. “Non-normalled” or “open” means there\
1696 are no internal connections, and each input sends the signal through its correspond\
1697 ing output.”),
1698 quiz::Quiz("Half-Wave Rectifier","A half-wave rectifier passes only positive voltage\
1699 s, and replaces anything negative with 0v. In other words, anything “below zero” is \
1700 clipped off.”),
1701 quiz::Quiz("Hall Program","A setting of a digital delay/reverb effects unit that app\
1702 proximates concert halls. Hall programs are characterized by pre-delay of up to 25 ms\
1703 .”),
1704 quiz::Quiz("Hard Knee","In compression, refers to a more abrupt introduction of comp\
1705 ression of the signal once the sound level crosses the threshold. (See also “Knee.”)\
1706 ”),
1707 quiz::Quiz("Hard Sync","This is the most common type of oscillator sync where the sl\
1708 ave oscillator will reset its waveform whenever it receives a sync pulse. If the typ\
1709 e of sync is not specified, then it’s probably hard sync.”),
1710 quiz::Quiz("Harmonic Distortion","The presence of harmonics in the output signal of \
1711 a device which were not present in the input signal, usually for the purpose of chan\
1712 ging the instrument’s timbre.”),
1713 quiz::Quiz("Harmonic","A single harmonic is the purest sound possible: It contains n\
1714 o overtones or other identifying characteristics aside from its pitch and loudness. \
1715 The shape of its vibration – whether it be vibrating the air so you can hear it, or \
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1716 causing the electrical vibrations of a voltage going up and down – is a sine wave. M\
1717 ost of the time, overtones have a very specific pitch relationship to each other. Th\
1718 e first or lowest harmonic – known as the ‘fundamental’ – is the pitch of the sound,\
1719 just as the lowest note of a chord is its ‘root.’ The other harmonics are higher, a\
1720 nd spaced out as integer multiples of the fundamental: two times its frequency, thre\
1721 e times, four times, and so forth. The first few harmonics happen to have a nice mus\
1722 ical spacing: an octave; an octave and a fifth; two octaves. But the higher they get\
1723 , the less musical they may seem."),
1724 quiz::Quiz("Harmonics","Whole number multiples of the fundamental frequency that occ\
1725 ur naturally within the playing of a tone. Mathematically, if the fundamental freque\
1726 ncy is x, the harmonics would be 2x, 3x, 4x, etc. For example, if the fundamental fr\
1727 equency of the note played is 440Hz (or A-440), the harmonics would be 880Hz, 1320Hz\
1728 , 1760Hz, and so on. The presence of harmonics in the tone is what creates the timbr\
1729 e of an instrument or voice."),
1730 quiz::Quiz("Head","In tape recording, an electromagnetic transducer that magneticall\
1731 y affects the tape passing over it. Recording/playback heads change the audio signal\
1732 from electrical energy to magnetic energy and back, for recording and playback purp\
1733 oses. An erase head creates a powerful electromagnetic field to the tape to erase pr\
1734 evious signals from the tape."),
1735 quiz::Quiz("Headroom","The difference in dB between normal operating level and clipp\
1736 ing level in an amplifier or audio device. Also describes the difference in dB betw\
1737 een the peak levels of a recording and the point at which the signal distorts. (Also\
1738 called “Margin.”)"),
1739 quiz::Quiz("Hertz/Volt","A system where a change of 1 volt at the input results in a\
1740 change in pitch of a fixed number of hertz (cycles per second), rather than a fixed\
1741 musical interval."),
1742 quiz::Quiz("Hertz","(Abbreviated Hz) 1) The unit of measurement for frequency, speci\
1743 fically, the number of complete wave cycles that occur in a second (cycles per secon\
1744 d). 1 Hz = 1 complete wave per second. 2) A popular rental car company (not typicall\
1745 y used in recording except for transport to the studio)."),
1746 quiz::Quiz("Hi-Hat","In drum sets, double cymbal on a stand, usually positioned next\
1747 to the snare, which can be played with a foot pedal and/or by the top cymbal being \
1748 hit with a stick."),
1749 quiz::Quiz("Hi-Z","(abbreviated Hi-Z) Described as an impedance or resistance of sev\
1750 eral thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more o\
1751 hms. (See also “Impedance.”)"),
1752 quiz::Quiz("High (gate)","When a gate signal is at the voltage level (typically 5 vo\
1753 lts, although it can be more) that indicates it is “on” – such as when a note is bei\
1754 ng held down on a keyboard controller – it is said that the gate is high."),
1755 quiz::Quiz("High Impedance","(abbreviated Hi-Z) Described as an impedance or resista\
1756 nce of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000\
1757 or more ohms. (See also “Impedance.”)"),
1758 quiz::Quiz("High Pass Filter","An audio filter that attenuates signals below a certa\
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1759 in frequency (the cut-off frequency) and passes signals with frequencies that are hi\  
1760 gher."),  
1761 quiz::Quiz("High-End","Highs or High-End – Short for “high frequencies,” loosely the\  
1762 frequencies above 4000 Hz. Usually meant in the context of “highs, mids and lows” i\  
1763 n an audio signal."),  
1764 quiz::Quiz("High-Pass Filter","The high pass filter (HPF) design passes harmonics ab\  
1765 ove its cutoff or corner frequency untouched, and reduces the level of lower harmoni\  
1766 cs depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) hi\  
1767 gh pass filter, harmonics one octave below the cutoff frequency (in other words, one\  
1768 half the cutoff frequency) are reduced in level by 12 dB; harmonics two octaves bel\  
1769 ow the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pa\  
1770 ss filters are typically used to create bright sounds where the higher harmonics are\  
1771 much stronger than the fundamental and lower harmonics – for example, the sound of \  
1772 a harpsichord."),  
1773 quiz::Quiz("Horizontal Pitch","HP = Horizontal Pitch. In the Eurorack format for syn\  
1774 thesizer modules, the width of a module is defined as the number of hp (horizontal p\  
1775 itch) units. Each hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, alt\  
1776 hough some are odd numbers. Also, modules tend to be ever so slightly less than exac\  
1777 tly some multiple of 0.2” wide, just to make sure you don’t run into problems with e\  
1778 ver so slightly too wide modules overlapping."),  
1779 quiz::Quiz("Horn","1) A speaker or speaker enclosure where sound waves are sent by a\  
1780 speaker cone or driver into a narrow opening which flares out to a larger opening. \  
1781 2) One of several different types of brass musical instruments."),  
1782 quiz::Quiz("House Sync","A reference signal such as SMPTE time code that is used to \  
1783 keep all devices in the room in sync."),  
1784 quiz::Quiz("HP","HP = Horizontal Pitch. In the Eurorack format for synthesizer modul\  
1785 es, the width of a module is defined as the number of hp (horizontal pitch) units. E\  
1786 ach hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, although some are\  
1787 odd numbers. Also, modules tend to be ever so slightly less than exactly some multi\  
1788 ple of 0.2” wide, just to make sure you don’t run into problems with ever so slightl\  
1789 y too wide modules overlapping."),  
1790 quiz::Quiz("HPF","The high pass filter (HPF) design passes harmonics above its cutof\  
1791 f or corner frequency untouched, and reduces the level of lower harmonics depending \  
1792 on how far below the cutoff they are. In a 12dB/oct (decibel/octave) high pass filte\  
1793 r, harmonics one octave below the cutoff frequency (in other words, one half the cut\  
1794 off frequency) are reduced in level by 12 dB; harmonics two octaves below the cutoff\  
1795 (one quarter the frequency) are reduced by 24dB, and so forth. High pass filters ar\  
1796 e typically used to create bright sounds where the higher harmonics are much stronge\  
1797 r than the fundamental and lower harmonics – for example, the sound of a harpsichord\  
1798 ."),  
1799 quiz::Quiz("Hum","1) The low-frequency pitch that occurs when power line current is \  
1800 accidently induced or fed into electronic equipment. The hum reflects the fundamenta\  
1801 l frequency of the current (60 Hz in the U.S., and 50 Hz in many European countries)\
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1802 . 2) To vocalize a pitch without opening one's mouth."),
1803 quiz::Quiz("Hybrid Power Supply","A hybrid power supply uses a lower weight, more ef\
1804 ficient switching power supply to perform most of the drop in voltage – say, from 12\
1805 0v AC to 15v DC – and then uses a linear power supply for the remaining much smaller\
1806 drop, such as from 15v to 12v. These are becoming the preferred design in many modu\
1807 lar synthesizer enclosures. Shortcomings with the power supply – too noisy, or not e\
1808 nough – tend to be at the cause of many unexpected problems in modular synthesizers.\
1809 "),
1810 quiz::Quiz("Hypercardioid","A variation of the cardioid microphone pick up sensitivi\
1811 ty pattern in which the shape of the optimal pickup area is tighter and more directi\
1812 onal than cardioid. Hypercardioid microphones are most sensitive directly on-axis in\
1813 front of the microphone, and begins rejecting sounds between 90-150 degrees off-axi\
1814 s, depending on the tightness of the pattern."),
1815 quiz::Quiz("Hz/V","A system where a change of 1 volt at the input results in a chang\
1816 e in pitch of a fixed number of hertz (cycles per second), rather than a fixed music\
1817 al interval."),
1818 quiz::Quiz("Hz","An abbreviation for the term Hertz, or the unit of frequency."),
1819 quiz::Quiz("IADSR","This is an Attack/Decay/Sustain/Release (ADSR) envelope generato\
1820 r that allows you to start the attack phase at an initial level – the “I” – rather t\
1821 han the customary 0 volts. The envelopes in the Prophet VS, as well as a module from\
1822 Ladik, have this capability."),
1823 quiz::Quiz("IC","Integrated Circuit – A miniature circuit of many components set on \
1824 semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”\
1825 "),
1826 quiz::Quiz("Imaging","Refers to the ability to localize a specific sound within the \
1827 sound space. In recording environment, it refers to “placing” instruments within the\
1828 stereo or surround field so that it when the sound is played through speakers, it f\
1829 ools our ears into thinking the sound source is in emanating from a specific point i\
1830 nstead of from the speakers. In live audio and sound reinforcement, the principle of\
1831 imaging is the same, the goal being to make the audience perceive the sounds as com\
1832 ing from performers on the stage, rather than from the speakers."),
1833 quiz::Quiz("Impedance","Refers to the resistance of a circuit or device to alternati\
1834 ng current, which can be mathematically described as the ratio of voltage to current\
1835 . Differences in impedance between devices in the studio can affect how they work to\
1836 gether. Impedance is abbreviated by the letter Z, and measured in ohms (W)."),
1837 quiz::Quiz("In Line Console","An audio mixing console that is designed and configure\
1838 d so each channel strip can be used for both recording and monitoring functions duri\
1839 ng multitrack recording. This configuration is in contrast to split mixing consoles,\
1840 which requires separate channels on the board for recording and monitoring function\
1841 s."),
1842 quiz::Quiz("In Phase","The desirable situation in which two or more devices (and the\
1843 ir respective audio signals) are on the same side of the polarity spectrum, producin\
1844 g waveforms that do not conflict or cancel each other out."),
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1845 quiz::Quiz("In Port","A jack on a MIDI device or computer that will accept an incomi\
1846 ng data signal."),
1847 quiz::Quiz("Inductance","A characteristic of electrical conductors in which electric\
1848 al charge (voltage) is produced or stored magnetically due to the natural resistance\
1849 to change in the electrical current. Inductance is an electromagnetic principle tha\
1850 t can either assist in audio applications (as in loudspeakers) or cause resistance (\
1851 as in using speaker wire whose gauge is too low for the application)."),
1852 quiz::Quiz("Inductor","A device (usually a coil of wire) that converts electrical en\
1853 ergy into stored magnetic energy as electrical current passes through it. Commonly f\
1854 ound in a variety of audio applications such as guitar pickups and loudspeakers."),
1855 quiz::Quiz("Infinite Baffle","A loudspeaker mount or enclosure designed so that soun\
1856 d waves coming from the front theoretically do not reach the back, preventing the so\
1857 und waves from cancelling each other out. The term "infinite" comes from the idea th\
1858 at mounting the speaker on a wall with no end points would not allow sound waves to \
1859 migrate behind it. Of course, this is physically impossible, so infinite baffles are\
1860 designed to replicate this as much as possible. Examples of infinite baffles are mo\
1861 unting the speaker on a wall of an enclosed room, or building it inside a sealed cab\
1862 inet large enough to prevent rear sounds from affecting the cone from the back."),
1863 quiz::Quiz("Initial/Attack/Decay/Sustain/Release","This is an Attack/Decay/Sustain/R\
1864 elease (ADSR) envelope generator that allows you to start the attack phase at an ini\
1865 tial level - the "I" - rather than the customary 0 volts. The envelopes in the Proph\
1866 et VS, as well as a module from Ladik, have this capability."),
1867 quiz::Quiz("Input / Output (I/O)","I/O - An abbreviation for "Input/Output." In audi\
1868 o, it refers to any device, program or system involving the transferring of electric\
1869 al/audio signals or data."),
1870 quiz::Quiz("Input Impedance","The opposition to current flow by the first circuits o\
1871 f a device."),
1872 quiz::Quiz("Input Monitoring","A setting on many DAWs that allows you to monitor the\
1873 live input signal coming into the DAW (as opposed to the recorded signal)."),
1874 quiz::Quiz("Input","The jack or physical location where a device receives a signal. \
1875 Also refers to the incoming signal itself."),
1876 quiz::Quiz("Insert","An access in the signal chain (usually in the mixing console or\
1877 virtually within a DAW) in which a device, signal processor or digital plug-in can \
1878 be "inserted" into the circuit between pre-amplification and the channel or bus outp\
1879 ut. Commonly used to add processing such as reverb, compression or EQ to a channel o\
1880 r group of channels."),
1881 quiz::Quiz("Instrument Amplifier","A device that has a power amplifier and speaker t\
1882 o reproduce the signal put out by an electric instrument."),
1883 quiz::Quiz("Instrument Out Direct","Feeding the output of an electric instrument (li\
1884 ke an electric guitar) directly to the recording console or tape recorder, as oppose\
1885 d to miking the amplifier."),
1886 quiz::Quiz("Insulator","A substance such as glass, air, plastic, etc., that will (fo\
1887 r all practical purposes) not conduct electricity."),
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1888 quiz::Quiz("Integrated Circuit","Integrated Circuit (Abbreviated “IC”) – A miniature\  
1889 circuit of many components set on semiconductor material, used in electronics. A fa\  
1890 ncy term for “chip” or “microchip.””),  
1891 quiz::Quiz("Integrator","This function smoothens out an incoming signal so that the \  
1892 change in voltage level. “Integrator” is the technical name for this math function; \  
1893 you are more likely to see this module called a slew limiter (where I go into more d\  
1894 etail on its uses) or less often as a lag generator or processor.”),  
1895 quiz::Quiz("Interface","Any device or connection point that allows one unit to work,\  
1896 drive or communicate with another unit, or that allows a human to interact with a c\  
1897 omputer or other electronics. There are many examples of interfaces in professional \  
1898 audio situations, including MIDI (Musical Instrument Digital Interface); audio inter\  
1899 faces which connect audio inputs to your computer; and even your DAW program, which \  
1900 displays a screen that enables you to assign instruments, adjust settings, record, m\  
1901 ix and playback. Even the mixing console is an interface of sorts, connecting the ma\  
1902 ny elements of the control room.”),  
1903 quiz::Quiz("Intermodulation (IM) Distortion","Distortion caused by two or more audio\  
1904 signals of different frequencies interacting with one another. The sum and differen\  
1905 ce of the frequencies produce new (usually unwanted frequencies) that didn’t exist i\  
1906 n any of the original frequencies.”),  
1907 quiz::Quiz("Inverse Square Law","A mathematical rule that describes an inverse relat\  
1908 ionship between one quantity and the square of another quantity. In plain English, o\  
1909 ne number goes down by a certain amount each time the other number doubles. In audio\  
1910 and acoustics, the inverse square law says that in an open sound field with no obst\  
1911 ructions, the sound pressure level will drop by half (6dB) each time the distance fr\  
1912 om the sound source is doubled. (This equation is quite useful to audio engineers tr\  
1913 ying to provide sound in open-air settings, for example.)”),  
1914 quiz::Quiz("Inverter","An inverter multiplies an incoming control voltage by -1. In \  
1915 the case of a gate or logic inverter, it reverses the high and low states so that (f\  
1916 or example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pola\  
1917 rizer, as it changes the polarity (+ versus -) of a signal. A control voltage invert\  
1918 er is often combined with an offset voltage to adjust the output voltage into the de\  
1919 sired range. For example, if you had an envelope generator that had an output range \  
1920 of 0 to +8 volts, and you just inverted it, the result would be 0 to -8 volts. Since\  
1921 some modules such as voltage controlled amplifiers usually expect only positive vol\  
1922 tages, you would then need to add 8 volts to that result to get an upside-down (inve\  
1923 rted) envelope that still had an overall range of 0 to +8v.”),  
1924 quiz::Quiz("Inverting Mixer","Most signal mixers make an effort to keep the same pol\  
1925 arity of a signal as it passes through the mixer. However, some mixers may invert th\  
1926 e polarity or “phase” of a signal (as it’s a simpler design); other mixers may allow\  
1927 you to invert a signal on purpose so that you can experiment with tricks like addin\  
1928 g one waveform or filter mode output out of phase with another coming from the same \  
1929 oscillator or filter.”),  
1930 quiz::Quiz("Isolation","The process of containing sound within a certain area so tha\  

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1931 t it doesn't interact with other sounds. For example, acoustically treated isolation\  
1932 booths are often used to record vocals or instruments in the studio to keep outside\  
1933 noises from bleeding into the recording microphone, or likewise to keep vocals or o\  
1934 ther sounds away from instrument mics during live recording sessions."),  
1935 quiz::Quiz("IV Cable","You often need to send one signal to multiple destinations. O\  
1936 ptions for doing this include using dedicated multiples, free-floating widgets with \  
1937 multiple jacks wired together, or fancy cables that allow you plug one or two extra \  
1938 cables into them. The IV cable is one the latter: Made by Erthenvar, it has an extra\  
1939 3.5mm jack molded into the mid-point of the cable (loosely resembling an intravenou\  
1940 s or "IV" drip), in addition to having 3.5mm plugs at either end."),  
1941 quiz::Quiz("Jack","That hole you plug your patch cables into on the face of your syn\  
1942 thesizer modules? That's called a jack. The size and type of jack - 3.5mm, banana, o\  
1943 r 1/4" - often is one of the defining features of different synth module formats: 3U\  
1944 /Eurorack, 4U, and 5U/MU respectively. (No, a plug is not called a Jill. Actually, i\  
1945 t's the other way around: A plug is sometimes referred to as a male connector, and a\  
1946 jack is referred to as a female connector.)"),  
1947 quiz::Quiz("Jam Sync","A process available on some clock or syncing devices which re\  
1948 ads an external time code and recreates (or "jams") a new time code identical to the\  
1949 original external code for the syncing of devices. This function is mainly used for\  
1950 replacing code that has become degraded."),  
1951 quiz::Quiz("Karplus Strong","This is a physical modeling synthesis algorithm designe\  
1952 d to replicate the sound of plucked, vibrating strings - although it has also proven\  
1953 useful for some percussion sounds as well. A short sample - originally noise, altho\  
1954 ugh it can be a high frequency chirp or other sound - is sent to both the output, an\  
1955 d to a delay line. The output of a delay line is connected to a filter - originally \  
1956 a one-pole low pass filter; changing the filter has a huge effect on the character o\  
1957 f the sound - and then back to both the main output and the input of the delay line.\  
1958 A few modules implement Karplus Strong synthesis, although it is an interesting cha\  
1959 llenge to patch yourself and play with the results."),  
1960 quiz::Quiz("Key","1) In music, the note scale in which a piece of music is written o\  
1961 r played, identified by the first note (tonic) of the scale, as in, "Key of C." 2) T\  
1962 he control of a dynamics processing device by an external audio signal through the u\  
1963 se of a side chain. 3) A digital or data code that unlocks the use of a device or so\  
1964 ftware. Example: Pro Tools is licensed through an iLok ID via the use of a physical \  
1965 USB key."),  
1966 quiz::Quiz("Keyboard Controller","A piano-styled keyboard that sends out MIDI signal\  
1967 s to control other MIDI devices. Most keyboard instruments are equipped with MIDI co\  
1968 ntrol capabilities, but dedicated MIDI keyboard controllers emit no audio signals, o\  
1969 nly MIDI data."),  
1970 quiz::Quiz("Keyboard Tracking","Most modular synths follow a strict relationship bet\  
1971 ween voltage and pitch, such as 1 volt per octave; any deviation would cause tuning \  
1972 errors. Because of this sensitivity, 1v/oct and similar signals and connections are \  
1973 sometimes specifically distinguished as keyboard tracking rather than just "CV" (con
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1974 trol voltage) to make it clear they are not attenuated or otherwise modified when co\  
1975 ntrolling a function on a module."),  
1976 quiz::Quiz("Keyboard","Any musical instrument or computer controlled by pressing a k\  
1977 ey."),  
1978 quiz::Quiz("Keytar","A strap-on, lightweight, portable keyboard meant to allow keybo\  
1979 ardists the same freedom (not to mention posturing opportunities) as guitarists."),  
1980 quiz::Quiz("Kick Drum","The bass drum on a trap drum set, so called because it is pl\  
1981 ayed with a kick pedal."),  
1982 quiz::Quiz("Kilohertz (kHz)","kHz – An abbreviation for kilohertz (1000 Hz, or 1000 \  
1983 cycles per second). Example: 2000 Hz = 2 kHz. Most commonly used in the studio for d\  
1984 escribing audio frequency ranges or digital sampling rates."),  
1985 quiz::Quiz("Knee","A function on a compressor that determines how abruptly or gradua\  
1986 lly compression begins once the sound level crosses the threshold. So-called because\  
1987 the graphic “bend” in the response curve is reminiscent of a knee. “Hard knee” refe\  
1988 rs to an abrupt activation of the compressor, while “soft knee” refers to a more gra\  
1989 dual change."),  
1990 quiz::Quiz("Krell Patch","Recreating this patch is a challenge many modular musician\  
1991 s like to tackle. It is based on the 1959 movie Forbidden Planet, in a segment where\  
1992 they supposedly play the music of the ancient Krell race. In general terms, each no\  
1993 te has a random pitch, envelope, and duration."),  
1994 quiz::Quiz("Lag Generator","This function smoothes out an incoming signal so that th\  
1995 e change in voltage level cannot exceed a certain number of volts per second. This c\  
1996 auses the result to “lag behind” changes in the input. It is sometimes called a slew\  
1997 limiter or technically as an integrator."),  
1998 quiz::Quiz("Layering","Refers to almost any blending of similar multiple musical par\  
1999 ts or sounds at once, often combined on one channel or assigned to one controller. I\  
2000 n audio recording, layering usually involves recording similar takes of the same ins\  
2001 trument or vocal (or duplicating parts with slight delays or chorusing effects) to c\  
2002 reate a fuller, richer sound than the vocal/instrument by itself. In sound design, i\  
2003 t also refers to blending multiple samples (example: two or more drum sounds) to cre\  
2004 ate a fuller sound."),  
2005 quiz::Quiz("Lead Sheet","A shorthand form of music notation (similar to a chord char\  
2006 t) that displays the basic essential elements of a song so musicians can follow alon\  
2007 g without the full notation of every note or expression. Lead sheets most commonly i\  
2008 nclude a melody line written in music notation with chord changes above the staff, a\  
2009 nd lyrics below it. (See also “Chord Chart.”)" ),  
2010 quiz::Quiz("Leakage","Sounds from other instruments and sound sources that were not \  
2011 intended to be picked up by the microphone."),  
2012 quiz::Quiz("Level","The amount of signal strength; the amplitude, especially the ave\  
2013 rage amplitude."),  
2014 quiz::Quiz("LFO","This module produces repetitive, cycling waves ranging in frequenc\  
2015 y from the low end of the audio spectrum to as slow as many seconds or even minutes \  
2016 per cycle. They are used to produce effects such as tremolo (when controlling the lo
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2017 udness of a signal), vibrato (when controlling the pitch of a signal), repetitive fi\
2018 lter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an \
2019 oscillator, and more."),
2020 quiz::Quiz("Limiter","A type of compressor that sharply reduces (limits) the gain of\
2021 the signal when the audio level reaches a certain threshold, typically used to prev\
2022 ent overload and signal peaking. A compressor effectively becomes a limiter when its\
2023 ratio is 10:1 or higher. (See also "Compressor.")"),
2024 quiz::Quiz("Line Input","Line Input ("Line In") – An input designed to take a line l\
2025 evel signal."),
2026 quiz::Quiz("Line Level","Most consumer and lower-cost professional audio equipment u\
2027 se a signal level reference known as line level or -10dBV (decibel volts). The most \
2028 common connectors are RCA (phono) or 3.5mm, although 1/4" is also used; the signal i\
2029 s "unbalanced" (it uses two wires: signal and ground). In the line level standard, a\
2030 sine wave that varies between +/-0.447 volts is considered to be at -10dBV. By cont\
2031 rast, a typical oscillator signal in a modular synthesizer is +/-5 to +/-8 volts. As\
2032 a result, you will need either an output module in your modular synth or one heckuv\
2033 a input attenuator on your mixer or recorder to plug your synth into equipment that \
2034 runs at line level. Similarly, you will need to substantially boost a line level sig\
2035 nal to get it up to modular standards to process in your modular synth."),
2036 quiz::Quiz("Line Output","Line Output ("Line Out") – Any output that sends out a lin\
2037 e level signal, such as the output of a console that feeds a recorder."),
2038 quiz::Quiz("Linear FM","This is often the preferred input response for frequency mod\
2039 ulating (FM'ing) an oscillator, as the result stays in tune while you change the mod\
2040 ulator."),
2041 quiz::Quiz("Linear Power Supply","A linear power supply design takes a higher incomi\
2042 ng voltage and reduces it to a lower voltage using components such as transformers. \
2043 In very general terms, they tend to introduce less noise into the output power signa\
2044 l, at the cost of increased heat and weight (they're not very efficient). Many are m\
2045 oving to a hybrid power supply that combines a switcher with a small linear supply o\
2046 r regulator to get the best of both worlds."),
2047 quiz::Quiz("Linear VCA","A linear voltage-controlled amplifier (VCA) uses a simple m\
2048 athematical relationship between control voltage input and signal level output – for\
2049 example, 50% of nominal control voltage in would result in the output signal being \
2050 at 50% of the level of the input signal. This, however, is not how our ears perceive\
2051 loudness; a sound must be amplified by 10x in order to be perceived as twice as lou\
2052 d. This makes a linear VCA desirable for scaling control voltages, but perhaps less \
2053 so for scaling audio signals. If you connect an envelope generator with an exponenti\
2054 al output to a linear VCA, then you will get the desired aural result. Confusing? Th\
2055 at's why it's great when an envelope generator or VCA has a switch or control to var\
2056 y it between linear and exponential response. A linear mixer is similar to a linear \
2057 VCA: "half" on the input level control equals the output having half the voltage swi\
2058 ng as the input. Again, this is fine for altering control voltages, but not for mixi\
2059 ng audio signals; in that case you want a mixer with exponential controls."),
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2060 quiz::Quiz("Linear VCO","A linear voltage-controlled oscillator (VCO) follows the vo\
2061 lts/hertz (v/Hz) standard; more common is the exponential volts/octave (v/oct) stand\
2062 ard. The exception is frequency modulation (FM), where a linear control voltage inpu\
2063 t is often preferred to recreate classic style FM as it does not change the fundamen\
2064 tal pitch of the carrier oscillator."),
2065 quiz::Quiz("Live Recording","A recording session where all the musicians are playing\
2066 at once with no overdubbing."),
2067 quiz::Quiz("Live Room","The large, main room of the recording studio where most of t\
2068 he instruments and/or vocalists perform. So called, not just because there is room f\
2069 or live performances, but because the room has been acoustically treated to produce \
2070 a pleasing amount of live reverberation."),
2071 quiz::Quiz("Live","1) A term describing a space with a reverberant or reflected soun\
2072 d. In a “live” space, the sound waves are active or “live.” 2) Occurring in real ti\
2073 me, as opposed to previously recorded."),
2074 quiz::Quiz("Local On/Off","Local On/Off – A MIDI message that controls the internal \
2075 sound module of a synthesizer or MIDI controller. “Local On” triggers the internal m\
2076 odule when the keyboard is played; “Local Off” disconnects it. “Local Off” is freque\
2077 ntly used to prevent unwanted looping of MIDI messages in some configurations, or wh\
2078 en controlling the internal module via another controller."),
2079 quiz::Quiz("Logic Functions","In a modular synth, control voltages tend to be contin\
2080 uous in nature, while gate and trigger signals are binary: on or off; high or low. T\
2081 his is the same as logic signals in digital circuitry. Therefore, some make digital \
2082 logic modules. A common logic function is OR: If either signal A or signal B is high\
2083 (on), then output a high gate signal (on); otherwise output a low gate (off). Anoth\
2084 er is AND: If and only if signal A and signal B are both, then output a high gate (o\
2085 n); otherwise, output a low gate (off). These are great functions for combining beat\
2086 triggers from different timing sources."),
2087 quiz::Quiz("Logic","Binary or Boolean logic is a way of combining gate signals (on o\
2088 r off voltages) to create new outputs. Each section of a logic module typically incl\
2089 udes 1 to 3 inputs, with 2 being the most common. An OR function says if there is a \
2090 gate on (or “high”) signal at any of the inputs (i.e. input 1 or input 2 or input 3,\
2091 etc.), to output a gate on signal. An AND function says only output a gate on signa\
2092 l if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all ha\
2093 ve gate ons). Adding an “N” to the front of a function’s name says “not” this functi\
2094 on – in other words, a NOR function would only output a high signal if all inputs we\
2095 re low (not input 1 nor input 2 are high)."),
2096 quiz::Quiz("Loop","1) Effectively, any piece of music or data that repeats endlessly\
2097 . Before digital audio and sampling, loops were created by looping tape. Today, loop\
2098 s are used in samples to sustain a sampled note for as long as the note is triggered\
2099 , while drum loops and other music loops are common in modern music production. 2) A\
2100 nother term for antinode, or the points of maximum displacement of motion in a vibra\
2101 ting stretched string or a sound wave. (See also “Standing Wave.”)"),
2102 quiz::Quiz("Looping","Sometimes it’s useful to have a module loop or repeat its func\
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2103 tions. For example, an envelope generator that can be set to loop becomes a low freq\
2104 uency oscillator: as it attacks to a maximum value and decays back to zero, it start\
2105 s that attack phase again. Quite often you want a note sequencer to loop: When it re\
2106 aches the last note in the sequence, it would be useful for it to then look back to \
2107 or return to the first note and start over. Audio recorders with looping features ar\
2108 e also popular for live performance."),
2109 quiz::Quiz("Loudness","A term referring to how the human ear perceives incoming soun\
2110 d waves. This term seems self-explanatory, but it's deceptive. We commonly think of \
2111 loudness as it relates to the volume of a sound, but this is an indirect relationshi\
2112 p. In acoustic terms, volume is more about the amplitude of the sound waves, while l\
2113 oudness describes how our ears hear the intensity of those waves."),
2114 quiz::Quiz("Low (gate)","Most often, this is shorthand for saying a gate or trigger \
2115 signal is in its "off" condition (typically 0 or -5 volts, in contrast to a "high" o\
2116 r "on" signal of +5 volts)."),
2117 quiz::Quiz("Low Frequency Oscillator","This module produces repetitive, cycling wave\
2118 s ranging in frequency from the low end of the audio spectrum to as slow as many sec\
2119 onds or even minutes per cycle. They are used to produce effects such as tremolo (wh\
2120 en controlling the loudness of a signal), vibrato (when controlling the pitch of a s\
2121 ignal), repetitive filter wah-wah effects, pulse width modulation to vary the wavesh\
2122 ape of a pulse in an oscillator, and more."),
2123 quiz::Quiz("Low Impedance","(abbreviated Lo-Z) Described as impedance of 500 ohms or\
2124 less. (See also "Impedance.")"),
2125 quiz::Quiz("Low Pass Filter","The low pass filter (LPF) design passes harmonics belo\
2126 w its cutoff or corner frequency untouched, and reduces the level of lower harmonics\
2127 depending on how far above the cutoff they are. In a 12dB/oct (decibel/octave) low \
2128 pass filter, harmonics one octave above the cutoff frequency (in other words, double\
2129 cutoff frequency) are reduced in level by 12 dB; harmonics two octaves above the cu\
2130 toff (four times the frequency) are reduced by 24dB, and so forth. This is the most \
2131 common type of filter used, as most natural sounds have stronger low harmonics and w\
2132 eaker high harmonics - especially as a note fades to silence."),
2133 quiz::Quiz("Low Pass Gate","By strict definition, a low pass gate (LPG) is a low pas\
2134 s filter whose cutoff frequency goes down into the subsonic range as its control vol\
2135 tage goes towards 0 volts, resulting in the input signal being filtered almost into \
2136 silence. Some replicate this by combining a low pass filter and a voltage controlled\
2137 amplifier into the same module, with both following the same control voltage. In ei\
2138 ther case, as an input envelope falls from a high level to 0 volts, the output gets \
2139 duller (higher harmonics are filtered more) as it falls to silence. This mimics the \
2140 way many natural sounds work."),
2141 quiz::Quiz("Low-Frequency Oscillator (LFO)","A circuit that emits low-frequency elec\
2142 tronic waveforms below the audible level of human hearing (20 Hz or less). This low-\
2143 frequency waveform creates a rhythmic pulse that is used to modulate various paramet\
2144 ers in the audio signal, such as pitch or volume. LFOs are frequently used in sample\
2145 rs, synthesizers and signal processors to create such effects as vibrato, tremolo, a\
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2146 nd phasing."),
2147 quiz::Quiz("low-pass-filter","An audio filter or device that attenuates signals above a
2148 certain frequency (the cut-off frequency) and passes signals with frequencies that
2149 are lower than the cut-off."),
2150 quiz::Quiz("Lows or Low-End","Short for “low frequencies,” loosely referring to bass-
2151 frequency signals below 250 Hz. Usually meant in the context of “highs, mids and lows”
2152 in an audio signal."),
2153 quiz::Quiz("LPF","The low pass filter (LPF) design passes harmonics below its cutoff or
2154 corner frequency untouched, and reduces the level of lower harmonics depending on how
2155 far above the cutoff they are. In a 12dB/oct (decibel/octave) low pass filter, harmonics
2156 one octave above the cutoff frequency (in other words, double cutoff frequency) are
2157 reduced in level by 12 dB; harmonics two octaves above the cutoff (four times the
2158 frequency) are reduced by 24dB, and so forth. This is the most common type of filter
2159 used, as most natural sounds have stronger low harmonics and weaker high harmonics –
2160 especially as a note fades to silence."),
2161 quiz::Quiz("LPG","By strict definition, a low pass gate (LPG) is a low pass filter whose
2162 cutoff frequency goes down into the subsonic range as its control voltage goes towards
2163 0 volts, resulting in the input signal being filtered almost into silence. Some replicate
2164 this by combining a low pass filter and a voltage controlled amplifier into the same
2165 module, with both following the same control voltage. In either case, as an input envelope
2166 falls from a high level to 0 volts, the output gets duller (higher harmonics are filtered
2167 more) as it falls to silence. This mimics the way many natural sounds work."),
2168 quiz::Quiz("M2.5","A common screw thread size used to mount Eurorack modules. This size
2169 is most common when using a system of loose nuts that slide along the rails that the
2170 modules are attached to."),
2171 quiz::Quiz("M3","A common screw thread size used to mount Eurorack modules. This size
2172 is most common when using module mounting rails that have been pre-drilled."),
2173 quiz::Quiz("Magnetic Tape","Recording tape consisting of a plastic strip coated by magnetic
2174 materials, finely ground iron oxide (rust) particles. Commonly used for analog
2175 recording."),
2176 quiz::Quiz("Magnetism","A natural attractive energy of iron based-materials towards other
2177 iron-based materials."),
2178 quiz::Quiz("MArF","The rare Buchla Model 248 MArF (Multiple Arbitrary Function Generator)
2179 is a cross between a sequencer and an envelope generator (both described elsewhere in this
2180 glossary) in that it typically contains 16 or 32 stages (sometimes referred to as “segments”),
2181 and a rate control to interpolate between these stages. This means very complex envelope
2182 shapes and other control voltage sequences can be created. Later on, Buchla used the term
2183 MARF to describe the multi-step envelopes in instruments such as the Buchla 400."),
2184 quiz::Quiz("Margin","See “Headroom.”"),
2185 quiz::Quiz("Masking","The characteristic of hearing by which loud sounds prevent the ear
2186 from hearing softer sounds of similar frequency. Also refers to the obscuring of
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2189 f softer sounds by louder ones."),
2190 quiz::Quiz("Master","1) The main output control of a console or DAW, setting the lev\
2191 el of the mixed signal as it leaves the console. (Also called “master fader.”) 2) Th\
2192 e final-mixed original recording from which copies are made."),
2193 quiz::Quiz("Mastering","The final process of fine-tuning and “sweetening” the mix on\
2194 a song or collection of songs, from which the master will be created."),
2195 quiz::Quiz("Measure","The grouping of a number of beats in music. (See also “Bar.”)\
2196 ),
2197 quiz::Quiz("Meg","A slang abbreviation based on the prefix “Mega-, meaning 1,000,000\
2198 . Often used as shorthand for megahertz (1,000,000 Hertz, Mhz) or megabytes (1,000,0\
2199 00 bytes, MB)."),
2200 quiz::Quiz("Meter","1) A device that measures and displays the signal level in audio\
2201 or digital equipment. Meters usually measure peak values or RMS values. (See also “\
2202 Peak Value,””RMS Value.”) 2) The rhythmic structure of music, typically describing t\
2203 he number of beats in a measure."),
2204 quiz::Quiz("Mic / Line Switch","Mic, Mike – Abbreviations for “microphone.”"),
2205 quiz::Quiz("Microphone (Mic) Input","The input of a console or other device designat\
2206 ed for a microphone signal."),
2207 quiz::Quiz("Microphone (Mic) Level","The very low audio voltage level emitted by a s\
2208 tudio microphone. The signal must go through a preamplifier to be increased to line \
2209 level before entering the console. (See also “Line Level,” “Preamplifier.”)"),
2210 quiz::Quiz("Microphone (Mic) Pad","A setting on a microphone or preamp, or a separat\
2211 e adapter/connector, that reduces the level of the microphone signal before it enter\
2212 s the preamplifier to prevent overload."),
2213 quiz::Quiz("Microphone","A transducer which converts sound pressure waves into elect\
2214 rical signals."),
2215 quiz::Quiz("Mid-Side Miking (M/S)","(Abbreviated M/S) A stereo coincident microphone\
2216 placement technique in which one cardioid pattern microphone is aimed directly at t\
2217 he sound source, and a bi-directional microphone placed sideways and as close as pos\
2218 sible to the first mic."),
2219 quiz::Quiz("MIDI Clock","A clock signal conveyed by MIDI that is used by the connect\
2220 ed sequencers and musical devices to stay in sync with one another. Not to be confus\
2221 ed with MIDI time code (MTC), MIDI clock is tied to the Beats-Per-Minute (BPM) tempo\
2222 , advancing 24 steps per quarter note."),
2223 quiz::Quiz("MIDI Controller","Can refer to two different elements of MIDI, depending\
2224 on the context. 1) A device or software that sends MIDI data to connected devices, \
2225 either through pre-programmed sequencing or through live performance by a musician. \
2226 2) Any of a number of smaller controls on a MIDI device that is assigned to control \
2227 specific parameters of the sound or performance."),
2228 quiz::Quiz("MIDI Interface","A device that converts a MIDI signal into the digital f\
2229 ormat of a computer so it can store and use the MIDI signal."),
2230 quiz::Quiz("MIDI over Bluetooth","Bluetooth Low Energy (BLE) is a wireless connectio\
2231 n specification supported by the majority of mobile computing devices. BLE (also cal\
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2232 led Bluetooth SMART) can extend battery life for mobile devices using connected acce\
2233 ssories (such as MIDI keyboards and controllers) that don't continuously stream data\
2234 . An MMA Working Group evaluated Bluetooth LE MIDI performance (latency and jitter) \
2235 and decided on a specification for MIDI over Bluetooth which would enable products f\
2236 rom different manufacturers to interoperate. The Specification for MIDI over Bluetoo\
2237 th Low Energy (BLE-MIDI) is based on Apple's implementation which appeared in iOS8 a\
2238 nd OSX 10.10, so that products from early adopters would remain compatible with the \
2239 industry standard."),
2240 quiz::Quiz("MIDI Sample Dump Standard (SDS)", "A sub-protocol that was added into MID\
2241 I to enable the transfer of digitally recorded samples between instruments, storage \
2242 units or sound modules without converting them to analog."),
2243 quiz::Quiz("MIDI Sequencer", "A device or software that can record and play back MIDI\
2244 data, controlling the performance of MIDI musical instruments or devices in a serie\
2245 s of timed steps. MIDI sequencers can exist on board MIDI controllers, keyboards or \
2246 workstations, as standalone devices, or as computer software."),
2247 quiz::Quiz("MIDI Thru Box", "A unit with one MIDI In Port and several MIDI Thru Ports\
2248 to relay the MIDI signal to multiple devices. MIDI users often prefer this as an al\
2249 ternative to "daisy chaining" devices, which can cause slight delays in the MIDI sig\
2250 nal."),
2251 quiz::Quiz("MIDI Thru", "A port that puts out a MIDI signal that is the same as the i\
2252 ncoming MIDI signal, effectively relaying the signal to another device without alter\
2253 ing or changing it. (Many MIDI devices have three MIDI ports: In, Out and Thru.)"),
2254 quiz::Quiz("MIDI Time Code (MTC)", "The translation of the information in SMPTE time \
2255 code into MIDI data, enabling MIDI sequencers and connected devices to sync with SMT\
2256 PE code (usually in relation to video). (See also "SMPTE Time Code.")"),
2257 quiz::Quiz("MIDI", "Short for Musical Instrument Digital Interface. MIDI is a common \
2258 language to connect one synthesizer to another, and synthesizers to a computer. Alth\
2259 ough it is a digital language, it is easy to buy a MIDI to CV/Gate (control voltage \
2260 and gate) converter module that handles both note events and MIDI clocks for driving\
2261 sequencers and the such. The biggest thing to watch out for is what type of connect\
2262 or is required: the traditional 5-pin DIN, or a USB computer-style connection."),
2263 quiz::Quiz("Mids", "Abbreviation for "mid-range frequencies," the audio frequencies f\
2264 rom about 250 Hz through 6000 Hz. Meant in the context of "highs, mids and lows" in \
2265 an audio signal."),
2266 quiz::Quiz("Mini Keys", "A number of keyboard controllers and even keyboard synths us\
2267 e a key size that is much smaller than a typical piano key. Mini keys is the term co\
2268 mmonly used (sometimes derisively, although the space and cost savings can be quite \
2269 significant) to refer to this hardware choice."),
2270 quiz::Quiz("Mix Down", "Mixdown or Mix Down - The processes of creating a final mix b\
2271 y combining multiple audio tracks into a single track (or two-channel stereo track) \
2272 prior to the mastering stage. This can include the traditional method of mixing the \
2273 multiple channels of analog tape into a two-track master, or the more modern method \
2274 of creating a digital mixdown using a DAW (which is often referred to as "rendering")\
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2275 )."),
2276 quiz::Quiz("Mix","1) The blending of audio signals together into one composite signal.
2277 2) Can also refer to the blending of a portion of an effected audio signal back i\
2278 nto the direct signal."),
2279 quiz::Quiz("Mixer","This module combines signals together. You may use a mixer to co\
2280 mbine audio signals, in which case you may want one with exponential level controls \
2281 and perhaps stereo panning, or to combine control voltages, in which case you may wa\
2282 nt linear level controls plus additional functions to invert and offset the voltages\
2283 going through it."),
2284 quiz::Quiz("Modular","A modular synth breaks down the main components of a synthesiz\
2285 er – the tone-generating oscillators, the tone-modifying filters, the amplitude-shap\
2286 ing VCAs, and the modulation sources that create envelopes, tremolos, and more – int\
2287 o individual modules you can purchase and install. At the most basic level, this all\
2288 ows you to play mix-and-match in building your own custom synth."),
2289 quiz::Quiz("Modulation Noise","Noise that is present only when the audio signal is p\
2290 resent."),
2291 quiz::Quiz("Modulation","When you vary a parameter of a synthesizer module using vol\
2292 tage control, it is said that you're modulating that parameter. For example, when a \
2293 low frequency oscillator (LFO) varies the cutoff frequency of a filter to create a w\
2294 ah-wah effect, it is said that the LFO is modulating the cutoff. When an envelope ge\
2295 nerator causes a voltage controlled amplifier (VCA) to open up to allow a sound to b\
2296 ecome suddenly loud, and then fades it back down to silence, you can also say the en\
2297 velope is modulating the amp (although some like to restrict the term "modulate" to \
2298 a repetitive action). Therefore, we call the sources of these changes modulators."),
2299 quiz::Quiz("Modulator","We touched on the general subject of modulation and modulato\
2300 rs in the definition above. However, quite often when someone uses the term modulato\
2301 r, they're usually discussing a synthesis techniques where one usually audio-rate si\
2302 gnal "modulates" (varies) another audio signal. For example, in frequency modulation\
2303 (FM) synthesis, the modulator (or modulating oscillator) varies the frequency (pitc\
2304 h) of the main signal generator (oscillator), called the carrier. In ring, balanced,\
2305 or amplitude modulation, the modulator is varying the loudness of the carrier signa\
2306 l. So the term modulator is a way to make it clear which component you're talking ab\
2307 out in one of these patches: not the main tone generator, but the module that is dri\
2308 ving that generator crazy."),
2309 quiz::Quiz("Module","A self-contained group of circuits and controls. In the recordi\
2310 ng studio, modules are often contained in interchangeable housing for installation o\
2311 n rack mounts, and can include amplifiers, equalizers, effects processors and sound \
2312 modules (MIDI instruments to be activated by an external controller). In the digital\
2313 space, plug-ins, software synths, samplers and plug-ins are also described as modul\
2314 es."),
2315 quiz::Quiz("Monaural (Mono)","(Abbreviated "Mono") Describing an audio signal coming\
2316 through a single, as opposed to stereo, which is two channels. (See also "Monophoni\
2317 c.")"),
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2318 quiz::Quiz("Monitor Mix","A mix of the live and/or recorded audio signals that is fe\  
2319 d to the musicians so the can hear the music while performing, whether live onstage \  
2320 or in the studio. Monitor mixes are on a separate signal path from the main mix (oft\  
2321 en controlled by a separate, smaller console) and do not affect the FOH mix (in live\  
2322 audio) or the signal going into the multitrack recorder/DAW. In live performance se\  
2323 ttings, the monitor mix is often controlled by a separate audio engineer running a s\  
2324 eparate sound board."),  
2325 quiz::Quiz("Monitor Mixer Section","Monitor Section/Monitor Mixer Section – The sect\  
2326 ion of the console that is used to create a rough mix so the engineer can hear what \  
2327 is being recorded without effecting the levels being fed to the multitrack recorder \  
2328 or DAW."),  
2329 quiz::Quiz("Monitor Path","A signal path separate from the channel path that allows \  
2330 the engineer to listen to what is being recorded without affecting the signal being \  
2331 fed to the multitrack recorder or DAW. (See also “Channel Path.”)),  
2332 quiz::Quiz("Monitor","1) To listen to the music for the purpose of checking quality \  
2333 or avoiding peaks. 2) A speaker in the studio (usually one of a pair) that is used t\  
2334 o listen to the audio signals. This can include studio monitors in the control room \  
2335 for listening to the mix, and headphones in the booths or live room for the performe\  
2336 rs to hear a mix of the tracks while they are performing."),  
2337 quiz::Quiz("Monophonic","(Abbreviated “Mono”) 1) A single sound source or single-cha\  
2338 nnel transmission (as opposed to stereo). 2) A melody line in which only one note at\  
2339 a time is played. 3) Describing an instrument or synthesizer setting that only play\  
2340 s one pitch (or “voice”) at a time. (See also “Voice.”)),  
2341 quiz::Quiz("Morphing","In the context of a modular synth, morphing refers to an osci\  
2342 llator that can more or less smoothly change the shape of its output waveform – and \  
2343 therefore, the resulting sound – as you play it. This is usually the domain of digit\  
2344 al oscillators which internally crossfade (or in some cases, switch) from one wavesh\  
2345 ape to another, although it is sometimes applied to analog oscillators that give you\  
2346 real time control over waveshapes."),  
2347 quiz::Quiz("Mother-32","A very popular semi-modular synthesizer by Moog. It comes in\  
2348 its own case, but can be mounted in a Eurorack-format case. It comes with one VCO (\  
2349 sawtooth and pulse waveforms), one LFO (triangle and square waveforms), one Moog-sty\  
2350 le transistor ladder filter that can be low pass or high pass, and one AD or AR enve\  
2351 lope generator. It also has a very capable step sequencer plus a miniature one-octav\  
2352 e keyboard. What makes it a semi-modular is a nice patch panel that allows alternat\  
2353 e routings for the way the synth voice is internally wired, and for it to be patched \  
2354 to external modules. As so many of these were sold, I’m using it as a representative\  
2355 of a typical semi-modular or “starter” synthesizer voice when discussing how to exp\  
2356 and a basic modular system. I have an online introductory course to the Mother-32 co\  
2357 ming out this spring, and will have a course plus ongoing weekly series on adding di\  
2358 fferent modules to this starter system."),  
2359 quiz::Quiz("Moving Coil Microphone","A microphone in which sound pressure waves are \  
2360 converted to an electrical audio signal by an induction coil moving within a magneti\
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2361 c field—a process often compared to a loudspeaker working in reverse. Dynamic microp\
2362 hones are less sensitive than condenser microphones, but can be effective for miking\
2363 louder sound sources or for close-miking applications."),
2364 quiz::Quiz("Moving Fader Automation","A feature in some consoles in which fader chan\
2365 ges can be pre-programmed to occur automatically during playback of a multitrack rec\
2366 ording."),
2367 quiz::Quiz("MU","Refers to modules that are 5U (rack units) or 8.75" (22.2 cm) high,\
2368 which is most often associated with the vintage Moog standard and those who have fo\
2369 llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
2370 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
2371 fers to a standardized width of 2.125" (5.4 cm) wide per MU. Given that this standar\
2372 d is both historical and physically large, some users "5U" as a badge of honor that \
2373 they're traditional and cool. (And the are.) There was also a briefly popular 5U for\
2374 mat from MOTM that used a different width and power connection. It has since been di\
2375 scontinued, but there are still diehard MOTM format users today."),
2376 quiz::Quiz("Multi-Tap Delay","A delay works by in essence putting audio in one end o\
2377 f a pipe and grabbing it again when it comes out the other. A multi-tap delay says "\
2378 Why wait until the audio snapshots go all the way through the pipe? Let's grab it wh\
2379 en it's only part way through the pipe." Those points where it's prematurely grabbed\
2380 are the "taps" – kind of like additional water taps added along a long pipe."),
2381 quiz::Quiz("Multimeter","A small device that tests electrical voltage, current, and \
2382 resistance. Multimeters are useful in recording studios for calibrating electrical s\
2383 ystems and troubleshooting problems."),
2384 quiz::Quiz("Multiple Arbitrary Function Generator","The rare Buchla Model 248 MArF (\
2385 Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelop\
2386 e generator (both described elsewhere in this glossary) in that it typically contain\
2387 s 16 or 32 stages (sometimes referred to as "segments"), and a rate control to inter\
2388 polate between these stages. This means very complex envelope shapes and other contr\
2389 ol voltage sequences can be created. Later on, Buchla used the term MARF to describe\
2390 the multi-step envelopes in instruments such as the Buchla 400."),
2391 quiz::Quiz("Multiple","Quite often you need to split or copy a signal to send to mor\
2392 e than one destination. This is commonly done with a multiple ("mult" for short) whe\
2393 re you plug one source in, and then plug in additional patch cables to go off to mul\
2394 tiple destinations."),
2395 quiz::Quiz("Multiplexer","Multiplexing is a technical way to describe signal routing\
2396 , where multiple signals may be routed to one destination. In synth modules, this is\
2397 usually extended to include the possibility of one input being switched between mult\
2398 iple outputs. A sequential switch is a type of multiplexor, as it chooses among mult\
2399 iple inputs to decide which one to send to the output (or the other way around). The\
2400 re are some modules that do this at audio rate, using an oscillator's output to swit\
2401 ch between variations of another waveshape to create complex, chopped mixtures of th\
2402 ose waveforms."),
2403 quiz::Quiz("Multitimbral","Refers to the ability of a synthesizer or module to play \
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2404 several different sounds, patches or “timbres” at once.”),
2405 quiz::Quiz("Multitrack Recording", "Also called tracking or multitracking) The heartb\
2406 eat of the recording studio, multitrack recording is process of recording a collecti\
2407 ve of sound sources onto separate tracks, each with its own audio channel, then comb\
2408 ining the tracks to play back simultaneously. Recording can be done either one track\
2409 or instrument at a time (to be combined later) or by recording the performers onto \
2410 separate tracks as they play together live. These signals were originally recorded o\
2411 nto multitrack analog tape, but today they can also be recorded digitally as separat\
2412 e audio files into a digital audio workstation (DAW)."),
2413 quiz::Quiz("Multitrack Tape", "A piece/reel of magnetic tape which can be used to sto\
2414 re two or more discrete signals in sync with each other."),
2415 quiz::Quiz("Musical Instrument Digital Interface (MIDI)", "Short for Musical Instrume\
2416 nt Digital Interface. MIDI is a common language to connect one synthesizer to anothe\
2417 r, and synthesizers to a computer. Although it is a digital language, it is easy to \
2418 buy a MIDI to CV/Gate (control voltage and gate) converter module that handles both \
2419 note events and MIDI clocks for driving sequencers and the such. The biggest thing t\
2420 o watch out for is what type of connector is required: the traditional 5-pin DIN, or\
2421 a USB computer-style connection."),
2422 quiz::Quiz("Mute Switch", "A switch on a console or other piece of audio equipment th\
2423 at turns off the input or output, or a matching button on the virtual audio control \
2424 space of a DAW. The individual channels on a console each have a mute switch that ca\
2425 n cut the signal for that channel."),
2426 quiz::Quiz("Mute", "Sometimes you need to silence or disconnect a signal. A circuit t\
2427 hat allows you to do so is called a mute."),
2428 quiz::Quiz("Nanowebers per Meter (NW/m)", "The standard unit in measuring the amount \
2429 of magnetic strength on analog tape. A Weber is a unit of magnetic strength, but it \
2430 is too large a unit to apply to the magnetism in tape recorders, so nanowebers is us\
2431 ed instead. Nanowebers per meter of tape effectively describes the signal strength t\
2432 hat is being recorded to tape."),
2433 quiz::Quiz("Narrowband Noise", "Noise (random energy) that occurs over a limited freq\
2434 uency range."),
2435 quiz::Quiz("Near Field", "The area between 1-5 feet from the sound source. Studio mon\
2436 itors are generally considered “near-field” speakers because they are meant to be li\
2437 stened to at close range. (See also “Far Field.”)"),
2438 quiz::Quiz("Near-Coincident Miking", "A stereo miking technique in which two micropho\
2439 nes are placed near each other at an outward angle to create a stereo image (as oppo\
2440 sed to “Coincident Miking” which angles the microphones toward each other). Common \
2441 versions of near-coincident miking include DIN stereo (90-degree angle, 20cm apart),\
2442 NOS stereo (90-degree angle, 30 cm apart) and ORTF (110-degree angle, 17 cm apart).\
2443 "),
2444 quiz::Quiz("Negative Feedback", "A portion of the output signal that is fed back to t\
2445 he input of an amplifier with its phase inverted from the original output signal. Th\
2446 is has a dampening effect on the output, effectively cancelling out a portion of the\
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2447     volume."),
2448     quiz::Quiz("Noise Floor","The level of the noise present below the audio signal, mea\
2449     sured in dB. Every electronic device emits a minimum level of noise, even when no au\
2450     dio is traveling through it; this is described as its noise floor. Generally speakin\
2451     g, the lower the noise floor in these devices, the higher the quality of the device.\
2452     The noise floor also translates to the recorded signal; the noise floor of a record\
2453     ing is the sum of all the noise generated by connected devices. The objective is alw\
2454     ays to keep the noise floor as low as possible."),
2455     quiz::Quiz("Noise Gate","A gate that is used reduce audible noise by automatically t\
2456     urning off an audio channel when the signal is not present."),
2457     quiz::Quiz("Noise Reduction","Any of a number of processes to remove noise from a si\
2458     gnal, device or system."),
2459     quiz::Quiz("Noise","Describes any unpleasant, objectionable or unintended sound freq\
2460     uencies present in the audio signal. All electronic equipment produces some type of \
2461     noise, which may be described as a hiss or buzz that can be heard during quiet or ot\
2462     herwise silent passages. (See also "Noise Floor.") Bad connections, improper groundi\
2463     ng, radio interference and other issues can also cause introduce noise into the sign\
2464     al. Engineers may also deliberately run a noise signal through a sound system for te\
2465     sting purposes. (See also "White Noise, "Pink Noise.")"),
2466     quiz::Quiz("Non-destructive Editing","A feature in recording systems (most common in\
2467     Digital Audio Workstations, or DAWs) in which the original signal or content stays \
2468     intact while edits are performed, allowing the engineer to revert to the original ve\
2469     rsion at any time. (Sometimes also called "Nonlinear editing.")"),
2470     quiz::Quiz("Nondirectional","In microphones, picking up evenly from all directions."),
2471     ),
2472     quiz::Quiz("Normalize","To apply a fixed amount of gain to audio so that the highest\
2473     peak is set at the highest acceptable recording level."),
2474     quiz::Quiz("Normalled","The power of modular synthesizers is that you can patch a si\
2475     gnal to flow the way you prefer through your system. This can also be a time-consumi\
2476     ng bummer when you're just trying to patch a "typical" signal flow. Therefore, some \
2477     manufacturers have created "semi-modular" synths that have all of these typical conn\
2478     ections pre-wired for you, with the important feature that many of these wirings can\
2479     be overridden by inserting patch cables into the correct jacks. These pre-wired con\
2480     nections are often referred to as being normalled. For example: An internal noise so\
2481     urce may normally be connected to one channel of a mixer that appears before the fil\
2482     ter, but if you insert a patch cable into a jack usually labeled external input, thi\
2483     s "normalled" connection is broken and replaced by your external connection."),
2484     quiz::Quiz("Notch Filter","This is a particular type of filter mode where audio freq\
2485     uencies or harmonics around the corner or cutoff frequency setting are removed, nor \
2486     "notched out" of the overall spectrum. It is the opposite of a bandpass filter, whic\
2487     h only passes harmonics around the cutoff frequency. Notch filters tend to have a su\
2488     btle effect on the sound; moving (modulating) the cutoff frequency can result in a w\
2489     eak phasing sort of sound. Notch filters are often used in sound systems to weaken o\
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2490 r remove a problematic frequency, such as ground loop hum, a resonance in a room, or\  
2491 other annoying peak in the harmonic spectrum of a sound. Think of using a notch fil\  
2492 ter in a patch to hollow out a sound, leaving room in the harmonic spectrum for othe\  
2493 r sounds to exist with less competition, or just to create a sound more likely to ca\  
2494 tch the ear because something that is expected is instead missing."),  
2495 quiz::Quiz("Notch","A narrow band of audio frequencies."),  
2496 quiz::Quiz("NW/m","The standard unit in measuring the amount of magnetic strength on\  
2497 analog tape. A Weber is a unit of magnetic strength, but it is too large a unit to \  
2498 apply to the magnetism in tape recorders, so nanowebers is used instead. Nanowebers \  
2499 per meter of tape effectively describes the signal strength that is being recorded t\  
2500 o tape."),  
2501 quiz::Quiz("Nybble","Nybble (or Nibble) – One half byte of computer data, or 4 bits.\\  
2502 "),  
2503 quiz::Quiz("Nyquist Frequency","In digital recording, the highest frequency that can\  
2504 be recorded and reproduced properly, equivalent to a one-half the sampling rate. (F\  
2505 or example, with the common sampling rate of 44,100 kHz per second, the Nyquist freq\  
2506 uency would be 22,050 kHz.) Aliasing begins to occur with frequencies that exceed th\  
2507 is threshold. (See also “Aliasing.”)"),  
2508 quiz::Quiz("Nyquist Rate","he lowest sampling rate that can be used to record and re\  
2509 produce a given audio signal, equivalent to twice the highest frequency. If the high\  
2510 est frequency found in an analog signal or sound is 18,000 kHz, theoretically the si\  
2511 gnal must be sampled at a minimum of 36,000 kHz per second—otherwise, the signal is \  
2512 considered to be undersampled and aliasing will occur. This is essentially the inver\  
2513 se principle of the Nyquist Frequency. (NOTE: the sample rate of 44,100 kHz/second i\  
2514 s considered the standard sample rate because it easily covers the upper range of hu\  
2515 man hearing, which is about 20,000 kHz.)"),  
2516 quiz::Quiz("Octave Divider","A module that creates a new tone one or two octaves bel\  
2517 ow the fundamental harmonic – the “pitch” – of the sound coming into it, to emphasiz\  
2518 e the bass. Sometimes also known as a suboctave or sub bass function."),  
2519 quiz::Quiz("Octave","An octave is a typical musical interval. For example, all of th\  
2520 e “C” notes on a keyboard are octaves apart from each other. To play a note that is \  
2521 one octave higher in tuning, you need to double its pitch; to play an octave lower, \  
2522 you need to cut the pitch in half. In patch terms, this typically means adding or su\  
2523 btracting 1 volt to get a one octave change in pitch; some oscillators also have oct\  
2524 ave switches on their front panels that add or subtract these voltages for you (all \  
2525 they are not always perfectly accurate; you often need to re-tune after switching oc\  
2526 taves). Suboctave or subharmonic generators divide the input pitch by 2 or 4 to crea\  
2527 te new waveforms that are one or two octaves lower in pitch, which adds bass."),  
2528 quiz::Quiz("Off Axis","Veering away from the imaginary line (axis) directly in front\  
2529 of the receiving end of a microphone. Measured as degrees of an angle. (For example\  
2530 , a sound coming from directly behind the microphone is said to be 180 degrees off-a\  
2531 xis.)"),  
2532 quiz::Quiz("Offset Time","1) The SMPTE time that will trigger a MIDI sequencer to be\  

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2533   gin. 2) The amount of position difference needed to get two reels to play the music \
2534   in time."),
2535   quiz::Quiz("Offset","In simple terms, Offset modules usually add or subtract a volta\
2536   ge from a signal passing through - such as shifting a 0 to +10v signal to instead va\
2537   ry between -5 and +5 volts."),
2538   quiz::Quiz("Ohm's Law","The mathematical relationship between voltage, current and r\
2539   esistance."),
2540   quiz::Quiz("Ohm","The unit used to measure the amount of opposition (impedance) to e\
2541   lectrical current flow in a signal or device. (See also "Impedance.")"),
2542   quiz::Quiz("Omni Mode","A setting that enables a MIDI device to recognize and respon\
2543   d to all MIDI channels at once."),
2544   quiz::Quiz("Omni","A prefix meaning "all.""),
2545   quiz::Quiz("Omnidirectional Pattern","In microphones, picking up evenly from all dir\
2546   ections (sometimes also called "Nondirectional"). 2) In speakers, sending out the si\
2547   gnal evenly in all directions."),
2548   quiz::Quiz("On Axis","The position directly in front of the diaphragm of a microphon\
2549   e, in line with its movement."),
2550   quiz::Quiz("Open Circuit","An electrical circuit that is disconnected, interrupted o\
2551   r incomplete, preventing the flow of electricity."),
2552   quiz::Quiz("Operating Level","(Sometimes called "Reference Level") The maximum level\
2553   that should not be exceeded in normal operation."),
2554   quiz::Quiz("Operational Amplifier","(Abbreviated "Op Amp") An amplifying circuit use\
2555   d in most audio and electronic devices."),
2556   quiz::Quiz("Operational Transconductance Amplifier","An OTA (operational transconduc\
2557   tance amplifier) circuit is one that converts an input voltage to an output current.\
2558   This is a popular amplifier design as it can be less prone to going into saturation\
2559   (clipping), has good bandwidth, and is also known for a "warm" sound. Therefore, yo\
2560   u may find it in VCAs (voltage controlled amplifiers). Current can be thought of as \
2561   the inverse of resistance, so what you have in an OTA circuit is in essence a voltag\
2562   e to resistance device that makes it possible to add voltage control to circuits suc\
2563   h as filters. In general, when someone touts they have an OTA based filter, they usu\
2564   ally mean it has a "warm" sound...unless it's an MS-20 filter clone, in which case i\
2565   t's thinner and more edgy. In reality, using an OTA is more about convenience of des\
2566   ign than creating a specific sound."),
2567   quiz::Quiz("Operator","There are a few different synthesis techniques where one usua\
2568   lly audio-rate signal does something to another audio signal. For example, in freque\
2569   ncy modulation (FM), a second signal (called the modulator) varies the frequency (pi\
2570   tch) of the main signal, called the carrier. These two signals or oscillators are of\
2571   ten referred to as operators, particularly in FM patches. You're more likely to hear\
2572   this term used when working with a dedicated FM synthesizer like a Yamaha DX-7 and \
2573   its descendants, than with a modular system."),
2574   quiz::Quiz("OR function","One of the most common Boolean or binary logic functions, \
2575   OR says if there is a gate on (or "high") signal at any of the inputs (i.e. input 1 \
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2576 or input 2 or input 3, etc.), to output a gate on signal. A NOR function has an inve\
2577 rted output: it would only be on (high) if all inputs were low (not input 1 nor input
2578 2 are high). An XOR (Exclusive OR) would only output a high signal if one of the i\
2579 nputs was high, but not if both inputs were high (or low). Finally, an XNOR is the i\
2580 nvert of an XOR function."),
2581 quiz::Quiz("Oscillator", "At its core, to oscillate means to vary back and forth in a\
2582 repeating pattern. The main sound generator in a modular system is called an oscill\
2583 ator because its output varies up and down (oscillates) in voltage in a repeating pa\
2584 ttern. This pattern is referred to as its waveshape (such as a square wave, that alt\
2585 ernates between high and low voltages); how fast this pattern repeats is called its \
2586 frequency or pitch. An acoustic instrument equivalent of an oscillator is a string t\
2587 hat vibrates back and forth on a guitar, a drum head that vibrates up and down, or t\
2588 he vibrations in the reed of a woodwind instrument. The vibrations of a modular synt\
2589 h's oscillator just happen with electricity going down a wire rather than a physical\
2590 object vibrating in air. (Eventually this electricity is routed to a speaker, which\
2591 then vibrates the air with the same pattern sent to it over a wire.)"),
2592 quiz::Quiz("Oscilloscope", "This is a piece of test equipment that displays voltage fl\
2593 uctuations as graphical waveforms. A 'scope can run at a wide range of frequencies,\
2594 displaying slowly changing voltages like LFOs or envelopes, or quickly changing vol\
2595 tages like oscillators and noise. Oscilloscopes used to be bulky pieces of external \
2596 equipment, but now you can get USB scopes that offload the display portion of the jo\
2597 b to your computer, or scopes as modules."),
2598 quiz::Quiz("OTA", "An OTA (operational transconductance amplifier) circuit is one tha\
2599 t converts an input voltage to an output current. This is a popular amplifier design\
2600 as it can be less prone to going into saturation (clipping), has good bandwidth, an\
2601 d is also known for a "warm" sound. Therefore, you may find it in VCAs (voltage cont\
2602 rolled amplifiers). Current can be thought of as the inverse of resistance, so what \
2603 you have in an OTA circuit is in essence a voltage to resistance device that makes i\
2604 t possible to add voltage control to circuits such as filters. In general, when some\
2605 one touts they have an OTA based filter, they usually mean it has a "warm" sound...u\
2606 nless it's an MS-20 filter clone, in which case it's thinner and more edgy. In reali\
2607 ty, using an OTA is more about convenience of design than creating a specific sound.\
2608 "),
2609 quiz::Quiz("Out of Phase", "1) Being similar to another signal in amplitude, frequenc\
2610 y and wave shape but being offset in time by part of a cycle. 2) Having the opposite\
2611 polarity."),
2612 quiz::Quiz("Outboard Equipment", "Equipment that is used with, but is not a part of, \
2613 a console."),
2614 quiz::Quiz("Output Impedance", "The opposition to the flow of electrical current by t\
2615 he output circuits of an amplifier (or other device)."),
2616 quiz::Quiz("Output Level", "The signal level at the output of a device."),
2617 quiz::Quiz("Output", "1) The jack or physical location of where a device sends out a \
2618 signal. 2) The signal put out by a device."),
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2619 quiz::Quiz("Overdubbing", "The process of recording an additional musical performance\  
2620 over an existing recording, usually on its own track. Overdubbing has become a comm\  
2621 on recording technique with the advent of multitrack recording, first on multitrack \  
2622 analog tape, and more recently via computers and Digital Audio Workstations (DAWs).\  
2623 ),  
2624 quiz::Quiz("Overtone", "Any harmonic in a tone except the fundamental frequency. (See\  
2625 also "Partial.")"),  
2626 quiz::Quiz("Pad", "1) A device or circuit that attenuates an incoming signal, usually\  
2627 to prevent overload of an amplifier that follows along the signal path. (Also somet\  
2628 imes called "Attenuator pad.") 2) A device with a surface that can be hit by a drum \  
2629 stick; hitting the pad produces an output signal pulse (or MIDI command) that causes\  
2630 a drum machine or synthesizer to sound a drum sound. 3) A type of synthesizer patch\  
2631 /program used to create sustained background or atmospheric sounds."),  
2632 quiz::Quiz("Pan (Panning)", "The process of "placing" a particular sound within the s\  
2633 tereo field. This is accomplished by controlling the balance of the signal between t\  
2634 he left and right speakers so the ear hears the sound as coming from a particular po\  
2635 int in the sonic space between left and right. This sonic space is sometimes called \  
2636 the "stereo panorama," from which the word "panning" is derived. In surround sound, \  
2637 panning occurs in a 360° sound space, not just left-right."),  
2638 quiz::Quiz("Panpot (or Pan Pot)", "Short for "Panoramic Potentiometer," a panpot is a\  
2639 knob in the channel strip that controls the panning of the audio signal in the ster\  
2640 eo (or surround) space by controlling how much of the signal is sent to each speaker\  
2641 or channel."),  
2642 quiz::Quiz("Parallel Jacks", "Several jacks that are wired so that each connection is\  
2643 wired to the corresponding connection of other jacks."),  
2644 quiz::Quiz("Parallel Port", "A connector that is able to transmit and receive digital\  
2645 data at the same time though different pins."),  
2646 quiz::Quiz("Parameter", "Parameter is the fancy name given to any value or property o\  
2647 r control of a synthesizer module that you're trying to change. For example, an osci\  
2648 llator's parameters typically include its pitch and the width of its pulse wave. A f\  
2649 ilter's parameter will include its cutoff frequency (pitch), the amount of resonance\  
2650 (feedback), and possibly other controls such as a blend between its different outpu\  
2651 ts. Parameter was a popular term to describe a value you could change in software, a\  
2652 nd it's been carried over by some to hardware modular synths."),  
2653 quiz::Quiz("Parametric Equalization", "An equalizer in which all parameters of equali\  
2654 zation can be adjusted to any amount, including the center frequency, the amount of \  
2655 boost or cut, and the bandwidth."),  
2656 quiz::Quiz("Paraphonic", "A paraphonic synth is one where all of the notes being play\  
2657 ed go through a single filter (VCF) and amplifier (VCA). This was a popular scheme i\  
2658 n the early days of polyphonic synths in that a separate oscillator (or organ-like f\  
2659 requency divider, in the case of "string synths" and the such) was used for each not\  
2660 e played, but they were mixed before all going to the filter and amp to articulate t\  
2661 he note(s). It was not uncommon for some monophonic synths to allow two to four inde\
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2662 pendent notes to independently control the pitch of its oscillators, while still goi\
2663 ng through a single filter. This works great for chords; it doesn't always work all \
2664 that great for when a new note is played while others are being held as all of the n\
2665 otes will be re-articulated together."),
2666 quiz::Quiz("Partial","1) Another word for overtone. 2) One of a number of sine waves\
2667 that makes up a complex sound, helping to define the timbre. This concept is a key \
2668 part of creating sounds in synthesizers: in additive synthesis, a number of partials\
2669 are combined to create a certain tone."),
2670 quiz::Quiz("Pass Band","The frequency range of signals that will be "passed" by a fi\
2671 lter, rather than reduced."),
2672 quiz::Quiz("Passive Device","A component that does not generate or control electrica\
2673 l current (as opposed to an "Active Device"). In audio applications, this usually re\
2674 fers to a piece of gear that does not include an amplifier as part of its design. Fo\
2675 r example, active speakers are self-powered, while passive speakers require an exter\
2676 nal amplifier in order to reproduce sound. (See also "Active Device.")"),
2677 quiz::Quiz("Passive","Means no active (i.e. connected to a power supply) electronics\
2678 are involved - such as sending a signal straight through a potentiometer control, i\
2679 nstead of using op amps and other electronics to create a mixer circuit around it. P\
2680 assive is cheap and easy, and does not add noise to a signal. But passive electronic\
2681 s cannot buffer one signal from another (meaning they might interact in undesirable \
2682 ways), and cannot boost, offset, or invert a signal."),
2683 quiz::Quiz("Patch Bay","Patch Bay (or Patchbay, Patch Field, Patch Panel) - A panel \
2684 or component containing a series of jacks with connections for most of the inputs an\
2685 d outputs of the console and components in the studio, used for the purpose of organ\
2686 izing, managing and regulating signal flow."),
2687 quiz::Quiz("Patch Cable","The cables used to connect together the different inputs a\
2688 nd outputs in a modular synthesizer, carrying electrical control voltages and audio.\
2689 The term came from the old telephone patch boards where an operator had to physical\
2690 ly connect two callers together using electrical cables. As different modular format\
2691 s often use different connector standards, you need to make sure the connectors at t\
2692 he ends of the wire in a patch cord are the size you need (3.5mm for Eurorack, 1/4" \
2693 for 5U/Moog Unit, or banana for Serge or Buchla control voltages)."),
2694 quiz::Quiz("Patch Cord (or Patch Cable)","An insulated cable with plugs on each end \
2695 used to route audio signals. Patch cords are typically thought of as short cables us\
2696 ed to make connections in the patch bay (hence the name); however, patch cords facil\
2697 itate almost any kind of audio connection between devices, can come in a wide range \
2698 of lengths, and can include a number of different types of connectors."),
2699 quiz::Quiz("Patch Field","A panel or component containing a series of jacks with con\
2700 nnections for most of the inputs and outputs of the console and components in the stu\
2701 dio, used for the purpose of organizing, managing and regulating signal flow."),
2702 quiz::Quiz("Patch Librarian","A computer program allowing for the storing of sound p\
2703 atches outside of a synthesizer via MIDI."),
2704 quiz::Quiz("Patch Panel","A panel or component containing a series of jacks with con\
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2705 nections for most of the inputs and outputs of the console and components in the stu\
2706 dio, used for the purpose of organizing, managing and regulating signal flow."),
2707 quiz::Quiz("Patch","The shorthand term used to refer how a series of modules are int\
2708 erconnected to create a sound, derived from the fact that patch cords are used to co\
2709 nnect the modules together. 1) To route or reroute the signal in an audio system (su\
2710 ch as a console) by using short cables with plugs inserted into jacks. 2) A sound se\
2711 tting or program on a synthesizer."),
2712 quiz::Quiz("Path","Short for Signal Path, the way in which current does or may trave\
2713 l in a circuit or through a device."),
2714 quiz::Quiz("PCM","Pulse Code Modulation - A process by which analog signals are tran\
2715 slated to digital code. This is done by taking samples of the amplitude of the analo\
2716 g signal at regular rapid intervals, then translating it into binary numbers as a di\
2717 gital representation of the original signal. The faster the sample rate, the better \
2718 the digital reproduction. PCM is the most common form of A/D conversion in digital a\
2719 udio."),
2720 quiz::Quiz("PD","Phase Distortion synthesis was used by Casio originally in the 80s \
2721 in the CZ line of synths. It is related to FM (frequency modulation), with enough di\
2722 fferences to avoid problems with the patent used by Yamaha's FM synths of the era. I\
2723 ntriguingly, it did a good job at mimicking many "analog" synth effects including th\
2724 e sound of a resonant filter."),
2725 quiz::Quiz("Peak Filter","An EQ circuit/filter that boosts or cuts the middle (cente\
2726 r frequencies in an audio signal, as opposed to high-pass or low-pass filters. (NOT \
2727 to be confused with amplitude peaks.)"),
2728 quiz::Quiz("Peak Meter","A meter which detects the absolute peak value of a waveform\
2729 , as opposed to the RMS value. (See also "Peak Value," "Root-Mean-Square," "RMS Mete\
2730 r.")"),
2731 quiz::Quiz("Peak to Peak Value","The measure of the total amplitude between positive\
2732 and negative peaks in an audio signal. Equal to twice the peak value for a sine wav\
2733 e. (See also "Peak Value.")"),
2734 quiz::Quiz("Peak Value","Peak Value (also called Peak Level) - The measure of the max\
2735 imum positive or negative value (amplitude) of a waveform at any moment. In audio, t\
2736 his is visually depicted as the farthest point of the waveform above or below the ze\
2737 ro axis."),
2738 quiz::Quiz("Pedal Board","A board with several guitar pedals attached and inter-conn\
2739 ected so that a guitar player can conveniently activate a number of different effect\
2740 s."),
2741 quiz::Quiz("Phantom Power","A system used to supply DC voltage to condenser mics and\
2742 other components through the audio cables, eliminating the need for external power \
2743 supplies."),
2744 quiz::Quiz("Phase Addition","The increased audio energy that happens when waveforms \
2745 are in similar phase relationships, resulting in an increase in volume up to twice w\
2746 hat it should be."),
2747 quiz::Quiz("Phase Cancellation","The opposite of phase addition, this is the reducti\
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2748 on of energy that occurs when two similar waveforms that are out of phase with one a\  
2749 nother and begin cancelling each other out, either greatly reducing or eliminating t\  
2750 he volume. When two identical wave forms are completely out of phase (by 180 degrees\  
2751 ), the result in theory is a total silencing or cancellation of the signal."),  
2752 quiz::Quiz("Phase Distortion Synthesis","Phase Distortion synthesis was used by Casi\  
2753 o originally in the 80s in the CZ line of synths. It is related to FM (frequency mod\  
2754 ulation), with enough differences to avoid problems with the patent used by Yamaha's\  
2755 FM synths of the era. Intriguingly, it did a good job at mimicking many "analog" sy\  
2756 nth effects including the sound of a resonant filter."),  
2757 quiz::Quiz("Phase Distortion","A change in the sound because of a phase shift in the\  
2758 signal. Sometimes used in synthesizers as a method of altering the wave shape or ad\  
2759 ding harmonics to the sound."),  
2760 quiz::Quiz("Phase Lock","Any of a number of processes used to help synchronize signa\  
2761 ls or devices by correcting phase differences. For example, in analog tape machines,\  
2762 phase locking helps to keep multiple machines synced together by sensing phase diff\  
2763 erences in the playback of pilot tones by the two machines and adjusting the speed t\  
2764 o eliminate the phase difference. In synthesizers, phase locking controls one tone g\  
2765 enerator so that it begins its waveform in phase with the signal from another tone g\  
2766 enerator. Phase-locked loops (PLL) are reference signals used in the clock functions\  
2767 of electronic devices."),  
2768 quiz::Quiz("Phase Locked Loop","A phase locked loop is, in essence, an oscillator th\  
2769 at tries to match the frequency of - or more importantly, a division or multiple of \  
2770 the frequency of - another signal. This is most commonly used to create a frequency \  
2771 that is much higher than the incoming reference signal - such as a timing module tha\  
2772 t can create an output clock that is 2, 4, 8, or more times the tempo of an incoming\  
2773 clock, or a very high frequency oscillator that is locked to a multiple of an incom\  
2774 ing pitch - perhaps to drive a special circuit such as a switched-capacitor filter.")\  
2775 ),  
2776 quiz::Quiz("Phase Modulation","Some would say this is the pedantically correct term \  
2777 for frequency modulation (FM), as the act of causing a carrier oscillator to play ba\  
2778 ck faster and slower (quickly changing its frequency to be higher and lower) is the \  
2779 same as advancing and retarding position (phase) of the normal playback of a wavefor\  
2780 m. But don't get bogged down by terminology when creating an FM patch; just connect \  
2781 the output of one oscillator to the pitch input of another and go for it."),  
2782 quiz::Quiz("Phase Reversal","A change in a circuit to get the waveform to shift by 1\  
2783 80 degrees."),  
2784 quiz::Quiz("Phase Shift","A delay introduced into an audio signal measured in degree\  
2785 s delayed."),  
2786 quiz::Quiz("Phase Shifter","This effect splits a signal into two copies. One copy is\  
2787 fed through an "all pass filter" which does not attenuate any of the original harmo\  
2788 nics like a low pass or high pass filter does, but which does alter the phase of the\  
2789 signal, causing those harmonics to have varying amounts of phase shift in relation \  
2790 to the original depending on their frequency. Mix these two copies back together, an\  

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2791 d different harmonic components of the original sound cancel each other out (see Pha\
2792 se), resulting in a notch filter effect. Each “stage” – all-pass filter section – of\
2793 a phase shifter creates one of these notches. More stages create more notches, and \
2794 a deeper effect.”),
2795 quiz::Quiz("Phase-Locked Loop","PLL - Any of a number of processes used to help syn\
2796 chronize signals or devices by correcting phase differences. For example, in analog \
2797 tape machines, phase locking helps to keep multiple machines synced together by sens\
2798 ing phase differences in the playback of pilot tones by the two machines and adjusti\
2799 ng the speed to eliminate the phase difference. In synthesizers, phase locking contr\
2800 ols one tone generator so that it begins its waveform in phase with the signal from \
2801 another tone generator. Phase-locked loops (PLL) are reference signals used in the c\
2802 lock functions of electronic devices.”),
2803 quiz::Quiz("Phase","A measurement (expressed in degrees) of the time difference betw\
2804 een two similar waveforms. One cycle of a waveform is considered to have 360 degrees\
2805 , just like a circle. How far you move around the circle (or through the waveform) c\
2806 an be defined by the phase. For example, if you are one-quarter of the way through a\
2807 waveform’s cycle, your phase is 90°.”),
2808 quiz::Quiz("Phasing","An effects sound created by varying the phase shift of an audi\
2809 o signal, then mixing it with the direct signal.”),
2810 quiz::Quiz("Phon","A unit of apparent loudness, numerically equal to the same number\
2811 of dB as a tone playing at 1000 Hz. For example, a sound is said to be 60 phon if i\
2812 t is perceived to be as loud as a 1000-Hz tone playing at 60dB.”),
2813 quiz::Quiz("Phone Plug","A plug (or its mating jack) with a diameter of 1/4 inch and\
2814 a length of 1 1/4 inches used for interconnecting audio.”),
2815 quiz::Quiz("Phono Plug","A common audio connector found on most stereo systems with \
2816 a center pin as one connection and an outer shell as the second connection.”),
2817 quiz::Quiz("Physical Modeling","One approach to (often digital) synthesis is to recr\
2818 eate the components of actual instruments – such as a vibrating string or tube, or a\
2819 resonating body such as the shell of a guitar or drum – and string those together t\
2820 o create sounds. There are a handful of modules available which perform this modelin\
2821 g to create their sounds.”),
2822 quiz::Quiz("Pickup Pattern","The shape of the area in front of or around the microph\
2823 one from where it evenly picks up sound. Many use this term interchangeably with “po\
2824 lar pattern,” but a polar pattern gives more detail about microphone sensitivity. (S\
2825 ee also “Polar Pattern.”),
2826 quiz::Quiz("Pickup","1) A device on an electric guitar or other instrument that puts\
2827 out an audio signal according to the string motion on the instrument. 2) See “Conta\
2828 ct Microphone.””),
2829 quiz::Quiz("Pinch Roller","A rubber (or plastic) wheel on a tape recorder that pinch\
2830 es the tape between it and the capstan, allowing the capstan to pull the tape.”),
2831 quiz::Quiz("Ping-Ponging (Bouncing)","The technique of combining and mixing multiple\
2832 tracks onto one or two tracks (mono or stereo). This can be done in real-time or an\
2833 alog by playing the tracks through the console and recording them onto separate trac\
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2834 ks, or digitally through a digital audio workstation. Bouncing was once used frequen\
2835 tly by engineers to free up additional tracks for recording, but in digital workstat\
2836 ions where tracks are virtually unlimited, this practice is basically obsolete. Toda\
2837 y, engineers typically bounce tracks for the purpose of creating a preliminary or fi\
2838 nal mix of a song."),
2839 quiz::Quiz("Pink Noise","A noise signal similar to white noise, containing all audib\
2840 le frequencies, but with equal energy per octave as opposed to all frequency bands. \
2841 Engineers frequently use pink noise as a tool to tune and calibrate audio equipment.\
2842 (See also "White Noise.") Noise is a random, unpitched signal that, at audio rates,\
2843 can sound like hissing or the wind. Pink noise has equal energy (sound level) per o\
2844 ctave. As each higher octave has double the frequency of the octave below it which s\
2845 preads out the energy over a wider range of frequencies, pink noise tends have a mor\
2846 e natural, less electronic sound with more bass and less high end – especially when \
2847 compared to white noise, which has an equal energy per number of hertz (frequency) a\
2848 nd therefore tends to sound very bright."),
2849 quiz::Quiz("Pitch Bend","A mechanism on a synth, keyboard or controller that can cau\
2850 se the pitch of the note to move up or down by a small amount."),
2851 quiz::Quiz("Pitch to Voltage Converter","A device that detects the frequency of an a\
2852 udio waveform and changes it into a control voltage, which is in turn fed to an osci\
2853 llator that produces a pitch at the same frequency."),
2854 quiz::Quiz("Pitch-to-MIDI Converter","A device that detects pitch in an analog audio\
2855 signal and translates it into MIDI information. (Also called "Audio-to-MIDI-Convert\
2856 er.")"),
2857 quiz::Quiz("Pitch-to-Voltage Converter","A device that detects the frequency of an a\
2858 udio waveform and changes it into a control voltage, which is in turn fed to an osci\
2859 llator that produces a pitch at the same frequency."),
2860 quiz::Quiz("pitch","1) The perception of frequency by the ear (a higher or lower ton\
2861 e of music). 2) A control on a tape transport which adjusts the speed slightly up or\
2862 down, changing the pitch and time of the music."),
2863 quiz::Quiz("Plate Reverb","A device that produces artificial reverberation by sendin\
2864 g vibrations across a metal plate via a transducer similar to a speaker driver. Phys\
2865 ical plate reverbs today are considered a vintage form of artificial reverb; nowaday\
2866 s, most plate reverb effects are emulated digitally by plugins or reverb units."),
2867 quiz::Quiz("Playback Head","A transducer that converts magnetic flux recorded on tap\
2868 e into an audio signal for playback."),
2869 quiz::Quiz("Playback Mode","A configuration on a console that allows quick playback \
2870 of the signal previously recorded on tape or via DAW via the monitor mixer."),
2871 quiz::Quiz("Playback","1) The reproduction of recorded audio. 2) In motion picture o\
2872 r video production, the reproduction of the music over loudspeakers so the performer\
2873 s/musicians can perform in time to the music for the camera."),
2874 quiz::Quiz("Playlist","1) See "Take." 2) A user-defined selection of songs; a featur\
2875 e available on most streaming and digital media players."),
2876 quiz::Quiz("PLL","A phase locked loop is, in essence, an oscillator that tries to ma\
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2877 tch the frequency of – or more importantly, a division or multiple of the frequency \
2878 of – another signal. This is most commonly used to create a frequency that is much h\
2879 igher than the incoming reference signal – such as a timing module that can create a\
2880 n output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a v\
2881 ery high frequency oscillator that is locked to a multiple of an incoming pitch – pe\
2882 rhaps to drive a special circuit such as a switched-capacitor filter."),
2883 quiz::Quiz("Plug","A connector, usually on a cable, that mates with a jack."),
2884 quiz::Quiz("Polar Pattern","1) In microphones, a graphic display of the area around \
2885 the microphone that is sensitive to sound waves, detailing the audio output levels i\
2886 n dB of sound arriving from different directions. Similar to “Pickup pattern,” but m\
2887 ore specific. 2) In speakers, a graphic display of the speaker’s dispersion of sound\
2888 ."),
2889 quiz::Quiz("Polarity","The direction of current flow or magnetizing force."),
2890 quiz::Quiz("Polarizer","An inverter multiplies an incoming control voltage by -1. In\
2891 the case of a gate or logic inverter, it reverses the high and low states so that (\
2892 for example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pol\
2893 arizer, as it changes the polarity (+ versus -) of a signal. A control voltage inver\
2894 ter is often combined with an offset voltage to adjust the output voltage into the d\
2895 esired range. For example, if you had an envelope generator that had an output range\
2896 of 0 to +8 volts, and you just inverted it, the result would be 0 to -8 volts. Sinc\
2897 e some modules such as voltage controlled amplifiers usually expect only positive vo\
2898 ltages, you would then need to add 8 volts to that result to get an upside-down (inv\
2899 erted) envelope that still had an overall range of 0 to +8v."),
2900 quiz::Quiz("Polarizing Voltage","In condenser and electret microphones, the introduc\
2901 tion of a small amount of electrical current to create the magnetism by which the ca\
2902 pacitor converts audio signals to electrical current. In condenser microphones, pola\
2903 rizing voltage is provided externally (see also “Phantom Power”); in electret microp\
2904 hones, the polarizing voltage is permanently impressed on the condenser during manuf\
2905 acturing."),
2906 quiz::Quiz("Pole Pieces","Iron or other magnetic material that conducts magnetic for\
2907 ce for use in transducers like record heads, playback heads, microphones, speakers, \
2908 etc."),
2909 quiz::Quiz("Pole","This is a technical term that helps describe the design of a filt\
2910 er. Each pole of a filter attenuates frequencies beyond its cutoff or corner frequen\
2911 cy by 6 decibels (dB)/octave; the more poles, the stronger the filtering effect. A 4\
2912 -pole low pass filter, for example, attenuates frequencies one octave above its cuto\
2913 ff frequency by 24 dB; frequencies two octaves above the cutoff are attenuated by 48\
2914 dB and so forth."),
2915 quiz::Quiz("Polyphonic","The term \"polyphonic\" refers to a synthesizer that can pl\
2916 ay more than one individually articulated note at a time; in most cases, those notes\
2917 all play a similar sound or patch. Able to play more than one pitch or “voice” at t\
2918 he same time. A term commonly used to describe synths and keyboards. (See also “Voic\
2919 e.”)"),
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2920 quiz::Quiz("Ponging (Bouncing)", "The technique of combining and mixing multiple trac\
2921 ks onto one or two tracks (mono or stereo). This can be done in real-time or analog \
2922 by playing the tracks through the console and recording them onto separate tracks, o\
2923 r digitally through a digital audio workstation. Bouncing was once used frequently b\
2924 y engineers to free up additional tracks for recording, but in digital workstations \
2925 where tracks are virtually unlimited, this practice is basically obsolete. Today, en\
2926 gineers typically bounce tracks for the purpose of creating a preliminary or final m\
2927 ix of a song."),
2928 quiz::Quiz("Pop Filter", "A device that is placed over a microphone or between the mi\
2929 crophone and vocalist to prevent loud "pop" sounds created by the vocalist's breath \
2930 directed toward the microphone."),
2931 quiz::Quiz("Port", "1) A connection point in computer or electronic device for transm\
2932 itting and receiving digital data, similarly to how a jack receives and transmits au\
2933 dio signals. 2) An opening or vent in a speaker case that resonates with air movemen\
2934 t in the speaker, used in bass reflex speakers and woofers to enhance low frequencie\
2935 s."),
2936 quiz::Quiz("Portamento", "A pitch change that smoothly glides from one pitch to anothe\
2937 r. Also refers to the synthesizer mode or MIDI command that allows or causes this t\
2938 o happen."),
2939 quiz::Quiz("Post Production", "Refers to the work of adding tracks, editing and other\
2940 fine tuning after primary recording or filming has taken place. Post-production in \
2941 recording includes such things as additional overdubs, editing, mixing and mastering\
2942 . Post-production in film includes a wide range of additional audio and visual effec\
2943 ts. NOTE: We mention film in this context because film post-production includes a l\
2944 ot of audio work (e.g., voiceovers, foley, audio mixing and editing) to the point th\
2945 at many audio engineers are involved in film post-production as a full-time career." \
2946 ),
2947 quiz::Quiz("Post Roll", "A segment of blank tape (or track silence, on a DAW) that ru\
2948 ns past the end of the recording. (See also "Pre-Roll.")"),
2949 quiz::Quiz("Post-Fader", "Refers to an aux send position or setting that places the s\
2950 end after the channel fader within the signal path. Sending a signal post-fader mean\
2951 s the fader itself affects the level of the send signal, as opposed to pre-fader. (S\
2952 ee also Pre-Fader.)"),
2953 quiz::Quiz("Post", "Refers to an aux send position or setting that places the send af\
2954 ter the channel fader within the signal path. Sending a signal post-fader means the \
2955 fader itself affects the level of the send signal, as opposed to pre-fader. (See als\
2956 o Pre-Fader.)"),
2957 quiz::Quiz("Pot", "Often thought of as a fancy word for "knob," a potentiometer is ba\
2958 sically any mechanism that controls input or output voltage by varying amounts (for \
2959 example, panning a signal left/right, volume control, or the amount of signal sent t\
2960 o an aux send or bus. Potentiometers can be knobs or faders, meaning that almost eve\
2961 ry control on a console that isn't a button or switch is a potentiometer. However, m\
2962 any engineers commonly refer to faders as "faders" and knobs as "pots.""),
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2963 quiz::Quiz("Potentiometer","(Abbreviated "Pot") Often thought of as a fancy word for\
2964 "knob," a potentiometer is basically any mechanism that controls input or output vo\
2965 ltage by varying amounts (for example, panning a signal left/right, volume control, \
2966 or the amount of signal sent to an aux send or bus. Potentiometers can be knobs or f\
2967 aders, meaning that almost every control on a console that isn't a button or switch \
2968 is a potentiometer. However, many engineers commonly refer to faders as "faders" and\
2969 knobs as "pots.""),
2970 quiz::Quiz("Power Amplifier","(abbreviated "Power Amp") A device that amplifies a li\
2971 ne level signal to drive a speaker or set of speakers. (See also "Line Level.""),
2972 quiz::Quiz("Power Distribution Board","This simple circuit board takes the output of\
2973 your modular system's power supply and creates multiple copies of it, routed to con\
2974 nectors that go to your individual modules."),
2975 quiz::Quiz("PPQN","When you send a clock signal (usually a gate signal or other elec\
2976 trical pulse) around a modular synth to move sequencers through their steps and the \
2977 such, it's good to know how fast that clock is pulsing. This is usually defined in t\
2978 erms of how many pulses there are per quarter note - PPQ or PPQN for short. If the c\
2979 lock is just happening every quarter note, then the clock speed is 1 PPQN; in the ca\
2980 se of DIN Sync (a popular standard among early Roland synths, with DIN being the typ\
2981 e of electrical connector used) or MIDI clocks, the standard is 24 PPQN. This means \
2982 the master pulse can define a triplet for every 8th note (8 x 3)."),
2983 quiz::Quiz("Pre / Post Switch","A switch on the input module that determines whether\
2984 the send control comes before or after the main channel fader in the signal path (S\
2985 ee also "Pre-Fader," "Post-Fader.""),
2986 quiz::Quiz("Pre Emphasis","A boosting of high frequencies during the recording proce\
2987 ss to keep the audible signal above the noise floor."),
2988 quiz::Quiz("Pre Fader","Refers to an aux send position or setting that places the se\
2989 nd before the channel fader within the signal path. Sending a signal pre-fader means\
2990 the fader does not affect the level of the send signal, as opposed to pre-fader."),
2991 quiz::Quiz("Pre-Delay","A parameter on a reverb unit or plugin that determines the a\
2992 mount of time (delay) between the original dry sound and the early reflections of re\
2993 verberation. This feature is often used to simulate the natural acoustic properties \
2994 of a room, but can also be used to create interesting unnatural effects."),
2995 quiz::Quiz("Pre-Echo","(Also called "Forward Echo") A compression artifact that ofte\
2996 n occurs in digital audio in which an "echo" of a sound (or part of a sound) is hear\
2997 d ahead of the sound itself, often due to the data inconsistencies in certain compre\
2998 ssed digital formats. A type of pre-echo can also sometimes occur in the end product\
2999 of a recording, occurring on tape as a result of low-level leakage caused by print-\
3000 through, and also on vinyl records due to physical differences and/or deformities in\
3001 the grooves between silence and a loud transient. In digital formats, pre-echo is g\
3002 enerally an unwanted problem that requires additional signal processing to resolve-b\
3003 ut in some cases it can also be used on purpose as a sound effect (not to be confuse\
3004 d with "Reverse Echo")."),
3005 quiz::Quiz("Pre-Fade Listen (PFL)","A function on the channel strip of a mixer or DA\

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3006 W that allows a channel signal to be heard and often metered before the channel fade\
3007 r."),
3008 quiz::Quiz("Preamplifier (Preamp)","A low-noise amplifier designed to take a low-lev\
3009 el signal (for example, from a microphone) and bring it up to normal line level befo\
3010 re sending it into the mixing console."),
3011 quiz::Quiz("Precedence Effect (Haas Effect)","Simply stated, a factor in human heari\
3012 ng in which we perceive the source of a sound by its timing rather than its sound le\
3013 vel. In his research, Helmut Haas determined that the first sound waves to reach our\
3014 ears help our brains determine where the sound is coming from, rather than its refl\
3015 ection or reproduction from another source. The reflection of the sound must be at l\
3016 east 10dB louder than the original source, or delayed by more than 30ms (where we ca\
3017 n perceive it as an echo), before it affects our perception of the direction of the \
3018 sound. This is what helps us distinguish the original sound source without being con\
3019 fused by reflections and reverberations off of nearby surfaces. Understanding the Ha\
3020 as effect is particularly useful in live audio settings, especially in large venues \
3021 where loudspeakers are time-delayed to match the initial sound waves coming from the\
3022 source."),
3023 quiz::Quiz("Precision Adder","Synthesizers are very sensitive to unintentional varia\
3024 tions in pitch control voltage – any error can result in the oscillators under contr\
3025 ol going out of tune. Therefore, whenever you add together pitch control voltages in\
3026 side a modular synth, you really should be using a precision adder that precisely ad\
3027 ds together the pitch voltages without introducing an error. Ordinary mixers might s\
3028 lightly attenuate or amplify a voltage passed through them, which in most cases wou\
3029 ld create tuning errors."),
3030 quiz::Quiz("Premix","1) The process of mixing a set of tracks as group, then managin\
3031 g the mixed group in the context of the other tracks by routing them to an auxiliary\
3032 channel. Consolidating tracks by bouncing is a form of premixing, but a premix is n\
3033 ot necessarily pre-recorded. (See also “Bouncing.”) 2) An important part of film pos\
3034 t-production in which the process of mixing a section of audio for combination with \
3035 the others. Dialogue, Foley, SFX and music may all be premixed before being combined\
3036 together under the video."),
3037 quiz::Quiz("Presence Frequencies","The range of audio frequencies between 4 kHz and \
3038 6 kHz that when boosted, can increase the sense of presence, especially on voices.")\
3039 ,
3040 quiz::Quiz("Presence","1) In amplification and mixing, the boosting of upper-mid fre\
3041 quencies to cause a sound or instrument to cut through, creating the impression that\
3042 the sound source is more “present,” right next to the listener. 2) See “Room Tone.”\
3043 "),
3044 quiz::Quiz("Preset","A factory programmed set of parameters on a synth, signal proce\
3045 ssor, plug-in or other electronic device."),
3046 quiz::Quiz("Pressure Microphone","(Also called “pressure operative microphone”) – A \
3047 microphone whose diaphragm responds to incoming sound wave pressure as it works agai\
3048 nst the normal or controlled air pressure inside the microphone case. This design ma\
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3049 kes the diaphragm sensitive to pressure regardless of direction, giving it an omni\
3050 rectional pickup pattern. (See also "Omnidirectional Pattern.")"),
3051 quiz::Quiz("Pressure Sensitivity (Aftertouch)","A feature in some keyboard instrumen\
3052 ts by which applying additional pressure to a key after it has been pressed can acti\
3053 vate an additional MIDI control command. a synthesizer or Keyboard Controller of Aft\
3054 er Touch (a control or operational function of a synthesizer where pressing a key aft\
3055 er it has been pressed, and before it is released, will activate a control command \
3056 that can be set by the player)."),
3057 quiz::Quiz("Pressure Zone Microphone (Boundary Microphone)","An omnidirectional micr\
3058 ophone designed to be placed flush against a flat surface (or boundary), effectively\
3059 creating a "half-Omni" pickup pattern while eliminating the danger of phase issues \
3060 from reflected sounds. A popular type of boundary microphone is Crown Audio's tradem\
3061 ark Pressure Zone Microphone (PZM)."),
3062 quiz::Quiz("Pressure-Gradient Microphone","(Also called "Velocity Microphone") A mic\
3063 rophone whose diaphragm is exposed front and back, with diaphragm movement being cau\
3064 sed by the pressure difference between its front and back. This creates a bi-directi\
3065 onal or "figure-8" pickup pattern (See also "Bi-Directional Pattern.")"),
3066 quiz::Quiz("Pressure","Some keyboards measure how hard you press down on the keys, a\
3067 nd convert this to a voltage (or other control signal such as MIDI, which can then b\
3068 e converted into a control voltage) that you can use to add expression to a note, su\
3069 ch as adding vibrato or opening the filter wider. Monophonic aftertouch measures one\
3070 pressure value for the entire keyboard, regardless of which key(s) you are pressing\
3071 ; polyphonic aftertouch produces a signal for each individual key. Important trivia:\
3072 Touch plate keyboards actually measure the surface area of the skin touching them r\
3073 ather than pressure or force - so you can increase or decrease the aftertouch amount\
3074 by rolling between the tip and length of your finger."),
3075 quiz::Quiz("Print Through","The unwanted transfer of magnetic flux from one layer of\
3076 analog tape to another."),
3077 quiz::Quiz("Pro Tools","Avid's trade name for its digital audio workstation (DAW) th\
3078 at has become an industry standard in professional recording studios."),
3079 quiz::Quiz("Producer","In music, the producer is the director of an audio recording \
3080 project; the person responsible for getting a final product of desired quality withi\
3081 n a budget."),
3082 quiz::Quiz("Production Studio","Broadly speaking, any space dedicated to production \
3083 within the arts, for example, film/video, animation or post production. In the conte\
3084 xt of audio, a production studio is effectively a recording studio that specializes \
3085 in the assembly and mixing of commercials and radio programs from pre recorded music\
3086 and effects with newly recorded dialogue."),
3087 quiz::Quiz("Production","1) The collective actions that go into producing music. 2) \
3088 Describing the quality of a recording—the end result of production decisions during \
3089 the recording and mixing process."),
3090 quiz::Quiz("Program Change","A MIDI message that tells the receiving device to chang\
3091 e presets."),
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3092 quiz::Quiz("Programmable","Able to have the parameters changed by the user, especial\
3093 ly in a computer controlled device."),
3094 quiz::Quiz("Prompt","A set of instructions for the user to follow, which appears on \
3095 a computer screen."),
3096 quiz::Quiz("Protocol","In digital and information technology, a set of rules governi\
3097 ng the structuring and transmitting of data in a standardized format so all related \
3098 devices can properly interpret the data."),
3099 quiz::Quiz("Proximity Effect","The natural boost in the microphone's output for bass\
3100 frequencies as the mic is placed closer to the sound source."),
3101 quiz::Quiz("Psychoacoustics","The study of how humans perceive and respond to sound,\
3102 not just in the context of interpreting the physical sound waves, but also taking p\
3103 sychological and emotional factors into account. This branch of science is helpful t\
3104 o audio engineers in understanding how the brain interprets various sounds and frequ\
3105 encies."),
3106 quiz::Quiz("Puck","Any circular piece of metal, fiber, rubber, etc., which drives so\
3107 mething from a rotating power source. A common example in the recording studio is th\
3108 e puck in a rotating Leslie speaker."),
3109 quiz::Quiz("Pulse Code Modulation (PCM)","A process by which analog signals are tran\
3110 slated to digital code. This is done by taking samples of the amplitude of the analo\
3111 g signal at regular rapid intervals, then translating it into binary numbers as a di\
3112 gital representation of the original signal. The faster the sample rate, the better \
3113 the digital reproduction. PCM is the most common form of A/D conversion in digital a\
3114 udio."),
3115 quiz::Quiz("Pulse Per Quarter Note","When you send a clock signal (usually a gate si\
3116 gnal or other electrical pulse) around a modular synth to move sequencers through th\
3117 eir steps and the such, it's good to know how fast that clock is pulsing. This is us\
3118 ually defined in terms of how many pulses there are per quarter note - PPQ or PPQN f\
3119 or short. If the clock is just happening every quarter note, then the clock speed is\
3120 1 PPQN; in the case of DIN Sync (a popular standard among early Roland synths, with\
3121 DIN being the type of electrical connector used) or MIDI clocks, the standard is 24\
3122 PPQN. This means the master pulse can define a triplet for every 8th note (8 x 3)."\
3123 ),
3124 quiz::Quiz("Pulse Width Modulation","Most oscillators that output a square waveform \
3125 also have an additional control voltage input that sets the width of the top portion\
3126 of the "square" wave (obviously, making the top portion wider makes the bottom port\
3127 ion narrower and vice versa). The act of varying the width of the resulting pulse wa\
3128 ve creates a sort of Doppler shift; varying the width back and forth - for example, \
3129 by modulating the pulse width with a low frequency oscillator - creates a chorusing \
3130 effect that can sound like a detuned pair of oscillators. The resulting effect is re\
3131 ferred to as pulse width modulation. The process of using a control voltage to vary \
3132 the width of a pulse wave form, essentially switching between square waves and pulse\
3133 waves. This has the effect of creating richer timbres, giving sounds a thicker, mor\
3134 e lush feel, or of giving a digital sound more analog properties."),
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3135 quiz::Quiz("Pulse","Pulse has a couple of different meanings in a modular synth. Whe\
3136 n you alter the shape of a square wave so that one portion is narrower than the othe\
3137 r, it is referred to a pulse wave (see Pulse Wave Modulation below). Also, a narrow \
3138 gate or trigger used as a clocking signal for sequencers and the such is often refer\
3139 red to as a pulse. 1) The steady beat in music based on its tempo, whether audible o\
3140 r perceived. 2) A type of sound wave commonly created and manipulated by synthesizer\
3141 s whose waveform is characterized by sharp rises and drops in amplitude like a squar\
3142 e wave, but whose peaks are shorter than its troughs, giving the wave a pulse-like f\
3143 eel. Also called "Pulse Wave.""),
3144 quiz::Quiz("Pumping and Breathing","In studio jargon, an effect created when a compr\
3145 essor is rapidly compressing and releasing the sound, creating audible changes in th\
3146 e signal level. "Pumping" generally refers to the audible increase of sound levels a\
3147 fter compression has taken place; "breathing" refers to a similar effect with vocals\
3148 , raising the signal volume just as the vocalist is inhaling. Pumping and breathing \
3149 is a sign of cheap compression or over-compression, and is usually undesirable, alth\
3150 ough some engineers and musicians use it on purpose occasionally to create a particu\
3151 lar effect."),
3152 quiz::Quiz("Punch In / Punch Out Recording","The process of activating and/or deacti\
3153 vating the record function on tape or DAW during playback of a passage, usually as t\
3154 he performer plays/sings along. This can be used either as a method of doing quick o\
3155 verdubs, or as a way of getting a better take on a certain passage without having to\
3156 start the track from the beginning."),
3157 quiz::Quiz("Pure Tone","A tone consisting of only the fundamental frequency with no \
3158 overtones or harmonics, graphically represented as a simple sine wave."),
3159 quiz::Quiz("PVC","PVC stands for pitch to voltage conversion. In the quest to play a\
3160 voltage-controlled synthesizer with something other than a keyboard-like thingy (to\
3161 uch plates included), some have designed modules or other equipment that attempt to \
3162 detect the pitch of an audio signal – say, from a guitar, flute, or singer – and con\
3163 vert that pitch to a corresponding voltage that can drive a VCO in unison with the o\
3164 riginal sound."),
3165 quiz::Quiz("PWM","Most oscillators that output a square waveform also have an additi\
3166 onal control voltage input that sets the width of the top portion of the "square" wa\
3167 ve (obviously, making the top portion wider makes the bottom portion narrower and vi\
3168 ce versa). The act of varying the width of the resulting pulse wave creates a sort o\
3169 f Doppler shift; varying the width back and forth – for example, by modulating the p\
3170 ulse width with a low frequency oscillator – creates a chorusing effect that can sou\
3171 nd like a detuned pair of oscillators. The resulting effect is referred to as pulse \
3172 width modulation."),
3173 quiz::Quiz("PZM","Abbreviation for Crown Audio's Pressure Zone Microphone. (See also\
3174 "Boundary Microphone.""),
3175 quiz::Quiz("Q – (Also called "Q Factor")","Stands for "Quality Factor," defining the\
3176 bandwidth of frequencies that will be affected by an equalizer. The lower the Q, th\
3177 e broader the bandwidth curve of frequencies that will be boosted or cut. If you com\
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3178 e from the pro audio world, you may be used to Q referring to the width or narrowness\
3179 s of a peak or notch filter. In a synthesizer filter, when you increase the resonance\
3180 e (feedback), a peak forms around the cutoff frequency of the filter's curve or shape\
3181 e. The higher the resonance, the higher and narrower this peak. As a result, some us\
3182 ed to use the audio term Q to refer to the resonance amount, although you don't hear\
3183 that term used nearly as much today."),
3184 quiz::Quiz("Quadraphonic","A now rarely-used system of four-channel sound where the \
3185 channels are designated as left front, left back, right front, right back, intended \
3186 to deliver sound from all four corners of a room. Quadraphonic sound was a precursor\
3187 to the surround-sound systems of today."),
3188 quiz::Quiz("Quadrature","You can define a full cycle of a waveform as consisting of \
3189 360 degrees, akin to a circle. One quarter of the way around this circle - or moving\
3190 to a point that is one quarter of the way through a cyclical wave - is 90°. A sine \
3191 and cosine wave are shifted 90° degrees or a quarter cycle out of alignment (phase) \
3192 with each other. Since this is a quarter of a cycle, this is often referred to as a \
3193 quadrature relationship."),
3194 quiz::Quiz("Quantization Distortion","Quantization Distortion/Quantization Error - The\
3195 he effective "error in translation" between an analog signal and its sampled counter\
3196 part due to the rounding of a large number of analog values to the nearest digital q\
3197 uantity. This often results in additional random frequencies in the sound, often hea\
3198 rd as noise."),
3199 quiz::Quiz("Quantization Noise","The modulation noise in a signal resulting from qua\
3200 ntization error. "),
3201 quiz::Quiz("Quantization","1) In digital music, the process of adjusting the rhythmi\
3202 c performance of music by moving the notes to precise locations on the time line, ef\
3203 fectively "rounding" the note occurrences to the nearest defined increment. 2) In an\
3204 analog-to-digital conversion, the use of the same mathematical quantization principles\
3205 to convert an analog signal into a smaller set of steps (a digital quantity)."),
3206 quiz::Quiz("Quantizer","A quantizer auto-corrects the input voltage to the nearest d\
3207 esired target, such as the voltage that corresponds to a semitone or other note in a\
3208 scale. These are occasionally built into modules like sequencers or oscillators, bu\
3209 t quite often they are standalone modules."),
3210 quiz::Quiz("Rack Ears","Rack Ears/Rack Flanges - Mounting brackets that can be atta\
3211 ched to equipment so it can be mounted in a standard equipment rack."),
3212 quiz::Quiz("Rack Mounted","Describing outboard gear that can be housed in an equipme\
3213 nt rack."),
3214 quiz::Quiz("Rack Rash","When you mount a module into a case, the head of the screw o\
3215 r bolt used to mount the module can scratch the faceplate of the module. These scrat\
3216 ches are referred to as rack rash. You can almost never see it when you mount a modu\
3217 le, as the scratches are behind the screw or bolt head, but nonetheless some will pa\
3218 y more for a used module that is unscratched. So buy a bag of plastic washers and pu\
3219 t them behind the screw or bolt head just to remove another reason for someone to no\
3220 t buy your used module."),
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3221 quiz::Quiz("Rack Unit", "Rack-mounted equipment usually follows a standard set of dim\
3222 ensions, including 19" (48.3 cm) for width, and a "rack unit" (or U for short) for he\
3223 ight equaling 1.75" (4.4 cm) per U. Many common modular synthesizer formats follow t\
3224 he rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25" o\
3225 r 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (\
3226 sometimes referred to as MU for Moog Unit)."),
3227 quiz::Quiz("Radiation Pattern", "A graphic depiction of speaker coverage. This is not\
3228 unlike the polar pattern of a microphone, with the exception that a polar pattern d\
3229 escribes the area where sound arrives at the microphone, while a radiation pattern d\
3230 escribes how sound is dispersed from the loudspeaker."),
3231 quiz::Quiz("Radiation", "The angle and pattern of coverage of a speaker."),
3232 quiz::Quiz("Ramp", "In general, a ramp refers to any voltage that is steadily raising\
3233 or falling; quite often it resets when it reaches a target voltage and starts over \
3234 again. A sawtooth oscillator waveform is sometime referred to as a ramp. Sometimes, \
3235 the individual stages of an envelope generator are also referred to a ramp as it rai\
3236 ses from 0 volts to a maximum level such as 5v for the attack stage, then falls from\
3237 this peak to the sustain level for the decay stage."),
3238 quiz::Quiz("Random Access Memory (RAM)", "The "short-term" memory in a computer that \
3239 is used in tandem with the processor for performing immediate tasks (as opposed to h\
3240 ard-drive storage memory where projects are saved and recalled). In the recording st\
3241 udios, the more RAM a computer has, the more ability it has to handle large amounts \
3242 of data at a time (for example, in multi-track recording or working with virtual MID\
3243 I instruments)."),
3244 quiz::Quiz("Random Note Generator", "A device that generates random pitches at a set \
3245 rate, used in synthesizers."),
3246 quiz::Quiz("Random", "Most voltages moving around inside a modular synth are very pur\
3247 poseful in their variations: the repeating waveforms of an audio rate or low frequen\
3248 cy oscillator; the rising then falling voltages of an envelope generator. However, i\
3249 t can also be useful to have randomly wandering voltages to create everything from s\
3250 ubtle variations in pitch to wildly varying volumes or filterings. Noise is an examp\
3251 le of an audio-rate random signal."),
3252 quiz::Quiz("Rap", "To perform a spoken rhythmic part to a music or percussion perform\
3253 ance."),
3254 quiz::Quiz("Rarefaction", "The reduced density of air particles during the trough of \
3255 a sound wave; in the context of "compression and rarefaction," it is the opposite of\
3256 compression. (See also "Compression.")"),
3257 quiz::Quiz("Ratcheting", "This is a trick used with sequencers where one stage of the\
3258 sequence may be triggered quickly multiple times, rather than just once as you step\
3259 to that stage. For example, the result may be a series of quarter notes, with a bur\
3260 st of four sixteenth notes appearing instead for one or more stages."),
3261 quiz::Quiz("Rate", "This word is used sometimes to refer to the speed or frequency of\
3262 a low frequency oscillator or similar repetitive function, such a sequencer's tempo\
3263 clock."),
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3264 quiz::Quiz("Rated Load Impedance","The input impedance, or opposition to current flo\
3265 w by an input of a device, that a piece of equipment is designed to feed."),
3266 quiz::Quiz("RCA Plug","(Also called Phono Plug) A common audio connector found on mo\
3267 st stereo systems with a center pin as one connection and an outer shell as the seco\
3268 nd connection."),
3269 quiz::Quiz("Read Only Memory (ROM)","A type of data storage that cannot be erased or\
3270 reprogrammed by the user. The most common form of ROM in audio/video settings today\
3271 is optical storage media (i.e, CD, DVD, CD-ROM and DVD-ROM)."),
3272 quiz::Quiz("Read","To retrieve information bits from a storage device; in digital au\
3273 dio, the reproduction of digital signals."),
3274 quiz::Quiz("Reason","Popular music software program from Propellerhead Software. It\
3275 offers the digital equivalent of hardware synthesizers, samplers, signal processors\
3276 , sequencers and mixers. Reason works as a virtual music studio, or as a set of virt\
3277 ual musical instruments which can be played live or used with other sequencing softw\
3278 are."),
3279 quiz::Quiz("Recapping","Electronic components can age. Certain types of capacitors -\
3280 namely, electrolytic and tantalum, often used in the power supply section - are the\
3281 most likely to deteriorate over time; some put the maximum safe life of an electrol\
3282 ytic capacitor to be 25 years. Therefore, serious vintage synth owners "recap" (repl\
3283 ace the age-sensitive capacitors in) their older equipment."),
3284 quiz::Quiz("Record Head","A device on an analog tape machine that changes electrical\
3285 current to magnetic energy; the changes of the magnetism match the waveshape of the\
3286 audio signal fed to the head."),
3287 quiz::Quiz("Record Level","A control on a tape machine that determines the amount of\
3288 magnetic flux recorded on the tape, or the DAW control that determines the level of\
3289 the digital signal recorded to the sound file."),
3290 quiz::Quiz("Record Monitor","On some tape machines, a switch position that allows th\
3291 e VU meter and sound output of the tape machine electronics to monitor the input sig\
3292 nal to the tape machine."),
3293 quiz::Quiz("Record Ready","A control state of a multitrack tape recorder where the d\
3294 esignated track will begin recording when the record function of the tape recorder i\
3295 s activated."),
3296 quiz::Quiz("Recording Bus","A bus that sends a mix signals from the console channels\
3297 to the multitrack recorder or DAW. (See also "Bus.")"),
3298 quiz::Quiz("Recording Session","A bloc of time in which music is being recorded in t\
3299 he studio."),
3300 quiz::Quiz("Rectifier","A circuit that makes sure a voltage stays only positive or n\
3301 egative. In power supplies, it is used to remove the negative component of AC voltag\
3302 e, or to protect you from plugging in module's power connector backwards. As a modul\
3303 e, a half-wave rectifier passes only positive voltages and replaces anything negativ\
3304 e with 0v; a full-wave rectifier takes any negative voltages and inverts them so the\
3305 y become positive. This effectively doubles the frequency of many simple waveforms, \
3306 like the triangle and sine."),
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3307 quiz::Quiz("Red Noise", "Also referred to as brown noise, technically it's a type of \
3308 noise whose power density (spectral loudness) decreases 6 dB per octave with increas\
3309 ing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distanc\
3310 e. It can also be used a slowly changing random control voltage or modulation signal\
3311 , instead of as an audio source."),
3312 quiz::Quiz("Reel", "1) The hub and flanges onto which analog tape is spooled; recordi\
3313 ng and playback involves unspooling the tape from one reel and onto another. 2) Some\
3314 times also called "demo reel," a compilation of audio or video that demonstrates the\
3315 abilities of a musician, audio engineer, actor, or other audio/visual professional.\
3316 Unlike a demo, which is intended to pitch one or more songs, a reel is a demo inten\
3317 ded to promote the abilities of the professional rather than the product itself. The\
3318 term itself is a holdover from the days when this promotional material was delivere\
3319 d on reels."),
3320 quiz::Quiz("Reference Level", "1) A standard baseline level of volume used to measure\
3321 how much level is present in dB above or below the baseline. 2) See "Operating Level\
3322 1.""),
3323 quiz::Quiz("Reference Tone", "A single-frequency tone (often at 1000 kHz) used to cal\
3324 ibrate the levels of sound equipment; the tone used to set reference level. (See als\
3325 o "Test Tones.")"),
3326 quiz::Quiz("Reflected Sound", "Sound that reaches a microphone or listener after one \
3327 or more reflections from surrounding surfaces."),
3328 quiz::Quiz("Reflection", "In acoustics, the bouncing of sound waves off of a flat sur\
3329 face, as opposed to absorption. Reflection can have a great impact on how we perceiv\
3330 e the collective sound; reflected sounds from a distance is perceived as echo, while\
3331 reverberation is created from thousands of reflections. (See also "Absorption," "Ea\
3332 rly Reflection," "Echo," "Reverberation.""),
3333 quiz::Quiz("Regeneration", "Regeneration can have a couple of different meanings insi\
3334 de a synth, both meaning feedback. An echo unit can feed some of its output back int\
3335 o its input, causing the delayed signal to be repeated again; this is sometimes refe\
3336 rred to as regeneration. Also, very rarely you will hear resonance in a filter refer\
3337 red to as regeneration."),
3338 quiz::Quiz("Regulated Power Supply", "A device to supply power to electronic equipmen\
3339 t whose output voltage will not fluctuate when more equipment is turned on, or if th\
3340 ere is a change in voltage of the power line. A regulated power supply is designed t\
3341 o protect sensitive electronics from destructive power surges."),
3342 quiz::Quiz("Relay", "An electromagnetically activated switch that connects or disconn\
3343 ects two terminals when a control voltage is applied."),
3344 quiz::Quiz("Release Time", "In dynamics signal processors, the time it takes for the \
3345 output signal to return to original levels when the input signal crosses the designa\
3346 ted threshold."),
3347 quiz::Quiz("Release", "This refers to the final stage of an envelope that typically f\
3348 alls back to zero volts, usually resulting in silence. It is often used in the conte\
3349 xt of talking about an Attack/Release (AR) or Attack/Decay/Sustain/Release (ADSR) en\
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3350 velope generator, but can refer to any final stage of an envelope."),
3351 quiz::Quiz("Remote","1) A device that controls the functions of another device wirel\
3352 essly. 2) Describing on-site recording, as opposed to recording in the studio."),
3353 quiz::Quiz("Reset","The Reset input on a module accepts a trigger or gate signal, an\
3354 d tells the module to go back the beginning of whatever it was doing. In the case of\
3355 a clock divider, this means pretend the next clock is the first clock you should be\
3356 counting in the division (more on that in the full definition). In the case of a se\
3357 quencer, it means go back to the first stage. In the case of an envelope, it means g\
3358 o back to the start of the attack. In the case of a gate delay, it means to re-start\
3359 the timer for the delay."),
3360 quiz::Quiz("Residual Magnetization","The amount of magnetism left in a magnetic mate\
3361 rial after the magnetizing force is removed. Residual magnetism can accumulate in ta\
3362 pe machines over time, either creating distortions and noise in the sound output or \
3363 partially erasing the tape."),
3364 quiz::Quiz("Residual Noise","The noise level left on recording tape after it has bee\
3365 n erased."),
3366 quiz::Quiz("Resistance","The opposition of a substance to the flow of electrical cur\
3367 rent, measured in ohms."),
3368 quiz::Quiz("Resistor","An electrical component with a specific amount of resistance \
3369 to electrical current, used within the circuit to regulate the flow of current."),
3370 quiz::Quiz("Resonance","The natural tendency of physical substances to vibrate with \
3371 more energy at certain frequencies. The principle of resonance is a key element in t\
3372 he design of acoustic instruments; for example, the hollow chamber of a guitar or vi\
3373 olin is designed to resonate with the vibrations of the string. Resonance also plays\
3374 a role the acoustic design of a space, and even in developing good vocal technique \
3375 to project the voice. When the output of a filter is fed back into its input, the re\
3376 sult is an increased boost in the harmonics right around the filter's cutoff or corn\
3377 er frequency. The audible result is similar to playing a sound in a room that has a \
3378 resonance - sympathetic, reinforcing echo or vibration - at a certain frequency. The\
3379 refore, the term resonance is often used to refer to a filter's feedback amount."),
3380 quiz::Quiz("Resonant Frequency","A frequency at which a physical item vibrates natur\
3381 ally."),
3382 quiz::Quiz("Resonate","To vibrate at the resonant frequency. Also refers to the ling\
3383 ering reverberation that causes a sound to continue after the sound source has stopp\
3384 ed. This continuing sound is due to the sympathetic resonance of nearby objects."),
3385 quiz::Quiz("Resonator","Many acoustic instruments include a body or sound chamber th\
3386 at "resonates" - sympathetically vibrates at, or reinforces - one or more frequencie\
3387 s. To simulate this effect in modular synths, you can get a specialized filter or eq\
3388 ualization module that boosts the sound at typically three or so user-definable freq\
3389 uencies, each usually within a narrow band. This is one of the secrets of synthesizi\
3390 ng real-world sounds or spaces."),
3391 quiz::Quiz("Reverb (Reverberation)","1) Short for "Reverberation." (See "Reverberati\
3392 on.") 2) A signal processor or plug-in that creates artificial reverb to a signal.")\
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3393 ,
3394 quiz::Quiz("Reverb Time (RT)","The time it takes for the reverberation or echoes of \
3395 a sound source to die out after the direct sound has stopped. Specifically, the reve\
3396 rb time is measured between the point at which the sound source stops and the point \
3397 at which the reverberation levels fall by 60 dB."),
3398 quiz::Quiz("Reverb","Short for reverberation. This is an effect device that mimics b\
3399 eing in a room where you can hear the original sound reflect off the walls multiple \
3400 times, bouncing around in a wash of sound until it eventually decays into silence. A\
3401 reverb can greatly enhance the sound of a synthesizer, adding lushness and dimensio\
3402 n to what might otherwise be a stark sound. There are relatively few modules that im\
3403 plement a reverb effect, and even fewer that allow you to voltage control some of it\
3404 s parameters (the ErbeVerb being the most famous); many just use an external reverb \
3405 effect."),
3406 quiz::Quiz("Reverberant Field","Describes the space that is far enough from the soun\
3407 d source that the reverberations are louder than the direct sound."),
3408 quiz::Quiz("Reverberation Chamber","A device built to simulate room reflections."),
3409 quiz::Quiz("Reverberation Envelope","The attack, decay, sustain and release of the r\
3410 everberation volume; or how fast the reverberation reaches peak level and its rate o\
3411 f decay."),
3412 quiz::Quiz("Reverberation","The persistence of a sound after the source stops emitti\
3413 ng it, caused by many discrete echoes arriving at the ear so closely spaced in time \
3414 that the ear cannot separate them."),
3415 quiz::Quiz("RF Interference","The unwanted noise introduced into electronics, circui\
3416 ts and/or audio systems by the presence of RF signals. RF interference in a system c\
3417 an result in humming, buzzing, static or even the reproduction of radio transmission\
3418 s."),
3419 quiz::Quiz("RF Signals","RF Signals (or RF) – Short for Radio Frequency Signals, ele\
3420 ctromagnetic waves that carry wireless radio and television signals. The vast majori\
3421 ty of RF signals exist at frequencies higher than 100 kHz."),
3422 quiz::Quiz("Rhythm Section","The musical instruments in a band or ensemble that are \
3423 responsible for playing rhythmic parts rather than melody parts. In contemporary mus\
3424 ic, rhythm sections typically consist of drums and bass, along with some combinatio\
3425 n of percussion, piano/keyboard and/or guitars."),
3426 quiz::Quiz("Ribbon Controller","This is a long strip that is capable of measuring th\
3427 e position where you press it along its length, and the pressure used to press it. I\
3428 t can be used as an alternate keyboard or as a pitch bend controller, with the posit\
3429 ion determining pitch. Shorter versions also appeared sometimes as alternate control\
3430 lers on synthesizers, such as the Yamaha CS-80."),
3431 quiz::Quiz("Ribbon Microphone","A microphone that converts sound waves to electrical\
3432 current via a thin conductive ribbon set between magnetic poles. Ribbon microphones\
3433 are almost always responsive to sound on both sides of the ribbon, creating a bi-di\
3434 rectional or figure-8 pattern."),
3435 quiz::Quiz("Riff","A short melody repeatedly played in a tune often with variation b\
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3436 etween vocal lines."),
3437 quiz::Quiz("Ring Modulator","Balanced or ring modulation is a special type of amplit\
3438 ude modulation, where one bipolar (swinging both above and below 0 volts) signal - t\
3439 he modulator - is used to vary the amplitude of a second bipolar signal, known as th\
3440 e carrier. The modulator's frequency is both added to and subtracted from the carri\
3441 e's frequency; the resulting harmonics replace the original carrier and modulator.")\
3442 ,
3443 quiz::Quiz("Ringing Out a Room","The process of identifying and compensating for pro\
3444 blem frequencies within a room for the purpose of optimizing live audio within that \
3445 space. This is typically done by sending pink noise through the speakers, turning up\
3446 the microphones to the point of feedback, and using EQ to notch out the offending f\
3447 requencies."),
3448 quiz::Quiz("Rise Time","The rate at which an audio waveform makes a sudden increase \
3449 to a higher amplitude."),
3450 quiz::Quiz("RMS Meter","A meter that recognizes and responds to the effective averag\
3451 e, the RMS level, or the effective average value of an AC waveform, rather than to t\
3452 he peak level. (See also "Root-Mean-Square," "Peak Meter.")"),
3453 quiz::Quiz("Roll Off","The reduction of signal level as the frequency of the signal \
3454 moves away from the cut-off frequency, especially when the cut-off rate is mild."),
3455 quiz::Quiz("Room Equalization","In live audio, an equalizer inserted in the monitor \
3456 system that attempts to compensate for frequency response changes caused by room aco\
3457 ustics."),
3458 quiz::Quiz("Room Sound","The natural ambience of a room, including the reverberation\
3459 and background noise."),
3460 quiz::Quiz("Room Tone","The natural background noise occurring in a room without mus\
3461 ic playing or people speaking. In recording audio for film and TV, on-set sound mixe\
3462 rs capture a take of room tone for the purpose of providing continuity between clips\
3463 of dialogue during post-production."),
3464 quiz::Quiz("Root Mean Square (RMS)","The effective average value of an AC waveform. \
3465 Used as a measure of the overall level of the sound rather than just measuring by th\
3466 e peaks. (See also "RMS Metering," "Peak Metering.")"),
3467 quiz::Quiz("Rotating Head","A circular head with two (or more) gaps that rotates aga\
3468 inst the direction of tape motion at a slight angle to the tape travel."),
3469 quiz::Quiz("Rumble","A low-frequency noise, typically caused by earth/floor vibratio\
3470 n or by uneven surfaces in the drive mechanism of a tape recorder or playback unit.")\
3471 ),
3472 quiz::Quiz("Rythm Tracks","The recording of the rhythm instruments in a music produc\
3473 tion."),
3474 quiz::Quiz("S-trig","Some systems - such as the original Moog modular - use an s-tri\
3475 gger (switch or shorting trigger) instead of a normal gate, which was a wire that wa\
3476 s shorted to 0 volts ground, like the closing of a switch wired to ground. You canno\
3477 t interconnect these two systems without some form of conversion between the two, wh\
3478 ich can be as simple as a special cable."),
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3522 quiz::Quiz("Saturation","On a simple level, saturation is a fancy word for clipping:\
3523 Once the input voltage goes higher (or lower) than a circuit can handle, it is inst\
3524 ead held at that limit. However, saturation usually implies a more rounded, shaped a\
3525 pproach to that clipping limit, resulting in a more pleasing (or at least less annoy\
3526 ing) form of distortion. Tubes circuits are often associated with this soft clipping\
3527 behavior, although it can be emulated in other circuits or even digital signal proc\
3528 essing. Different devices may be sought out for specific sonic character of the way \
3529 they. 1) The point at which magnetic tape reaches full magnetization due to an exces\
3530 s of sound level. This creates some distortion that some audiophiles describe as “an\
3531 alog warmth” a desirable quality in certain instances. 2) The audio distortion that \
3532 occurs by overdriving a signal through a tube amplifier or preamp—again producing co\
3533 lor and warmth in the sound that engineers often find appealing. 3) A digital plugin\
3534 that emulates tape or tube saturation."),
3535 quiz::Quiz("Sawtooth Wave","A waveform that jumps from a zero value to a peak value \
3536 and then immediately drops to a zero value for each cycle. (Sometimes also called “R\
3537 amp Wave.”)"),
3538 quiz::Quiz("Sawtooth","One of the most common waveforms produced in a synthesizer. T\
3539 his ramp-shaped wave contains both even and odd harmonics, strongest at the fundamen\
3540 tal frequency (the note being played) and diminishing at the higher frequencies. The\
3541 result is very bright, loud, “brassy” sound."),
3542 quiz::Quiz("Schmitt Trigger","This is a type of gate detector that looks at a varyin\
3543 g input signal and outputs either a “high” (typically 0, 10, or even 15 volts) signa\
3544 l or a “low” signal (typically 0 volts). When the input goes above one reference thr\
3545 eshold – say, 4 volts – the output goes high. When the input then goes back below a \
3546 second, different threshold – say, 1 volt – then the output goes back low."),
3547 quiz::Quiz("scope","This is a piece of test equipment that displays voltage fluctuat\
3548 ions as graphical waveforms. A 'scope can run at a wide range of frequencies, displa\
3549 ying slowly changing voltages like LFOs or envelopes, or quickly changing voltages l\
3550 ike oscillators and noise. Oscilloscopes used to be bulky pieces of external equipme\
3551 nt, but now you can get USB scopes that offload the display portion of the job to yo\
3552 ur computer, or scopes as modules."),
3553 quiz::Quiz("Scratch","1) A descriptive term meaning “temporary”. 2) A scratch vocal \
3554 is a vocal done during a basic recording session to help the musicians play their pa\
3555 rts. At a later date the final vocal track is overdubbed. 3) The action of a musicia\
3556 n or disc jockey quickly moving a record back and forth on a turntable reproducing t\
3557 he stylus motion to create a rhythm pattern of sound."),
3558 quiz::Quiz("Scrubbing","The action or function of shuttling a piece of recorded audi\
3559 o back and forth while monitoring it, typically to locate a certain point in the rec\
3560 ording. In earlier days, scrubbing was done with reel-to-reel analog tape by manuell\
3561 y turning the reels to pull the tape across the playhead. Today, scrubbing is primar\
3562 ily done digitally on a DAW by dragging the cursor back and forth across the wavefor\
3563 m."),
3564 quiz::Quiz("Second Engineer","An assistant recording engineer."),
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3565 quiz::Quiz("SEM","The Oberheim SEM (Synthesizer Expander Module) was one of their ea\
3566 rliest products. It was an entire synthesizer voice – two oscillators, two simple en\
3567 velopes, VCA, and a very popular two-pole state variable filter design with a knob t\
3568 hat crossfaded between low pass, notch, and high pass outputs plus a separate bandpa\
3569 ss setting – in a cube-like case. Most often today, when a modular manufacturer uses\
3570 the magic letters \"SEM\", they're referring to a filter meant to emulate that in t\
3571 he original Oberheim synth."),
3572 quiz::Quiz("Semi-modular","The components of a semi-modular synth – such as the osci\
3573 llator, filter and amplifier – are pre-wired behind the front panel in what the manu\
3574 facturer considers to be a typical, logical way. However, they also provide patch po\
3575 ints either to access some of its functions (such as the individual waveform outputs\
3576 of the oscillator) to send to other modules, or to override that pre-wiring. Many w\
3577 ho are new to modular synthesis dip their toe in the water by getting a semi-modular\
3578 synth, and then expanding it with additional modules."),
3579 quiz::Quiz("Semitone","Also known as a half step or half tone, this is the smallest \
3580 pitch division in most Western music – such as the difference between a C and a C#. \
3581 With equal temperament (the most common way of tuning a Western scale), this pitch d\
3582 ivision is 1/12 of an octave."),
3583 quiz::Quiz("Send Level","A control determining the signal level sent to a send bus." \
3584 ),
3585 quiz::Quiz("Sensitivity","1) In audio settings, describes the amount of output that \
3586 a microphone can produce from a standard level of sound, as compared to the output o\
3587 f another microphone from the same sound level. 2) In music, describes the artistic \
3588 persona in general."),
3589 quiz::Quiz("Sequence","1) A pre-programmed set of musical events, such as pitches, s\
3590 ounding of samples, and rests, to be played in order by a device. Also refers to the\
3591 action of programming the device to play this set of musical events. 2) Loosely ref\
3592 erring to a segment of music in general."),
3593 quiz::Quiz("Sequencer","The most common type of sequencer you're going to see in a m\
3594 odular synth contains a row of knobs (also known as steps or stages) that may each b\
3595 e set to output a different voltage. A sequencer then goes through steps one at a ti\
3596 me. This is most often used to create repetitive musical lines where each note has t\
3597 he same duration, which is popular in trance-like forms of music as well as the clas\
3598 sic Berlin School style (70s-era Tangerine Dream and Klaus Schulze; current Red Shif\
3599 t and Node). A computerized device or software that can be programmed to play a step\
3600 ped order of musical events, including playing of pitches, sounding of samples, and \
3601 rests."),
3602 quiz::Quiz("Sequential Switch","This module comes in a few different forms; in the m\
3603 ost common, a few different inputs are routed to one output (although they are usual\
3604 ly symmetrical – one input can be switched between several outputs). A pulse or gate\
3605 input then steps through the inputs one at a time, switching which ones is routed t\
3606 o the output. Fancyer sequential switches allow you to set the number of stages, to \
3607 divide an input clock so it switches at a slower tempo than the master clock, or mig\
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3608 ht directly route a series of inputs to corresponding outputs (with usually a summed\  
3609 output as well)."),  
3610 quiz::Quiz("Serial Data","A digital data stream where individual bits are transmitte\  
3611 d one after another over a single connection (as opposed to “parallel data,” in whic\  
3612 h multiple bits can be sent at once). Most data connections in the recording studio \  
3613 transmit serial data—for example, USB, Firewire and MIDI."),  
3614 quiz::Quiz("Series Connection","Connecting devices (especially circuit elements) so \  
3615 that the electrical signal flows from one thing to the next, to the next, etc."),  
3616 quiz::Quiz("Set Up","The positioning of microphones, instruments, connections and mo\  
3617 nitoring in the studio, as well as the controls and levels on consoles, DAWs, etc., \  
3618 in preparation for recording."),  
3619 quiz::Quiz("Shelf Filter","A name for the circuit in an equalizer used to obtain the\  
3620 shelf."),  
3621 quiz::Quiz("Shelf","A frequency response of an equalization circuit where the boost \  
3622 or cut of frequencies forms a shelf on a frequency response graph. A high-frequency \  
3623 shelf control affects signal levels at the set frequency and all frequencies above i\  
3624 t; a low-frequency shelf does the same for signals at and below the set frequency.")\  
3625 ,  
3626 quiz::Quiz("Shield","The outer conductive wrapping around an inner wire or wires in \  
3627 a cable, for the purpose of shielding the cable from picking up external electromagn\  
3628 etic interference."),  
3629 quiz::Quiz("Shielded Cable","Cable that has a shield around an inner conductor or in\  
3630 ner conductors."),  
3631 quiz::Quiz("Shock Mount","An elastic mount on microphone stand that reduces the impa\  
3632 ct of unwanted vibrations that may affect the stand (for example, floor vibrations f\  
3633 rom footsteps)."),  
3634 quiz::Quiz("Short Circuit","A direct connection between two points in a circuit that\  
3635 (usually) should not be connected."),  
3636 quiz::Quiz("Short Delay","Delay times under 20 milliseconds."),  
3637 quiz::Quiz("Shortest Path","A technique in recording that routes the signal through \  
3638 the least amount of active (amplified) devices during recording."),  
3639 quiz::Quiz("Shotgun Microphone","A microphone with a long line filter, a tube that a\  
3640 coustically cancels sound arriving from the side, to make the microphone pick up muc\  
3641 h better in one direction than in any other direction. This gives the shotgun mic a \  
3642 tight, hypercardioid pickup pattern. Shotgun microphones are commonly used to record\  
3643 dialogue in filming situations, usually held on a boom stand with a shock mount."),  
3644 quiz::Quiz("Sibilance","Energy from a voice centered around 7 kHz, caused by pronoun\  
3645 cing “s”, “sh” or “ch” sounds."),  
3646 quiz::Quiz("Sidechain","An auxiliary input to a signal processor that allows control\  
3647 of the processing to be triggered by an external source. A common use of sidechaini\  
3648 ng is in compressors, particularly in ducking effects where the presence of a partic\  
3649 ular audio signal triggers the compression of another audio signal. (See also “Ducki\  
3650 ng.”)"),
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3651 quiz::Quiz("Signal Flow","1) In the general sense, the path that an audio signal tra\
3652 vels from the sound source to the system output. (For example, from the vocalist's v\
3653 oice into the microphone, through the cables, into the preamp, out of the preamp int\
3654 o the console, through all inserts and buses, and output into the DAW for recording.\
3655 ) 2) Signal flow is often specifically meant to refer to the routing of an audio sig\
3656 nal through the console, from input to output."),
3657 quiz::Quiz("Signal Processing","The practice of altering the character or sound of a\
3658 n audio signal through a variety of devices or plug-ins, such as equalizers, compres\
3659 sors, reverb units, etc."),
3660 quiz::Quiz("Signal to Noise Ratio (SNR)","The comparison of the strength of a signal\
3661 level to the amount of noise emitted by the device, expressed in dB."),
3662 quiz::Quiz("Signal","1) In audio, an alternating current (or voltage) matching the w\
3663 aveform of, or being originally obtained from, a sound pressure wave. 2) Also in aud\
3664 io, an alternating current (or voltage) between 20 Hz and 20,000 Hz. 3) A digital au\
3665 dio bit stream."),
3666 quiz::Quiz("Sine Wave","1) In the general sense, the path that an audio signal trave\
3667 ls from the sound source to the system output. (For example, from the vocalist's voi\
3668 ce into the microphone, through the cables, into the preamp, out of the preamp into \
3669 the console, through all inserts and buses, and output into the DAW for recording.) \
3670 2) Signal flow is often specifically meant to refer to the routing of an audio signa\
3671 l through the console, from input to output."),
3672 quiz::Quiz("Sine","This is the purest waveform: It contains only the fundamental har\
3673 monic, and no higher harmonics. As a result, it's a great wave to use to create a su\
3674 b bass as well as a kick drum or other pure drum tone; it's also a great source wave\
3675 to use when exploring techniques such as frequency modulation (FM), amplitude modul\
3676 ation (AM), or wavefolding which add or shift harmonic content."),
3677 quiz::Quiz("Slap Echo (also called Slapback)","A single, distinct echo of a sound, w\
3678 hich can result naturally from higher frequencies reflecting off a non-absorbent wal\
3679 l, or artificially reproduced by a signal processing unit or plugin. Slap echo creat\
3680 es a "live" sounding effect similar to what you would hear in an arena."),
3681 quiz::Quiz("Slate","Slate (Slating) - 1) In video/film, the identification of a scen\
3682 e and take at the beginning of the clip for the purpose of video editing. This is do\
3683 ne by presenting the scene/take in written form in front of the camera on a clapboar\
3684 d, calling the scene/take verbally, then marking it audibly with the clapper for the\
3685 purpose of syncing audio to the video. 2) In audio recording, the similar practice \
3686 of identifying a take of music by an audible cue at the beginning of the recorded tr\
3687 ack. While some engineers still practice this, it was more necessary in the days of \
3688 analog tape recording because it helped editors keep track of the location of takes \
3689 on the recorder. Today, DAWs make it easier to keep track by identifying each take v\
3690 isually on the screen."),
3691 quiz::Quiz("Slave","1) In audio, any device which syncs to another device by reading\
3692 the clock information emitted by the master device. 2) In MIDI, any device or instr\
3693 ument that is being operated remotely by MIDI information sent from another device.")
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3694 ),
3695 quiz::Quiz("Slew Limiter","This function smoothes out an incoming signal so that the\
3696 change in voltage level cannot exceed a certain number of volts per second. As a re\
3697 sult, it is sometimes called a lag generator or processor, or more technically as an\
3698 integrator."),
3699 quiz::Quiz("Sliding Rails","This is a common system for mounting modules into a case\
3700 where the rails that the modules attach to contain channels rather than holes. A nu\
3701 mber of nuts are inserted into these channels, which can then be slid to any positio\
3702 n to accommodate the mounting hole spacing of your modules. In a Eurorack case, the\
3703 e nuts tend to have a 2.5mm or 3mm hole and corresponding thread."),
3704 quiz::Quiz("Slope Generator","A slope generator creates ramps: rising or falling vol\
3705 tages. It is essentially a gate generator and a slew limiter (see above) wired toget\
3706 her in the same module. A common example of a slope generator is an attack/decay (AD\
3707 ) or attack/release (AR) envelope generator. However, since it can be used for gener\
3708 alized control voltage functions – even creating a sawtooth or triangle wave oscilla\
3709 tor – some companies such as Buchla and Serge referred to by its elemental function \
3710 of generating sloping voltage changes."),
3711 quiz::Quiz("Slope","Most filters typically have a cutoff or corner frequency they ar\
3712 e tuned to. It then reduces (filters) the frequency spectrum of a signal going throu\
3713 gh it so that its harmonics get progressively quieter the further away they are from \
3714 this cutoff. The strength of this effect is referred to as its slope. Most filters h\
3715 ave slopes that are defined multiples of 6 decibels (dB) weaker for each octave furt\
3716 her away you get from the cutoff frequency. For example, a low-pass filter (LPF) wit\
3717 h a slope of 24 dB/octave would attenuate harmonics one octave above its cutoff freq\
3718 uency by 24 decibels."),
3719 quiz::Quiz("Smart FSK (Frequency-Shift Key)","Smart FSK – An updated form of Frequen\
3720 cy-Shift Key (FSK) sync that enables MIDI devices to sync to analog tape recorders a\
3721 nd/or other recording devices. A digital signal with MIDI Song Position Pointer (SPP\
3722 ) data is encoded onto a spare track, which identifies the exact bar, measure and be\
3723 at for MIDI sequencers/devices at any point in the recording. This enables the devic\
3724 e to start playing at exactly the right place and tempo no matter where you start th\
3725 e tape. (See also “Frequency-Shift Key.”)"),
3726 quiz::Quiz("SMPTE Time Code","(Abbreviated “SMPTE”) A standardized timing and sync s\
3727 ignal protocol created by the Society of Motion Picture and Television Engineers for\
3728 the purpose of syncing audio to video/film, which can also be used for syncing purp\
3729 oses in audio recording environments. Many audio professionals simply refer to this \
3730 time code as “SMPTE.”"),
3731 quiz::Quiz("SMPTE","1) Abbreviation for Society of Motion Picture and Television Eng\
3732 ineers. 2) See “SMPTE Time Code.”"),
3733 quiz::Quiz("Snare","1) Abbreviation for “snare drum.” 2) The metal strands stretched\
3734 across the bottom head of a snare drum, which help produce the piercing “cracking” \
3735 sound when the snare drum is struck."),
3736 quiz::Quiz("Sock Cymbal","A rarely used alternate term for “hi-hat,” left over from \
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3737 the days when hi-hat cymbals were placed at “sock level.” (See also “Hi-Hat.”)),
3738 quiz::Quiz("Soft Knee","In compression, refers to the gradual introduction of compre\
3739 ssion of the signal once the sound level crosses the threshold. (See also “Knee.”))\
3740 ,
3741 quiz::Quiz("Software Instrument (Virtual Instrument)","One of a number of software-b\
3742 ased synthesizers, samplers or sound samples that are stored and accessed via comput\
3743 er and performed by an external MIDI controller, rather than in a standalone synthes\
3744 izer or module. Because of the wide versatility available from these instruments, a \
3745 growing number of composers and electronic musicians are working with virtual instru\
3746 ments that can be stored in hard drives, rather than purchasing stacks of keyboards \
3747 and modules."),
3748 quiz::Quiz("Soldering","The action of making connections with solder, a soft metal a\
3749 lloy that is used to bond two metal surfaces by melting. In audio settings, solderin\
3750 g is used for a variety of purposes in building, modifying or repairing gear—perhaps\
3751 most often to repair or build audio cables as a cost-saving effort, as opposed to b\
3752 uying new ones or sending them off for repair."),
3753 quiz::Quiz("Solid State","In electronics, refers to the use of transistors and semic\
3754 onductors (solid materials) in the building of electronic devices, as opposed to tub\
3755 es. In the recording studio, solid state amplifiers have different properties than t\
3756 ube amps, and each has its own advantages and disadvantages. A more recent applicati\
3757 on of solid state construction is in computer devices, particularly solid state hard\
3758 drives (SSD), which transfer data more quickly than conventional spinning disc driv\
3759 es, and are less prone to breakage."),
3760 quiz::Quiz("Solo Switch","A switch that activates the solo function on a console or \
3761 DAW."),
3762 quiz::Quiz("Solo","1) A circuit in a console or DAW that allows one or more selected\
3763 channels to be heard or to reach the output, while other channels are automatically\
3764 muted. 2) In music, a segment of a song in which a vocalist or instrument is featur\
3765 ed above other instruments."),
3766 quiz::Quiz("Song Position Pointer (SPP)","A MIDI message that enables connected MIDI\
3767 devices to locate a given point in the song. Used in conjunction with MIDI clock as\
3768 a way of synchronizing devices or telling a connected device when to begin playing.\
3769 "),
3770 quiz::Quiz("Sound Blanket","A thick blanket that can be put on floors or hung to add\
3771 sound absorpction to the room, and help prevent sound reflections."),
3772 quiz::Quiz("Sound Effects (SFX)","Sounds other than dialogue, narration or music tha\
3773 t are added to audio, usually in the context of film/video."),
3774 quiz::Quiz("Sound File","A digital audio recording that can be stored in a computer \
3775 or on a digital storage medium (such as a hard disk)."),
3776 quiz::Quiz("Sound Modeling","A technique that recreates a sound without directly mod\
3777 eling the physical device. An example is additive synthesis, which uses a combinatio\
3778 n of sine waves and noise to recreate sounds."),
3779 quiz::Quiz("Sound Module","An electronic instrument (tone generator, synth or sample\
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3780 r playback unit) that has no playable interface, but instead responds to incoming MI\  
3781 DI message. Often sound modules were created as the “brains” of popular synthesizers\  
3782 , cheaper versions of the product that could be added to an existing MIDI configurat\  
3783 ion. Today, sound modules can also occur as software versions or plugins to be acces\  
3784 sed on a computer."),  
3785 quiz::Quiz("Sound Pressure Level (SPL)","In scientific/technical terms, the measure \  
3786 of the change in air pressure caused by a sound wave, measured in dB. We hear and pe\  
3787 rceive SPL in terms of amplitude, volume or loudness of the sound."),  
3788 quiz::Quiz("Sound Pressure Level","In scientific/technical terms, the measure of the\  
3789 change in air pressure caused by a sound wave, measured in dB. We hear and perceive\  
3790 SPL in terms of amplitude, volume or loudness of the sound."),  
3791 quiz::Quiz("Sound Source","The origin of a sound, whose vibrations create sound wave\  
3792 s."),  
3793 quiz::Quiz("Sound Wave","(Also called “Sound Pressure Wave”) A wave caused by a vibr\  
3794 ation that results in slight variations in air pressure, which we hear as sound."),  
3795 quiz::Quiz("Soundtrack","1) Broadly speaking, refers to any/all audio that accompani\  
3796 es an instance of visual media, whether music, dialogue or SFX. 2) In more common te\  
3797 rms, refers to the musical score and/or licensed music synced to a film, video, TV p\  
3798 rogram or video game."),  
3799 quiz::Quiz("Source of Uncertainty","This was the name for the Buchla 265 and 266 mod\  
3800 ules that create random control voltages. Its name is often used for random source m\  
3801 odules that follow or are inspired by the original Buchla template."),  
3802 quiz::Quiz("Spaced Pair","(Also called “A/B Technique“) A stereo microphone placemen\  
3803 t technique in which two cardioid or omnidirectional microphones are spaced somewher\  
3804 e between 3-10 feet apart from each other (depending on the size of the sound source\  
3805 ) to create a left/right stereo image."),  
3806 quiz::Quiz("Speaker","A device that converts electrical signals to sound; more techn\  
3807 ically, a transducer that changes an electrical audio signal into sound pressure wav\  
3808 es."),  
3809 quiz::Quiz("Speed of Sound","Generally speaking, the time it takes for a sound wave \  
3810 to travel through a medium. Sound travels at different speeds through solids, liquid\  
3811 s and gases, and though we usually think of sound as traveling through the air, diff\  
3812 erences in temperature, air pressure and humidity can also affect how fast sound tra\  
3813 vels. For a starting frame of reference, the speed of sound is generally defined by\  
3814 aerospace engineers as “Mach 1.0,” translating to 340.29 meters per second (approx\  
3815 761.1 mph, or 1116 feet per second), which is how fast sound travels through the ai\  
3816 r at sea level at a temperature of 15 degrees Celsius (59 degrees Fahrenheit). By co\  
3817 ntrast, at 70 degrees Fahrenheit under standard atmospheric conditions, the speed of\  
3818 sound is about 344 m/s, or 770 mph."),  
3819 quiz::Quiz("Splicing","Historically, the act of attaching previously cut pieces of a\  
3820 udio tape or film in precise locations by applying a special kind of adhesive tape o\  
3821 n the back. This is/was done for the purpose of shortening sections of audio or edit\  
3822 ing film. Today, splicing has become a very simple process by editing sections of au
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3823 dio or video digitally with a DAW or film editing software."),
3824 quiz::Quiz("Splitter","The short definition is something that can divide a signal in\
3825 to two or more copies, such as a splitter cable where two outputs are wired to one i\
3826 nput. For a deeper discussion, see the entry on multiple, as there are ways of going\
3827 about this beyond simple wiring."),
3828 quiz::Quiz("Spread","A few oscillator modules can produce more than one tone at the \
3829 same time. Slightly detuning or “spreading” these tones from each other creates an o\
3830 ften pleasing chorusing sound. Depending on the module, you might even be able to sp\
3831 read these tones to form intervals, triads, and chords."),
3832 quiz::Quiz("Spring Reverb","A device that simulates reverberation by creating vibrat\
3833 ions within a metal spring by attaching it to a transducer and sending the audio sig\
3834 nal through it. A pickup at the other end converts those vibrations into an electric\
3835 al signal which is mixed with the original audio signal. While the physical spring r\
3836 everbs still exist, most studios emulate spring reverb with the use of plug-ins or h\
3837 ardware reverb units."),
3838 quiz::Quiz("Square wave","This is a common waveform produced by a synthesizer’s osci\
3839 llator. It alternates between a high and low voltage (typically +/-5 or 8 volts for \
3840 an audio oscillator; sometimes low frequency oscillators go between 0v and a positiv\
3841 e voltage). Aside from being a really easy waveshape to generate with analog circuit\
3842 ry, it has an interesting harmonic series: it has a strong fundamental, then gradual\
3843 ly weaker odd harmonics: a component at three times the fundamental frequency, one a\
3844 t fives time the fundamental, and so forth. The result is a more open, hollow sound,\
3845 especially when compared to a sawtooth (ramp) wave that has both odd and even harmo\
3846 nics present. A wave shape in which the voltage rises instantly to one level, stays \
3847 at that level for a time, instantly falls to another level and stays at that level, \
3848 and finally instantly rises to its original level to complete the wave cycle."),
3849 quiz::Quiz("Stackable Cable","Many banana style cables are constructed that each plu\
3850 g has a jack built into its back, allowing you to plug another cable directly in top\
3851 of the original plug. These are used by Buchla and Serge-compatible systems. TipTop\
3852 makes a similar cable using 3.5mm plugs and jacks for Eurorack format users called \
3853 Stackables."),
3854 quiz::Quiz("Stage Monitor","A speaker on the stage that enables performers to hear t\
3855 hemselves and to hear what the other musicians are playing on stage."),
3856 quiz::Quiz("Stage","1) The partially enclosed or raised area where live musicians pe\
3857 rform. 2) In reverberation effects devices, an echo added before the reverberation t\
3858 o simulate echoes that would come from a concert stage. In the most general terms, a\
3859 stage is the next change in voltage among a series of changes. In an 8-step sequenc\
3860 er, for example, each new note that it produces in order is a stage. In an envelope \
3861 generator such as an ADSR (Attack/Decay/Sustain/Release), each phase – such as attac\
3862 k, where the envelope generally rises from 0 volts to the highest voltage it can out\
3863 put – is a stage. You might also hear it used to describe the number of sample stage\
3864 s in a BBD (Bucket Brigade Delay), described elsewhere."),
3865 quiz::Quiz("Standard Operating Level","A reference voltage level or maximum average \
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3866 level that should not be exceeded in normal operation."),
3867 quiz::Quiz("Standing Wave","An unwanted sound wave pattern that often occurs when th\
3868 e sound wave bounces between two reflective parallel surfaces in a room, and the ref\
3869 lected waves interfere with the initial wave coming from the sound source, in which \
3870 the combined wavelength of the affected frequency is effectively the length of the r\
3871 oom. This creates the audible illusion that the wave is standing still, so the frequ\
3872 ency is amplified to an unwanted level in certain parts of the room while nearly abs\
3873 ent in others. Standing waves are most common in square or rectangular rooms with pa\
3874 rallel surfaces, so acoustic designers try to prevent these waves by installing abso\
3875 rptive materials or introducing other items to offset the parallel surfaces."),
3876 quiz::Quiz("Step Mode","A setting in a sequencer or DAW in which notes are input man\
3877 ually, one note or step at a time."),
3878 quiz::Quiz("Step Sequencer","This usually refers to a type of sequencer where you st\
3879 ep to and pause on a stage, enter the note (and possibly the duration) for that stag\
3880 e, move on to the next step, and so forth."),
3881 quiz::Quiz("Step","Step is often used interchangeably with stage (see above), especia\
3882 lly when talking about sequencers."),
3883 quiz::Quiz("Stereo Image","The audible perception of stereo, in which different soun\
3884 ds sources appear to be coming from far left, far right or any place in between."),
3885 quiz::Quiz("Stereo Micing","Placement of two or more mics so that their outputs comb\
3886 ine to create a stereo image."),
3887 quiz::Quiz("Stereo","A recording or reproduction of at least two channels where posi\
3888 tioning of instrument sounds left to right can be perceived."),
3889 quiz::Quiz("Strike","This term appears on several Make Noise modules, although it ha\
3890 s been creeping into the general lingo. Some filters, amplifiers, and low pass gates\
3891 (LPGs) that use or simulate vactrols (a light sensitive resistor placed next to a li\
3892 ght source such as an LED, allowing a voltage to be turned into a resistance to con\
3893 trol a parameter) may have a strike input. When you flash an LED at a light sensitiv\
3894 e resistor, it does not change the resistance instantaneously and stay there - inste\
3895 ad, there is some delay as it glides to the desired resistance. When you turn the LE\
3896 D off, the resistance may not go instantaneously to full; instead it might take a br\
3897 ief moment to decay. These characteristics are useful for creating percussive sounds\
3898 and attacks. The purpose of a strike input is either to pass just a short pulse, or\
3899 to allow you to re-attack while the LED is otherwise still on. To put away equipmen\
3900 t and clean up after a recording session."),
3901 quiz::Quiz("Subcode","Additional information bits that are recorded alongside digita\
3902 l audio, used for control and playback purposes."),
3903 quiz::Quiz("Subframe","A unit smaller than one frame in SMPTE time code."),
3904 quiz::Quiz("Subgroup","A number of input channels on a console that can be controlle\
3905 d and adjusted as a single set before sending the combined signal to the master outp\
3906 ut. Sometimes also called "Submix," "Bus" or just "Group.""),
3907 quiz::Quiz("Subharmonic","A circuit that divides the fundamental harmonic of the inc\
3908 oming sound to produce lower frequencies, and therefore subharmonics. The most commo\
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3909 n is an octave divider or sub bass circuit that divides creates a subharmonic by div\  
3910 iding the fundamental by 2 (some can also create a subharmonic two octaves below the\  
3911 fundamental by dividing it by 4)."),  
3912 quiz::Quiz("Submaster / Sub-Master", "The fader which controls the combined level of \  
3913 sound from several channels during mixdown or recording."),  
3914 quiz::Quiz("Submix", "See “Subgroup.”"),  
3915 quiz::Quiz("Suboctave", "A module that creates a new tone one or two octaves below th\  
3916 e fundamental harmonic – the “pitch” – of the sound coming into it, to emphasize the\  
3917 bass. (Subharmonics are discussed in detail elsewhere in this glossary.) This tone \  
3918 is usually a square wave, although some clever modules may create something more sin\  
3919 e-like, or that more closely resembles the original waveform."),  
3920 quiz::Quiz("Subtractive Synthesis", "The most common synthesis technique: You start w\  
3921 ith one or more oscillators outputting waveforms with a large number of harmonics, a\  
3922 nd then pass this mix through a filter that removes some of the harmonics to create \  
3923 the desired sound or timbre. This modified tone is then sent to an amplifier that ad\  
3924 ds articulation to the note by varying its loudness. An old-school method of sound s\  
3925 ynthesis in which sounds are designed and created by generating harmonically rich wa\  
3926 veforms, then filtering out unwanted harmonics to arrive at the desired sound."),  
3927 quiz::Quiz("Sum", "To sum is a fancy way of saying you added two (or more) things tog\  
3928 ether; the sum is the result. It usually is used in the context of adding together c\  
3929 ontrol voltages, although it can also be used for audio or even mixes of harmonics. \  
3930 The opposite is difference, which subtracts one input from another. A signal that is\  
3931 the mix of the two stereo channels at equal level and in phase."),  
3932 quiz::Quiz("Summing", "The process of blending two or more signals into one mixed sig\  
3933 nal. In summing audio, each successive channel adds volume to the overall signal, so\  
3934 channels must be mixed in order to prevent peaking the combined signal."),  
3935 quiz::Quiz("Super-Cardioid Pattern", "A very tight cardioid microphone pattern with m\  
3936 aximum sensitivity on axis and the least amount of sensitivity approximately 150 deg\  
3937 rees off-axis."),  
3938 quiz::Quiz("Surround Sound", "A technique of recording and playback in which the list\  
3939 ener hears various aspects of the sound from front to back as well as side-to-side-a\  
3940 360-degree audio image, as opposed to the standard stereo left-right image. Surroun\  
3941 d sound can occur in various formats with different numbers of speakers arrayed thro\  
3942 ugh the room. Surround sound today is most commonly used in film and TV production."\  
3943 ),  
3944 quiz::Quiz("Sustain", "This is a common stage of an envelope generator where a voltag\  
3945 e – usually being sent to a filter’s cutoff frequency or an amplifier’s level – is b\  
3946 eing held a steady level while a note is still being held down. The knowledge that a\  
3947 note is being held is usually provided by a gate signal, that stays high as long as\  
3948 a note is held down, although some envelope generators may have a dedicated time co\  
3949 ntrol for how long the sustain stage should last. Envelopes that contain sustain sta\  
3950 ges include the ADSR (Attack/Decay/Sustain/Release) and AR (Attack/Release, which us\  
3951 ually assumes a sustain stage)."),
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3952 quiz::Quiz("SVF","A state variable filter (SVF) is a common design for synth filters\  
3953 . This design lends itself to allowing low pass, high pass, and bandpass all being a\  
3954 vailable simultaneously. Another side effect is that they are not prone to oscillati\  
3955 ng at high feedback (resonance) settings, although some have certainly figured out h\  
3956 ow to make this happen. The Oberheim SEM (Synthesizer Expander Module) filter is per\  
3957 haps the most famous state variable design."),  
3958 quiz::Quiz("Sweetening","A vague term referring to the fine-tuning of audio in the p\  
3959 ost-production stage of recording. Effectively, any small "tweaks" to to make the au\  
3960 dio sound better is considered sweetening."),  
3961 quiz::Quiz("Switch Trigger","Some systems - such as the original Moog modular - use \  
3962 an s-trigger (switch or shorting trigger) instead of a normal gate, which was a wire\  
3963 that was shorted to 0 volts ground, like the closing of a switch wired to ground. Y\  
3964 ou cannot interconnect these two systems without some form of conversion between the\  
3965 two, which can be as simple as a special cable."),  
3966 quiz::Quiz("Switch","A device that makes and/or breaks electrical connections."),  
3967 quiz::Quiz("Switchable Pattern Microphone","A microphone having the capability of tw\  
3968 o or more pickup patterns, which can be toggled by use of a switch on the microphone\  
3969 ."),  
3970 quiz::Quiz("Switching Power Supply","A switching power supply starts by directly con\  
3971 verting the incoming high-voltage AC signal into a high-voltage DC signal. They then\  
3972 rapidly switch that output on and off to average a lower output voltage. This swit\  
3973 ched voltage is then smoothed out to create a constant DC supply at the desired volta\  
3974 ge. Switching power supplies tend to be lighter, cooler, and less expensive, at the \  
3975 cost of often higher noise - both in the output voltage, and in radio frequencies (t\  
3976 his is why they are often surrounded by a shielding cage). Many are moving to a hybr\  
3977 id power supply that combines a switcher with a small linear supply or regulator to \  
3978 get the best of both worlds."),  
3979 quiz::Quiz("Sync Pop","A short tone (usually a sine wave at 1 kHz, and the length of\  
3980 a frame of film) that is placed exactly two seconds before the start of a piece of \  
3981 film or music. The sync pop is used to make sure that all related audio and video tr\  
3982 acks stay in sync with each other through all stages of post-production."),  
3983 quiz::Quiz("Sync24","Sync24 is an alternate name used for the Roland-created standar\  
3984 d DIN Sync, which sends a clock signal at the rate of 24 pulses per quarter note at \  
3985 the current tempo. Korg equipment used a variation of this running at 48 pulses per \  
3986 quarter note, also known as Sync48."),  
3987 quiz::Quiz("Sync","Sync can have two different meanings, depending on whether we're \  
3988 talking about oscillators or about clock signals. Some oscillators support a mode wh\  
3989 ere they reset their waveshapes to the beginning when they receive a signal from ano\  
3990 ther oscillator. If there is not a precise octave relationship between the two oscil\  
3991 lators, the result is a modified waveform that has been reset prematurely, following\  
3992 the frequency of the second oscillator. You can create some very cool "ripping" sou\  
3993 nds by modulating the frequency of the slave oscillator; a simple AD envelope works \  
3994 well. In the context of timing, when you are synchronizing sequencers or drum patter\
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3995 ns, it is common to send a master timing or sync signal around the modular for all t\  
3996 he relevant modules to follow. This is typically a gate or trigger signal. Short for\  
3997 “Synchronization.” In audio/studio settings, sync refers to the correlating of two \  
3998 or more pieces of audio or video in relation to each other. This can include syncing\  
3999 two recording/playback devices timed to a sync signal like SMPTE Time Code, synchro\  
4000 nizing audio with video in film or TV, and many other examples. Licensing a song or \  
4001 piece of music for placement in film, TV or video is also referred to as “syncing.”\  
4002 ),  
4003 quiz::Quiz("Synthesizer Expander Module","The Oberheim SEM (Synthesizer Expander Mod\  
4004 ule) was one of their earliest products. It was an entire synthesizer voice – two os\  
4005 cillators, two simple envelopes, VCA, and a very popular two-pole state variable fil\  
4006 ter design with a knob that crossfaded between low pass, notch, and high pass output\  
4007 s plus a separate bandpass setting – in a cube-like case. Most often today, when a m\  
4008 odular manufacturer uses the magic letters \"SEM\", they're referring to a filter me\  
4009 ant to emulate that in the original Oberheim synth."),  
4010 quiz::Quiz("Synthesizer","A musical instrument that uses electrical oscillators to g\  
4011 enerate tones artificially, either to simulate the sounds of other instruments or to\  
4012 create other sounds not possible with other instruments."),  
4013 quiz::Quiz("System Exclusive","System Exclusive (SysEx) – A MIDI message that will o\  
4014 nly be recognized by a unit of a particular manufacturer."),  
4015 quiz::Quiz("Tach/Tachometer","In analog tape recording, a device on the recorder tha\  
4016 t measures and regulates tape speed by emitting pulses as the tape moves across the \  
4017 head."),  
4018 quiz::Quiz("Tails Out","A method of winding audio tape so that the end of the last r\  
4019 ecorded selection is at the outside of the reel."),  
4020 quiz::Quiz("Take Notation","Writing down the takes of the tune being recorded on a t\  
4021 ake sheet or on the track log with comments. Take notation was/is recommended for an\  
4022 alog tape recording, but in most studios, this function is now accomplished on the D\  
4023 AW."),  
4024 quiz::Quiz("Take","The recording that is done between one start and stop of a tape r\  
4025 ecorder or DAW."),  
4026 quiz::Quiz("Talk Box","An effects unit that enables a musician to modulate the sound\  
4027 of his/her instrument via a tube placed into the mouth. Historically, talk boxes ha\  
4028 ve been used as an effect for guitars, but they can be used to modify other instrume\  
4029 nts, as well."),  
4030 quiz::Quiz("Talkback","A microphone in the control room carried on a separate circui\  
4031 t from the recorded channels, allowing the engineer to communicate with the musician\  
4032 s in the live room or sound booths through the monitoring system."),  
4033 quiz::Quiz("Tape Delay","A signal processing technique for creating artificial delay\  
4034 or echoes by manipulating time delays with analog tape machines. This technique beg\  
4035 an by routing the signal to a separate tape recorder and mixing the delayed response\  
4036 back in with the signal; it then evolved to the use of dedicated machines that coul\  
4037 d adjust the length of the delay by adjusting the distance between the record and pl\
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4038 ayback heads. Today, most tape delay effects in the studio are simulated digitally t\  
4039 hrough plug-ins in a DAW."),  
4040 quiz::Quiz("Tape Guide","Any stationary or rotating device which directs the tape pa\  
4041 st the heads on a tape machine, or from one reel to the other."),  
4042 quiz::Quiz("Tape Hiss","The natural high-frequency noise that occurs on analog tape \  
4043 due to the magnetic particles from which the tape is made. Tape hiss constitutes mos\  
4044 t of the noise floor that occurs in analog recording, and can be reduced by using ta\  
4045 pe constructed of finer magnetic particles. (See also "Noise Floor.")"),  
4046 quiz::Quiz("Tape Loop","A length of tape with the ends spliced together so that the \  
4047 recording will play continuously."),  
4048 quiz::Quiz("Tape Recording Equalization","The increase in amplitude of signals, in a\  
4049 tape machine's electronics, at the high frequencies as a tape is recorded to keep h\  
4050 igh-frequency signals recorded above the tape hiss."),  
4051 quiz::Quiz("Telephone Filter","A filter used to simulate the audio heard through a t\  
4052 elephone receiver by removing signals at frequencies below 300 Hz and above 3500 Hz.\  
4053 "),  
4054 quiz::Quiz("Tempo Mapping","The act of programming a sequencer or DAW to follow the \  
4055 tempo variations of a recorded performance. Unlike beat mapping or beatmatching, bot\  
4056 h of which effectively adjust the recording to fit a set tempo, tempo mapping adjust\  
4057 s the tempo of the project (especially the MIDI instruments) to match the natural te\  
4058 mpo nuances of the recorded material. (See also "Beat Mapping," "Beatmatching.")"),  
4059 quiz::Quiz("Tempo","The rate at which the music moves, measured in Beats Per Minute \  
4060 (BPM)."),  
4061 quiz::Quiz("Terminal","1) A point of connection between two wires, including the plu\  
4062 g on the end of a cable, and the jack on a piece of equipment. 2) Refers to the keyb\  
4063 oard and monitor of a computer that enable the user to enter information and to acce\  
4064 ss data."),  
4065 quiz::Quiz("Test Oscillator","A device that generates audio waveforms at various fre\  
4066 quencies for testing purposes."),  
4067 quiz::Quiz("Test Pressing","One of a few initial vinyl record copies pressed from th\  
4068 e first stamper made, which is listened to and visually inspected to approve the qua\  
4069 lity before more copies are pressed."),  
4070 quiz::Quiz("Test Tones","1) A recording of several single-frequency tones at the beg\  
4071 inning of a tape reel at the magnetic reference level that will be used to record th\  
4072 e program. 2) Artificially generated tones that are used to calibrate an audio syste\  
4073 m."),  
4074 quiz::Quiz("Thin Sound","A vague term describing an audio signal that that is lackin\  
4075 g in certain frequencies, especially on the low end. Over-filtering a signal with an\  
4076 EQ can produce a thin sound, for example."),  
4077 quiz::Quiz("Threaded Inserts","A common system for mounting modules into a case is c\  
4078 alled sliding rails or nuts. A number of nuts are inserted into these channels, whic\  
4079 h can then be slid to any position to accommodate the mounting whole spacing of your\  
4080 modules. Some don't like this system, so they replace the nuts with strip of metal \  

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4081 inserted into the channel that have been pre-drilled for the standard Eurorack mount\  
4082 ing hole spacing. They may be drilled for 2.5 or 3 mm screws; pay attention when buy\  
4083 ing the rails or a case that has them pre-installed."),  
4084 quiz::Quiz("Three-To-One Rule","A principle of microphone placement that says when m\  
4085 ultiple mics are used at once, the distance between microphones should be at least t\  
4086 hree times the distance between each microphone and its respective sound source. The\  
4087 three-to-one rule is used to prevent phasing issues between the audio signals."),  
4088 quiz::Quiz("Three-Way Speaker","A speaker system that has separate speakers to repro\  
4089 duce the bass, mid-range and treble frequencies."),  
4090 quiz::Quiz("Threshold of Hearing","Described as the sound pressure level at which pe\  
4091 ople can hear only 50 percent of the time."),  
4092 quiz::Quiz("Threshold","A threshold is generally a voltage level a signal needs to c\  
4093 ross before a module takes an action. For example, when the output of an envelope fo\  
4094 llower (a module that creates a voltage that corresponds to the current level of an \  
4095 audio signal) rises above a threshold level, then its gate signal will go high indic\  
4096 ating a note has started. When the output of the envelope follower falls below a th\  
4097 reshold (which may be the same or different than the note-on threshold), then the ga\  
4098 te goes low, indicating the note should be finishing. The level at which a dynamics \  
4099 processing unit will begin to change the gain of the incoming signal."),  
4100 quiz::Quiz("Throat","In a speaker, the small opening in a horn or in a driver throug\  
4101 h which the sound wave passes from the driver to the horn."),  
4102 quiz::Quiz("Through-Zero Frequency Modulation","TZFM is the abbreviation for Throug\  
4103 -Zero Frequency Modulation. Think of a patch where you feed the output of one oscill\  
4104 ator (the modulator) into the frequency control voltage input of a second oscillator\  
4105 (the carrier). As the waveform output of the modulator rises above zero volts, it i\  
4106 s added to the normal pitch control voltage for the carrier, and the pitch of the ca\  
4107 rrier goes up. As the waveform output of the modulator goes below zero, it is subtra\  
4108 cted from the normal pitch control voltage, and the pitch goes down. But what happen\  
4109 s if the result of subtracting the modulator from the pitch control goes below zero \  
4110 volts? In an oscillator that explicitly says it implements through-zero frequency mo\  
4111 dulation, the carrier will start playing backwards – in essence, a negative frequenc\  
4112 y. This generally produces a more pleasing result, and is a desirable characteristic\  
4113 for an oscillator."),  
4114 quiz::Quiz("Throw","1) In speakers and in microphones, describes the amount of unres\  
4115 tricted movement that the diaphragm can make. In microphone, this affects the mic's \  
4116 sensitivity; in speakers, it affects the distance of sound projection. (A speaker de\  
4117 signed for smaller spaces has a "short throw," while one designed for a farther proj\  
4118 ection has a "long throw." 2) In speakers, "throw" may also be used to describe the \  
4119 speaker's directional output, often based on the frequencies it emits. A horn, for e\  
4120 xample, emits high frequencies in a limited angle of direction, so it has a "long th\  
4121 row," while a subwoofer emits low frequencies in all directions and has a "short thr\  
4122 ow." 3) Something a producer, engineer or musician might do with whatever is in his\  
4123 her hand during a moment of intense frustration."),
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4124 quiz::Quiz("Tie Lines","Tie Lines – Cables with connectors at both ends, which are u\
4125 sually run through walls or floors in the studio, for the purpose of sending signals\
4126 between rooms. Tie lines provide a great semi-permanent way to route and configure \
4127 signal paths quickly through various parts of the studio and help the engineer keep \
4128 track of signal flow."),
4129 quiz::Quiz("Timbre","This word is often used to describe the unique tonal characteri\
4130 stic of a sound you are creating, separate from its pitch or loudness. Different sou\
4131 nds, by definition, have different timbres. When you change a parameter of a sound t\
4132 hat changes its tonal characteristic – such as changing the filter cutoff, pulse wid\
4133 th, amount of wavefolding, etc. – you are changing its timbre. The timbre often chan\
4134 ges during life of a note. The sound quality that makes one instrument sound differe\
4135 nt from other instruments, even while playing the same pitch. The timbre of a trumpe\
4136 t, for example, is what makes it sound like a trumpet and not like a flute. Timbre i\
4137 s largely shaped through the presence, absence and complexity of harmonics when the \
4138 instrument is played."),
4139 quiz::Quiz("Time Code","A standardized timing signal used to help devices sync with \
4140 one another, or to sync audio to video. Common time codes used in the studio are MID\
4141 I Time Code (MTC) and SMPTE time code."),
4142 quiz::Quiz("Time Compression / Expansion","(Also called “Time Stretching” or “Time S\
4143 hifting“) The process of speeding up or slowing down an audio recording without chan\
4144 ging the pitch of the sounds."),
4145 quiz::Quiz("Time Constant","A complex mathematical ides that basically describes the\
4146 time delay between when an electrical voltage is applied to a circuit and when the \
4147 circuit responds to it."),
4148 quiz::Quiz("Tini-Jax","This is a special design of jack made by Switchcraft that is \
4149 used by Buchla (and many of their clones) to carry audio signals. They are 3.5mm in \
4150 diameter, but differ slightly physically from a common 3.5 mm jack. 1/8” plugs would\
4151 be loose in when plugged into a Tini-Jax jack; a Tini-Jax plug might not fit into o\
4152 r might even damage a 1/8” jack."),
4153 quiz::Quiz("Toms","The small drums (as little as 10 inch diameter) that mount on rac\
4154 ks above the kick drum and the large drums in a drum set."),
4155 quiz::Quiz("Tone Generator","1) A device that puts out test tones at various frequen\
4156 cies to align a tape machine or for other testing purposes. 2) The circuits in a syn\
4157 thesizer that create the audio signals put out by the unit, usually to emulate the s\
4158 ound of another instrument."),
4159 quiz::Quiz("Tone","1) Any single-frequency signal or sound. 2) The sound quality of \
4160 an instrument’s sound relative to the amount of energy present at different frequenc\
4161 ies."),
4162 quiz::Quiz("Tonguing","The technique of controlling the start of a note in a brass o\
4163 r woodwind instrument with the tongue."),
4164 quiz::Quiz("Total Harmonic Distortion (THD)","The measure of the difference between \
4165 the level of harmonic frequencies at the output stage of an amplifier as compared wi\
4166 th the input stage, a ratio expressed as a percentage. It’s a fine-tuning specificat\
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4167 ion barely perceptible to many ears, but the lower the THD, the more accurately the \
4168 amplifier/speaker is reproducing the sound."),
4169 quiz::Quiz("Touch Sensitive","See "Velocity Sensitive.""),
4170 quiz::Quiz("Track & Hold","This is a variation of a Sample & Hold. Both have two inp\
4171 uts - a gate signal, and a voltage reference signal - and a voltage output. When a S\
4172 ample & Hold receives a gate high signal, it freezes and outputs the voltage referen\
4173 ce coming into the reference input. This voltage is maintained until a new gate high\
4174 signal; gate low signals are ignored. With a Track & Hold, when the gate is high, t\
4175 he reference input it passed along to the voltage output (this is the "tracking" pha\
4176 se); when the gate goes low, the input voltage at that instant is frozen and maintai\
4177 ned at the voltage output until a new gate high signal is received."),
4178 quiz::Quiz("Track Log / Track Assignment Sheet","Track Log/Track Assignment Sheet - \
4179 A sheet of paper kept with a multitrack tape which tells which instrument was record\
4180 ed on each track."),
4181 quiz::Quiz("Track","1) One audio recording made on a portion of the width of a multi\
4182 track tape, or created as a digital representation using a DAW. 2) One set of contro\
4183 l commands in a sequencer or DAW that is used to control one instrument over one MID\
4184 I channel. 3) See "Band Track.""),
4185 quiz::Quiz("Tracking","Tracking usually refers to how well an oscillator follows the\
4186 pitch control voltage (CV) sent to it. As the voltage rises, the oscillator "tracks\
4187 " it and produces a higher pitch. Most (but not all!) synths follow a 1 volt per oct\
4188 ave system where a rise of 1.00 volts on the pitch input should produce exactly a do\
4189 ubling (one octave rise) in the oscillator's pitch. If this is indeed what happens, \
4190 the oscillator has good tracking. If the oscillator goes slightly out of tune, it is\
4191 considered a tracking error, or to have poor tracking. Sometimes you will find volt\
4192 age-controlled filters have a "tracking" switch for a CV input where the pitch of th\
4193 e filter's corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding chan\
4194 ge of the pitch input. This can prevent high notes from sounding too bright without \
4195 the bass notes sounding too dull. Sometimes you will find voltage-controlled filters\
4196 have a "tracking" switch for a CV input where the pitch of the filter's corner freq\
4197 uency only rises at 1/3, 1/2, or 2/3 of the corresponding change of the pitch input.\
4198 The act of recording the individual tracks of a multitrack recording."),
4199 quiz::Quiz("Transducer","A device that converts energy from one medium to another. T\
4200 ransducers are prevalent throughout the equipment in a recording studio."),
4201 quiz::Quiz("Transient","The initial high-energy peak at the beginning of a waveform,\
4202 such as one caused by the percussive action of a pick or hammer hitting a string, o\
4203 r the strike of a drum."),
4204 quiz::Quiz("Transistor Ladder Filter","This term is often used to describe the desig\
4205 n of the much-loved Moog low-pass filter, which is still held up by many as being th\
4206 e gold standard in low pass filter sound. Moog actually received a patent for this d\
4207 esign (it has since expired); many of their competitors either outright copied it or\
4208 did their best to emulate it."),
4209 quiz::Quiz("Transport","1) The portion of a tape machine that moves the tape from th\
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4210 e supply reel, past the heads, to the take-up reel. 2) The set of controls found on \
4211 a DAW or sequencer for starting, stopping pausing, fast-forward and rewind, emulatin\
4212 g the functions of a tape machine transport."),
4213 quiz::Quiz("Transpose","In the simplest terms, to transpose the pitch of a musical l\
4214 ine is to shift it up or down by a fixed number of semitones or octaves. This is som\
4215 etimes referred to as “chromatic” transposition. A more sophisticated variation is “\
4216 scalar” transposition where each note is shifted by a set number of scale steps; thi\
4217 s differs from chromatic transposition because some scales may have differing number\
4218 s of semitones between steps than other scales. To shift a set of musical notes by a\
4219 fixed interval. This can happen in a number of ways—for example: 1) by rewriting an\
4220 entire piece of music in a new key; 2) by shifting the tuning of an instrument so t\
4221 hat it plays at a lower or higher interval than the note played (either artificially\
4222 , as with an electronic keyboard, or by the natural tuning of a transposed instrumen\
4223 t, like a trumpet); or 3) Transposing on-the-fly, playing at a set interval above or\
4224 below what is written (also known as transposing by sight)."),
4225 quiz::Quiz("Trap","1) A filter designed to reject audio signals at certain frequenci\
4226 es. 2) An object designed with acoustically absorptive material, placed into walls t\
4227 o reduce low frequency reflections in the room (also called “bass trap”). 3) Another\
4228 word for a drum set (as in “trap set”)."),
4229 quiz::Quiz("Tremolo","This is the effect of varying the amplitude (loudness) of a no\
4230 te. A way to create this effect on a modular synth is to patch a low frequency oscil\
4231 lator (LFO) to one of the control voltage inputs on an amplifier. Tremolo is differe\
4232 nt than vibrato; the latter is a warbling in pitch rather than loudness. A wavering \
4233 or “shaking” musical effect, created either by quick reiterations of the notes (as i\
4234 n a violin tremolo) or by rapid shifts in amplitude."),
4235 quiz::Quiz("Triangle","The triangle is a common synthesizer waveform. When selected \
4236 for the output of an oscillator, it was a more mellow sound than the standard square\
4237 or sawtooth waves, with fewer and weaker higher harmonics. It is also a popular out\
4238 put for low frequency oscillators (LFOs), as it produces a relatively smooth up and \
4239 down variation in whatever it controls, while being easier to create than the even s\
4240 moother sine wave."),
4241 quiz::Quiz("Triangular Wave","A harmonically rich waveform that appears triangular i\
4242 n shape when depicted graphically, due to a combination of the presence of odd harmo\
4243 nics and rapid rolloff."),
4244 quiz::Quiz("Trigger","A trigger is a very short electrical pulse signal, rising from\
4245 0 volts to a standard level such as 5 or 10 volts for a few milliseconds before fal\
4246 ling back to 0 volts. It is often used to start or “trigger” the playback of a percus\
4247 sion sound, including starting an envelope generator. They can also be used to pass\
4248 clock signals around a synth so connected modules all know when a note (or finer su\
4249 bdivision of a note) starts. A trigger usually has a fixed duration, compared to a g\
4250 ate signal which also rises from 0 volts to a higher voltage and falls back to zero \
4251 again, but which stays “high” a variable length of time depending on the length of a\
4252 note. The signal or the action of sending a signal to control the start of an event\
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4253 ."),
4254 quiz::Quiz("Trim / Trim Control","A device that reduces or increases the signal stre\
4255 ngth in an amplifier, often over a restricted range. Often used interchangeably with\
4256 gain, but usually referring to fine-tuning signal strength, rather than merely ampl\
4257 ifying it."),
4258 quiz::Quiz("Truncation","1) The shortening of an audio signal, sample or song, typic\
4259 ally by cutting off the end. 2) The dropping of bits of data when the bit resolution\
4260 is reduced (for example, from 24-bit to 16-bit), causing digital distortion unless \
4261 dithering is applied."),
4262 quiz::Quiz("Tune","The act of adjusting the pitch of a synthesizer's oscillator (the\
4263 main pitch-generating element) to match another oscillator, instrument, or referenc\
4264 e is known as tuning it."),
4265 quiz::Quiz("Tuning Fork","A metal fork with two prongs that vibrate with a fairly pu\
4266 re tone of one frequency when the fork is struck."),
4267 quiz::Quiz("Turntable","A device to support and rotate a phonograph record during pl\
4268 ayback."),
4269 quiz::Quiz("Tweeter","A speaker designed to reproduce only the higher frequencies of\
4270 the sound."),
4271 quiz::Quiz("Two Quadrant Multiplier","A two-quadrant multiplier performs a simple ve\
4272 rsion of amplitude modulation (AM), where that varies the amplitude or loudness of o\
4273 ne signal known as the carrier (typically an audio signal, swinging both above and b\
4274 elow 0 volts) with a second signal called the modulator. In the typical amplitude mo\
4275 dulation (AM) scenario, a low frequency oscillator with a positive voltage (say, bet\
4276 ween 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into th\
4277 e control input of a voltage controlled amplifier to add vibrato to an audio signal \
4278 passing through it. Any negative swings in the modulation signal are ignored; when p\
4279 atching tremolo, you may need to make sure an offset voltage is being added to your \
4280 LFO to make sure the sound doesn't cut out on the lower excursions of the LFO's wave\
4281 form. (The case where the modulator's negative as well as positive excursions are us\
4282 ed is referred to as a four quadrant multiplier.) "),
4283 quiz::Quiz("Two-Way Speaker","A speaker system with separate speakers to reproduce t\
4284 he lower frequencies (woofer) and the higher frequencies (tweeter)."),
4285 quiz::Quiz("TZFM","TZFM is the abbreviation for Through-Zero Frequency Modulation. T\
4286 hink of a patch where you feed the output of one oscillator (the modulator) into the\
4287 frequency control voltage input of a second oscillator (the carrier). As the wavefo\
4288 rm output of the modulator rises above zero volts, it is added to the normal pitch c\
4289 ontrol voltage for the carrier, and the pitch of the carrier goes up. As the wavefor\
4290 m output of the modulator goes below zero, it is subtracted from the normal pitch co\
4291 ntrol voltage, and the pitch goes down. But what happens if the result of subtractin\
4292 g the modulator from the pitch control goes below zero volts? In an oscillator that \
4293 explicitly says it implements through-zero frequency modulation, the carrier will st\
4294 art playing backwards - in essence, a negative frequency. This generally produces a \
4295 more pleasing result, and is a desirable characteristic for an oscillator."),
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4296 quiz::Quiz("U", "Rack-mounted equipment usually follows a standard set of dimensions,\  
4297 including 19" (48.3 cm) for width, and a "rack unit" (or U for short) for height equ\  
4298 aling 1.75" (4.4 cm) per U. Many common modular synthesizer formats follow the rack \  
4299 unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25" or 13.3 c\  
4300 m) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometime\  
4301 s referred to as MU for Moog Unit)."),  
4302 quiz::Quiz("Unbalanced Audio", "Most audio signals are passed around on cables with t\  
4303 wo wires: one for the voltage that represents the audio vibrations, and one for grou\  
4304 nd. This arrangement is often referred to as unbalanced audio."),  
4305 quiz::Quiz("Unbalanced Cable", "A cable with two conductors (a signal wire and a grou\  
4306 nd wire) and connectors on each end. Unbalanced cables are often susceptible to elec\  
4307 tromagnetic interference and noise. Examples of unbalanced cables are guitar/instrum\  
4308 ent cables (also called tip-sleeve or TS cables) and RCA cables."),  
4309 quiz::Quiz("Unidirectional Pattern", "A microphone pick-up pattern which is more sens\  
4310 itive to sound arriving from one direction than from any other."),  
4311 quiz::Quiz("Unipolar", "Many voltages in a modular synth – including the output of an\  
4312 audio oscillator, and most low frequency oscillators – fluctuates between positive \  
4313 and negative voltages. This is known as a bipolar voltage. Some voltages – such as t\  
4314 he output of an envelope generator – only vary between 0 volts and some maximum posi\  
4315 tive voltage; this is referred to as unipolar."),  
4316 quiz::Quiz("Unison", "Several performers, instruments or sound sources that are sound\  
4317 ing at the same time and with the same pitch."),  
4318 quiz::Quiz("Unity Gain", "The scenario in which there is no increase or decrease in s\  
4319 ignal strength at the output of an amplifier or device compared to the signal streng\  
4320 th at the input (typically described as 0 dB)."),  
4321 quiz::Quiz("Unity", "Usually used in the phrase "unity gain" this mean a signal keeps\  
4322 the exact same level from input to output."),  
4323 quiz::Quiz("Vacuum Tube", "A diode, a glass tube with the gases removed, through whic\  
4324 h electrical current can flow. In audio, vacuum tubes are used in amplifiers, oscill\  
4325 ators, and other analog devices."),  
4326 quiz::Quiz("Vamp and Fade", "A method of ending the recording of a song where the mus\  
4327 ic has a repeating part and the engineer reduces volume until the music fades out.")\  
4328 ,  
4329 quiz::Quiz("Vamp", "A part of a song or chord progression that is repeated, usually a\  
4330 t the end of the song, and usually the chorus or part of the chorus."),  
4331 quiz::Quiz("Vari-Speed", "A control on a tape machine that changes the play speed."),  
4332 quiz::Quiz("Variable-D", "A trademarked, patented technology of ElectroVoice in its m\  
4333 icrophone designs to vary the proximity effect in its microphones. Variable-D places\  
4334 several ports along the microphone body, each of which has a reduced level of sensi\  
4335 tivity to higher frequencies the further they are placed from the microphone's diaph\  
4336 ragm."),  
4337 quiz::Quiz("VCA Automation", "A system of mix automation in some mixing consoles in w\  
4338 hich sound levels or other functions are altered through the use of voltage controll\  

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4339 ed amplifiers."),
4340 quiz::Quiz("VCA Group","Several VCA faders that are fed control voltages from a grou\
4341 p master slide. A feature in higher-end mixing boards that enables the engineer to c\
4342 ontrol groupings of independent signals by a single fader that uses VCA to adjust th\
4343 e voltage sent to each channel."),
4344 quiz::Quiz("Velocity Message","In synthesizers and keyboard controllers, a MIDI mess\
4345 age that transmits data on how hard the key was struck. Velocity messages can be use\
4346 d to transmit volume information, as well as triggering different samples on a multi\
4347 -sampled instrument patch."),
4348 quiz::Quiz("Velocity Microphone","See “Pressure-Gradient Microphone.””),
4349 quiz::Quiz("Velocity Sensitive","(Also called “Touch Sensitive“) A feature on a MIDI\
4350 instrument such as a keyboard that transmits a MIDI velocity message depending on h\
4351 ow hard the key is struck."),
4352 quiz::Quiz("Vibrato","A smooth and repeated changing of the pitch up and down from t\
4353 he regular musical pitch, often done by singers or performed by string and wind play\
4354 ers."),
4355 quiz::Quiz("Virtual Instrument","(Also called Software Instrument) One of a number o\
4356 f software-based synthesizers, samplers or sound samples that are stored and accesse\
4357 d via computer and performed by an external MIDI controller, rather than in a standa\
4358 lone synthesizer or module. Because of the wide versatility available from these ins\
4359 truments, a growing number of composers and electronic musicians are working with vi\
4360 rtual instruments that can be stored in hard drives, rather than purchasing stacks o\
4361 f keyboards and modules."),
4362 quiz::Quiz("Vocal Booth","A room in the recording studio that is used for recording \
4363 vocals in isolation. This practice prevents bleed-through of the sounds of other ins\
4364 truments into the vocal microphone, and also reduces natural ambience and reverberat\
4365 ion in the vocal recording."),
4366 quiz::Quiz("Vocoder","An audio processing device effects device or plug-in that anal\
4367 yzes the characteristics of an audio signal and uses them to affect another synthesi\
4368 zed signal. Primarily developed for the purpose of producing synthesized voice effec\
4369 ts from human speech, a vocoder creates the characteristic robotic vocal sound or th\
4370 e “human synthesizer” effect that makes it sound like the synth is speaking or singi\
4371 ng words."),
4372 quiz::Quiz("Voice Over","The recording of vocal announcements or narration over a be\
4373 d of music in video, film or commercials."),
4374 quiz::Quiz("Voice","1) Besides the obvious definition of the sound humans make from \
4375 their mouths...in synthesizers, a voice refers to one of a number of sounds/pitches th\
4376 at may be played at the same time. “Monophonic” means only one voice plays at a time\
4377 , while “polyphonic” means multiple voices can sound at once. (See also “Polyphonic”\
4378 , “Monophonic.”) 2) In some synthesizers, like Yamaha, “voice” may also refer to a s\
4379 pecific sound patch available on the synth."),
4380 quiz::Quiz("Volatile Memory","Computer memory whose data will will be lost when the \
4381 computer is turned off. RAM (Random Access Memory) is the most common form of volati\
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4382 le memory."),
4383 quiz::Quiz("Voltage Controlled Amplifier (VCA)","An amplifier whose gain level is af\
4384 fected by an external voltage being sent to it. VCAs are commonly used in synthesize\
4385 rs, signal processors, and as a means of automation for some mixing consoles."),
4386 quiz::Quiz("Voltage Controlled Filter","A filter (especially a low-pass filter) that\
4387 will change its cutoff frequency according to a control voltage fed to its control \
4388 input."),
4389 quiz::Quiz("Voltage Controlled Oscillator (VCO)","An oscillator whose frequencies ar\
4390 e modified by voltage input. Most commonly found in synthesizers."),
4391 quiz::Quiz("Voltage","The difference in electrical force or pressure ("potential") b\
4392 etween two objects, causing a flow of electric current between them."),
4393 quiz::Quiz("Volume Unit (VU)","A unit to measure perceived loudness changes in audio\
4394 . The unit is basically the decibel change of the average level as read by a VU Mete\
4395 r. (See also "VU Meter.")"),
4396 quiz::Quiz("Volume","A common, non-technical term that either refers to sound pressu\
4397 re level (which we hear as loudness), or to audio voltage level."),
4398 quiz::Quiz("Vox","A Latin word meaning "voice," often used as an abbreviation for tr\
4399 ack logs in the studio."),
4400 quiz::Quiz("VU Meter","A meter that reads audio voltage levels in or out of a piece \
4401 of equipment and is designed to match the ear's response to sudden changes in level.\
4402 "),
4403 quiz::Quiz("Watt","Unit of electrical power."),
4404 quiz::Quiz("Wave","This is the pattern of vibrations - up and down fluctuations in v\
4405 oltage - output by an oscillator. Different patterns generate different sounds."),
4406 quiz::Quiz("Wavefolder","A wavefolder is a very specific design of waveshaper that u\
4407 ses a comparator and some other circuitry. What they do is look to see if the wave g\
4408 oes above (or below) a specific threshold. When it does, instead of clipping off the\
4409 top and bottom of the wave, they create a mirror image of it and reflect that porti\
4410 on of the wave back upon itself, creating more high harmonics and interesting spectr\
4411 a in the process."),
4412 quiz::Quiz("Waveform","This is the pattern of vibrations - up and down fluctuations \
4413 in voltage - output by an oscillator. Different patterns generate different sounds. \
4414 A visual representation or graphic of a sound wave, audio signal or other type of wa\
4415 ve, showing the wave's oscillations above and below the zero line."),
4416 quiz::Quiz("Wavelength","The physical length of one cycle of a wave, measured in fee\
4417 t, inches, etc. The longer the wavelength of a sound wave, the lower its frequency; \
4418 the shorter the wavelength, the higher the frequency."),
4419 quiz::Quiz("Waveshaper","It would be a bit obvious to say "a circuit that changes th\
4420 e shape of the waveform going through it", but that is the point. Waveshapers often \
4421 have specific goals in mind, such as converting an incoming triangle wave into an ou\
4422 tgoing sine wave, or to add tube-like soft clipping to the peaks and transients of w\
4423 aves. Many waveshapers are simply intended to mangle (er, add higher harmonics to) w\
4424 aveforms in interesting ways, creating noisier (er, more complex and bright) harmoni\
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4425 c spectra to create new sounds."),
4426 quiz::Quiz("Wavetable","This term can have two related but slightly different meanin\
4427 gs. A digital oscillator often produces sound by reading a table of numbers in order\
4428 , jumping from the level described by one number to the next. This table of numbers \
4429 describes one cycle of a wave, and therefore is often called a wavetable. Many digit\
4430 al oscillators have multiple wave tables lined up, and can move between these tables\
4431 - either by jumping suddenly (which the original PPG Wave synths did), or by crossf\
4432 ading between them (what most digital wavetable oscillators today do). Some people r\
4433 efer to each table as a "wave" and a set of individual waves as a wavetable."),
4434 quiz::Quiz("Weighting","An equalization curve used in audio tests that compensates f\
4435 or the Fletcher Munson Curve at various levels. (See also "Fletcher-Munson Curves.")\
4436 "),
4437 quiz::Quiz("West Coast Synthesis","The so-called \"West Coast\" approach to synthesi\
4438 s - traditionally associated with companies such as Buchla and Serge - is often base\
4439 d around adding harmonics to simple waveforms, rather than removing (filtering) them\
4440 from complex waveforms. This is often accomplished by using a pair of oscillators (\
4441 sometimes combined into what's called a \"complex oscillator\") where one modulates \
4442 the frequency (FM) or amplitude (AM) of the other; another common West Coast module \
4443 is a waveshaper or a wavefolder. You may also find two-stage envelope generators suc\
4444 h as an AD or AR (often called slope generators) rather than four-stage ADSRs, as we\
4445 ll as more of an emphasis on control voltage manipulation, A common feature is also \
4446 voltage controlled amplifiers that have low-pass filters built into them, creating w\
4447 hat's known as a Low Pass Gate (LPG). The West Coast approach also embraces non-trad\
4448 itional controllers, such as touch plates and the such. Today it's common to mix bot\
4449 h East Coast and West Coast approaches in the same system."),
4450 quiz::Quiz("wet sound","Sometimes people will say a filter has a "wet" sound. This u\
4451 sually refers to a fewer-than-4-pole filter sound - often low or bandpass - with res\
4452 onance turned up a bit, but not to the point of self-oscillation. It's a sound that \
4453 is popular in acid house and other similar techno styles."),
4454 quiz::Quiz("Wet","A sound with effects (such as reverb) mixed is referred to as \"we\
4455 t\"; a sound with no effects is referred to as \"dry.\" Effects units or mixers ofte\
4456 n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
4457 d and the fully-effected sound. Refers to a signal that has the full amount of an ef\
4458 fect (like reverb) applied to it, as opposed to "dry," which refers to the un-effect\
4459 ed sound. Many times, the preferred sound in mixing will be a blend of wet and dry s\
4460 ignals. (See also "Dry.")"),
4461 quiz::Quiz("White Noise","Noise is a random signal that does not have a distinct pit\
4462 ch, such as hissing, breath noise, or the sound of wind or the surf. Noise is often \
4463 described by different "colors" such as white, pink, red, or blue which have differe\
4464 nt frequency distributions. White noise has equal power per unit of frequency (such \
4465 as every 1000 hertz), resulting in a brighter, hissier sound. A noise signal contain\
4466 ing an equal spread of energy across all audible frequencies. Like pink noise, engin\
4467 eers often send a white noise signal through audio equipment for tuning and calibrat\
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4468 ion purposes, or in EQ-ing a live audio space. (See also "Pink Noise.")),
4469 quiz::Quiz("Whole Step","A change in pitch equivalent to two half steps, or the diff\
4470 erence in pitch between two piano keys."),
4471 quiz::Quiz("Wild Sound","In film and video, audio that is recorded separately from t\
4472 he visual that may be added to the audio track later, and does not need to be synchr\
4473 onized with the picture."),
4474 quiz::Quiz("Wind Controller","A device that is played like a wind instrument to cont\
4475 rol a synthesizer, module or virtual instrument via MIDI signals, as opposed to a ke\
4476 yboard controller."),
4477 quiz::Quiz("Windscreen","A covering that fits over a microphone to reduce the excess\
4478 ive noise resulting from wind blowing into the mic. Typically used for recording in \
4479 outdoor locations."),
4480 quiz::Quiz("Wireless Microphone","A microphone that transmits its signal over an FM \
4481 frequency to a receiver offstage, rather than traveling over an audio cable."),
4482 quiz::Quiz("Woofers","A speaker that is designed to reproduce bass frequencies only."),
4483 ),
4484 quiz::Quiz("Write Mode","A mode of operation in an automated console where the engin\
4485 eer is in control of channel gain and the computer is recording the gain changes ove\
4486 r time."),
4487 quiz::Quiz("XLR Cable","A balanced microphone cable utilizing XLR connectors. (See a\
4488 lso "XLR Connector.")),
4489 quiz::Quiz("XLR Connector","A balanced cable connector consisting of 3 or 7 pins, mo\
4490 st commonly used in microphone cables."),
4491 quiz::Quiz("XY Miking","A coincident stereo microphone placement technique in which \
4492 two cardioid microphones are placed with their heads toward each other at a 90-degre\
4493 e angle, and as close together as possible. (See also "Coincident Miking.")),
4494 quiz::Quiz("Y-Cord","A cable with three connectors so that one output may be sent to\
4495 two inputs. Basically, a signal splitter done with spliced wires rather than compo\
4496 nents."),
4497 quiz::Quiz("Zenith","In analog tape recording, refers to the tilt of the tape head i\
4498 n the direction perpendicular to the tape travel."),
4499 quiz::Quiz("Zero-Order Hold (ZOH)","Refers to the mathematical expression of the sig\
4500 nal processing done by a conventional digital-to-analog converter (DAC)."),
4501 };
4502
4503
4504 int main()
4505 {
4506     std::random_device rd;
4507     std::mt19937 gen(rd());
4508     std::uniform_int_distribution<> distria(1, 4);
4509     std::uniform_int_distribution<> distrib(0, game.size()-1);
4510     std::shuffle(std::begin(game), std::end(game), std::default_random_engine());
```

```

4511     std::vector<std::string> answers;
4512     std::string question;
4513     uint32_t n;
4514     uint8_t correct;
4515     uint32_t score=0;
4516     uint32_t tq=0;
4517
4518     for (uint32_t ctr=0;ctr<game.size();++ctr) {
4519         answers.clear();
4520         correct=distria(gen);
4521         for (uint8_t i=1;i<=4;++i) {
4522             if (i == correct) {
4523                 answers.push_back(game[ctr].getA());
4524                 question=game[ctr].getQ();
4525             } else {
4526                 answers.push_back(game[distrib(gen)].getA());
4527             }
4528         }
4529         std::cout << "\33c\e[3J";
4530         if (tq != 0) {
4531             std::cout << "[QUESTIONS: " << tq << " / " << game.size() << " SCORE: " <<
4532 << "]\n";
4533         }
4534         std::cout << "Question #" << tq+1 << ": " << question << "\n\n";
4535         std::cout << "Answer #1.\n" << answers[0] << "\n\n";
4536         std::cout << "Answer #2.\n" << answers[1] << "\n\n";
4537         std::cout << "Answer #3.\n" << answers[2] << "\n\n";
4538         std::cout << "Answer #4.\n" << answers[3] << "\n\n";
4539         std::cout << "What answer is correct (q=quit)? ";
4540         std::cin >> n;
4541         if (n == 0) {
4542             break;
4543         } else if (n == correct) {
4544             score++;
4545         }
4546         tq++;
4547         std::cout << n << " is the answer you gave. And the correct answer is: " << correc\
4548 t << '\n';
4549     }
4550
4551     std::cout << "\33c\e[3J";
4552     if (tq != 0) {
4553         std::cout << "[QUESTIONS: " << tq << " / " << game.size() << " SCORE: " << score\

```

```
4554     << "]\n";
4555         std::cout << "[" << ((double(score)/double(tqs))*100.0) << "% correct answers.]\n";
4556     ";
4557     }
4558     return 0;
4559 }
```

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# Appendix B: Audio Engineering Terminology (0 - 9 - A - Z)

0–5v

*Denotes a range of 0 to 5 volts, which is common for gates, triggers, and modulation control voltages in modular synthesizers. Gates and triggers – which initiate events such as new notes – typically rise from 0v to 5v (0 to 10v is also common), with roughly the middle of that onset starting the event. Gates are considered high when held at 5v (or 10v), and then low when they return to 0v.*

1 pole

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 6 decibels weaker for each octave further away you get from the cutoff frequency. A 6dB/octave filter is often referred to as a “one pole” filter (as each pole of a filter’s design results in 6dB of attenuation), and has a relatively weak effect on the signal going through it. Low Pass Gates (LPGs) typically – but not always – use 1 pole low pass filters, reducing the strength of higher harmonics by 6 decibels for every octave above its cutoff frequency.*

1 ppqn

*The most common sequencer clock division forwards it one step (pulse) per quarter note. This is often the core sync pulse that is distributed in a modular system, and is either multiplied or divided to create other musical divisions.*

1 v/oct

*The most common standard for controlling pitch in a modular synthesizer. Under the system, increasing the voltage going into a VCO (Voltage Controlled Oscillator) 1 volt – say, from 0.5v to 1.5v – would raise its pitch by one octave.*

1.2 v/oct

*Buchla compatible synths have standardized on the 1.2 volt per octave system, instead of the more common 1 v/oct. With 12 semitones to an octave in Western music, an equally tempered scale would work out to precisely 0.1 volts for a change in pitch of 1 semitone.*

1/4”

*The most common connector size used for 5U (Moog format) modular synthesizers. These are TS (tip/sleeve) jacks and plugs, similar to guitar and other instrument cables.*

1/8”

*Often used to incorrectly describe the connector size commonly used in Eurorack format modules, as well as Buchla audio signals. In fact, Eurorack modules use 3.5mm jacks and plugs (slightly larger than 1/8”); Buchla uses Switchcraft Tini-Jax connectors. Tini-Jax are 3.5mm in diameter, but are*

*slightly different physically from a common 3.5 mm jack. 1/8" plugs would be loose in both of these jacks, so make sure you get 3.5mm connectors ordering parts or cables for these formats.*

10 vpp

*An abbreviation for "10 volts peak to peak" with peak to peak being the difference between the lowest and highest voltage reached during a signal's travels. This is a common voltage range for both audio and modulation signals in a modular synthesizer. The actual range is between -5 and +5 volts. The precise range may be varied to change the depth of their effect, so don't get too hung up on specific voltage ranges. Pay more attention to whether they vary between 0v and some value, or swing in roughly equal amounts both above and below 0v (as 10vpp does).*

12 dB/oct

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 12 decibels weaker for each octave further away you get from the cutoff frequency. A 12dB/octave filter is often referred to as a "two pole" filter (as each pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and Oberheim instruments often featured 2-pole filters, often resulting in brighter sounds when compared to those with 4-pole instruments.*

16'

*Sometimes seen on octave selector switches on oscillators. It refers to the length of an organ pipe. Longer pipes = lower pitches; 16' is in the mid-bass range. A pipe or setting half as long (8') is one octave higher; a pipe half as long again (4') is two octaves higher; etc.*

18 dB/oct

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 18 decibels weaker for each octave further away you get from the cutoff frequency. It is often used a coded shorthand for when someone wants to refer to acid-type bass lines from a TB-303 without mentioning the instrument by name.*

2 Pole

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 12 decibels weaker for each octave further away you get from the cutoff frequency. A 12dB/octave filter is often referred to as a "two pole" filter (as each pole of a filter's design results in 6dB of attenuation). Vintage Arp, Korg, and Oberheim instruments often featured 2-pole filters, often resulting in brighter sounds when compared to those with 4-pole instruments.*

2.5 mm

*A common screw thread size used to mount Eurorack modules. This size is most common when using a system of loose nuts that slide along the rails that the modules are attached to.*

24 dB/oct



*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 24 decibels weaker for each octave further away you get from the cutoff frequency. This design is often used in vintage Moog and Roland synths. 4-pole filters are often associated with subjectively fatter, more “round” sounds than 2-pole filters – but generalizations are always dangerous.*

24 ppqn

*A common master clock division used in MIDI, DIN sync, and other systems common to electronic music and synthesizers. It means internally, 24 subdivisions of time are counted for every quarter note at the current tempo. This fast internal clock can then be divided down to create sixteenth notes ( $\div 6$ ), eighth notes ( $\div 12$ ), eighth note triplets ( $\div 8$ ), etc.*

2'

*Sometimes seen on octave selector switches for oscillators. It refers to the length of an organ pipe. Shorter pipes = higher pitches; 2' is rarely seen on modular oscillators as it's rather high in pitch – two octaves above middle C as a starting point. A pipe or setting twice as long (4') is one octave lower; a pipe twice as long again (8') is two octaves lower; etc.*

3 mm

*A common screw thread size used to mount Eurorack modules. This size is most common when using module mounting rails that have been pre-drilled.*

3 Pole

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 18 decibels weaker for each octave further away you get from the cutoff frequency. It is often used a coded shorthand for when someone wants to refer to acid-type bass lines from a TB-303 without mentioning the instrument by name.*

3.5 mm

*The standard connector size used for jacks and cables in Eurorack format modular synthesizers. Note that this is slightly larger than 1/8”.*

303

*The TB-303 Bass Line by Roland became a cult favorite in Acid House and other flavors of EDM (Electronic Dance Music) for its rubbery, slithery synth bass sound. Many attribute the sound of the 303 to its filter design.*

32'

*Sometimes seen on octave selector switches on oscillators. It refers to the length of an organ pipe. Longer pipes = lower pitches; 32' is the lowest setting you will see and is getting into earthquake territory. A pipe or setting half as long (16') is one octave higher; a pipe half as long again (8') is two octaves higher; etc.*

### 3U

*Refers to modules that are 3 rack units (U) high – the Eurorack standard, which is by far the most common modular format today, even though it's one of the youngest formats.*

### 4 Pole

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 24 decibels weaker for each octave further away you get from the cutoff frequency. This design is often used in vintage Moog and Roland synths. 4-pole filters are often associated with subjectively fatter, more “round” sounds than 2-pole filters – but generalizations are always dangerous.*

### 4-40

*A screw thread size occasionally used to mount Eurorack modules. This size is used by Pittsburgh Modular for their cases, for example.*

### 4U

*Refers to modules that are 4U (rack units) high – namely, Buchla and Serge systems, as well as do-it-yourself clones of these modules. Both Buchla and Serge lean toward a more experimental approach to synthesis and music, so some users wear “4U” as a badge of honor that they're non-conformist and cool. (And they are.)*

### 4'

*Sometimes seen on octave selector switches on oscillators. It refers to the length of an organ pipe. Shorter pipes = higher pitches; 4' is the highest octave setting you will see on most oscillators. A pipe or setting twice as long (8') is one octave lower; a pipe twice as long again (16') is two octaves lower; etc.*

### 5U

*Refers to modules that are 5U (rack units) or 8.75” (22.2 cm) high, which is most often associated with the vintage Moog standard and those who have followed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You will sometimes hear this used interchangeably with MU for Moog Units, which also refers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standard is both historical and physically large, some users “5U” as a badge of honor that they're traditional and cool. (And they are.) There was also a briefly popular 5U format from MOTM that used a different width and power connection. It has since been discontinued, but there are still diehard MOTM format users today.*

### 6 dB/oct

*This format of numbers and abbreviations (dB/oct = decibels per octave) is often used to refer to the frequency response behavior of a filter. A filter typically has a cutoff or corner frequency it is tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its loudness is multiples of 6 decibels weaker for each octave further away you get from the cutoff frequency. A 6dB/octave filter is often referred to as a “one pole” filter (as each pole of a filter's design results in 6dB of attenuation), and has a relatively weak effect on the signal going through it. Low Pass*

*Gates (LPGs) typically – but not always – use 1 pole low pass filters, reducing the strength of higher harmonics by 6 decibels for every octave above its cutoff frequency.*

808

*The TR-808 Rhythm Composer by Roland created all of its sounds using analog circuitry. When it first came out, it was not well loved, as the analog sounds weren't realistic enough. But later, music styles such as House and Hip-Hop adopted its big, booming synthetic sounds. When a module says it recreates "808" drums, this is the instrument they are trying to emulate. Most copied is the 808 kick drum sound, which tends to be a low-pitched, long-decaying sine-like wave often with a snappy attack.*

8'

*Sometimes seen on octave selector switches on oscillators. It refers to the length of an organ pipe. Shorter pipes = higher pitches; 8' is typically associated with middle C. A pipe or setting half as long (4') is one octave higher; a pipe or setting twice as long (16') is one octave lower.*

909

*The TR-909 Rhythm Composer was the follow-up to Roland's now-revered TR-808. It combined digital samples for the hi-hat and cymbal along with the 808's analog sounds, and has also become popular. When a module says it produces 909-like sounds, this is the instrument it is referencing.*

---

A-440

*This is the frequency in hertz (cycles per second) of the A above Middle C. It is often used as a tuning reference.*

A/B Technique

*A stereo microphone placement technique in which two cardioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart from each other (depending on the size of the sound source) to create a left/right stereo image. Also known as Spaced Pair.*

A/D

*Abbreviation of Analog-to-Digital Conversion, the conversion of a quantity that has continuous changes (like electrical signals) into numbers that approximate those changes (i.e., computer data).*

Absolute Phase

*This term describes a perfect polarity between an original signal (into the microphone) and the reproduced signal (through the speaker). When positive pressure exerted upon the microphone is translated as positive pressure to the loudspeaker, the two are in "absolute phase."*

Absorption

*In acoustics, absorption is what happens when sound waves are absorbed by a surface, as opposed to bouncing off the surface (reflection). Absorptive materials in a control room, for example, tend to "deaden" the sound of the room because the sound energy is absorbed rather than reflected. (See also "Reflection.")*

### AC Coupled

*An AC coupled input attempts to remove any constant DC voltage going through it. This is useful if you have an audio signal (such as the output of an oscillator) which is AC in nature, and you want to remove any accidental DC offset that might have crept into it. These offsets can cause one half of the AC waveform to clip prematurely, or can cause clicks at the start and end of envelopes or mutes. However, this coupling can mildly distort a wave going through it, as in essence AC coupling is a high pass filter that is attempting to remove very low frequency components.*

### AC

*Alternating Current - The type of electrical current found in standard electrical outlets and studio signals running through audio lines. In AC, the current “alternates” directions, flowing back and forth through the circuit. In modular terms, AC refers to a voltage that alternates between positive and negative values – such as the output of an oscillator.*

### Accelerometer

*A device that measures the acceleration to which it is subjected and creates an electric signal to match it. In music and audio, accelerometers are found in such things as microphones and guitar pickups.*

### Acorn Tube

*Named for its acorn-like shape, an acorn tube is a small vacuum tube used in ultra high frequency (UHF) electronics such as tube amplifiers.*

### Acoustic Amplifier

*The part of a musical instrument that vibrates in response to the initial vibration of the instrument, causing the surrounding air to move more efficiently and making the sound louder. For example: the body of an acoustic guitar, the bell of a horn, a drum’s shell, and the wooden soundboard of a piano.*

### Acoustic Echo Chamber

*A room designed with hard, non-parallel surfaces to create reverberation. In recording studios, they are used to add natural reverb to a dry signal.*

### Acoustics

*The science of the sound—more specifically, the science of the properties and behavior of sound waves. A good understanding of acoustics is essential to audio engineering and studio design.*

### Active Device

*A component that is designed with the ability to control electrical current (as opposed to a “Passive Device”). In the recording studio, active devices are generally components that include an amplifier. (See also “Passive Device.”)*

### Active Multiple

*Quite often you need to split or copy a signal to send to more than one destination. This is commonly done with a multiple, where you plug one source in, and then plug in additional patch cables to go off to multiple destinations. An active or buffered multiple is one that includes a buffer circuit between the input and output, making sure the signal does not lose its strength or integrity by being split too many times, and that no funny business happening on one of the outputs affects any of the other connections. Some modules have good buffering built into their outputs, and can drive multiple modules without issue. But if you try to use a passive mult to connect to, say, three oscillators, and*

*you realize the tracking isn't very good (they quickly go out of tune as you go up and down the scale), then you need a buffered mult instead.*

#### Actuator

*The part of a switch that causes change of the contact connections (e.g., toggle, pushbutton, or rocker).*

#### AD

*Shorthand for a two-stage Attack/Decay envelope. This simple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 volts) at a speed defined by its Attack parameter, and then immediately falls back to 0 volts at a rate defined by its Decay parameter. A variation on this is the AHD envelope: After finishing the Attack stage, it holds at the maximum level for a specified amount of time (in contrast to an AR envelope, which holds at the maximum level for as long as the note on gate is high), and then decays back to zero. I have heard there are some envelopes that a hybrid of AHD and AR in that they hold the maximum level for either the defined Hold time or the as long as the incoming gate is high.*

#### Additive Synthesis

*One of the main properties that make a sound unique is the mixture of harmonics – pure component frequencies – that it is built from. Additive synthesis is a technique that gives you direct control over each of those component harmonics, allowing you to directly dial in the mix you want. As immediate and intuitive as that sounds on paper (or on screen), in reality it takes a lot of work to craft the correct mixture to recreate another sound, especially since the strength of each harmonic usually varies over time. Additive synthesis oscillators are relatively rare in modular synths; two examples are the Verbo Harmonic Oscillator and the Make Noise tELHARMONIC.*

#### ADSR

*An envelope generator with four stages: Attack, Decay, Sustain, and Release. When this envelope generator receives a gate input, it typically starts at 0 volts (which is the equivalent of silence when connected to a Voltage Controlled Amplifier, or the lowest frequency when connected to a voltage controlled filter or oscillator) and raises to the maximum voltage it can output (typically 5 to 10 volts depending on system; it can often be set with an output level control) over a time set by the Attack control. Once it reaches that level, the output voltage immediately starts dropping to speed set by the Decay control until it reaches the voltage set by the Sustain control. If the input gate is still active, this level is maintained until the gate goes back to 0 volts (usually because you released the key on a controlling keyboard, etc.). At that time, the output voltage then starts dropping back to 0 volts at the rate set by the Release control.*

#### AES

*Audio Engineering Society.*

#### AES3

*(sometimes called AES/EBU) A digital audio transfer standard developed by the Audio Engineering Society and the European Broadcasting Union for carrying dual-channel digital audio data between devices. AES3 is the protocol behind XLR cables, as well as RCA and S/PDIF cables.*

#### AFG

*The AFG (Audio Frequency Generator) is a very full-featured analog oscillator released by Livewire*

*Electronics. It has since been discontinued, but refurbished B-stock units come up for sale every now and then. The expansion modules were, to the best of my knowledge, never released (at least not widely).*

#### Aftertouch

*(Also called “Pressure Sensitivity”) some keyboards measure how hard you press down on the keys, and convert this to a voltage (or other control signal such as MIDI, which can then be converted into a control voltage) that you can use to add expression to a note, such as adding vibrato or opening the filter wider. Monophonic aftertouch measures one pressure value for the entire keyboard, regardless of which key(s) you are pressing; polyphonic aftertouch produces a signal for each individual key. Important trivia: Touch plate keyboards actually measure the surface area of the skin touching them rather than pressure or force – so you can increase or decrease the aftertouch amount by rolling between the tip and length of your finger.*

#### AHDSR

*Attack, Hold, Decay, Sustain, and Release. This is a slightly fancier ADSR envelope that holds the voltage typically at its maximum value for a specified time after the attack is done rising and before the decay starts falling.*

#### Aliasing

*A type of digital signal distortion that occurs in a sampler when the incoming signal frequency exceeds the Nyquist frequency for that unit. The sampler reproduces it at an incorrect frequency, or an “alias,” causing a distortion or artifact in the sound. If you play back a digital audio file where half of the sample rate is an audible pitch, you will also hear a mirror image of the sound’s harmonic content reproduced started at that half-sample-rate pivot (unless some excellent filtering has taken place). (See also “Nyquist Frequency.”)*

#### Alternating Current (or AC)

*The type of electrical current found in standard electrical outlets and studio signals running through audio lines. In AC, the current “alternates” directions, flowing back and forth through the circuit.*

#### AM

*Amplitude Modulation (AM) is the name given to the technique of varying the amplitude or loudness of one signal known as the carrier (typically an audio signal, swinging both above and below 0 volts) with a second signal called the modulator. In the typical amplitude modulation (AM) scenario, a low frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into the control input of a voltage controlled amplifier to add vibrato to an audio signal passing through it. Technically, this is known as a two-quadrant multiplier or modulator, as any negative swings in the modulation signal are ignored; when patching tremolo, you may need to make sure an offset voltage is being added to your LFO to make sure the sound doesn’t cut out on the lower excursions of the LFO’s waveform.*

#### Ambience

*In most cases, this refers to the “atmosphere” of a certain place, like a restaurant. But in recording, it refers to the part of the sound that comes from the surrounding environment rather than directly from the sound source. For example, the sound waves coming into your ears from a cello being played*

*are coming directly from the source, but the sound of the same cello coming to you after bouncing off the back wall is ambient sound.*

#### Ambient Field

*The area away from the sound source where the reverberation is louder than the direct sound.*

#### Ambient Miking

*This refers to placing a microphone in the ambient field of a room to record the ambient reverberations of the sound. The recording engineer often does this in addition to direct micing of the instrument(s) to create a blend or mix of direct and reverberant sound in the recording.*

#### Amp

*An abbreviation for “Amplifier,” “Amplitude” or “Ampere,” depending on context.*

#### Ampere

*The unit of measure for electrical current, abbreviated Amp.*

#### Amplifier

*A device that increases the level or amplitude of an electrical signal, making the resulting sound louder.*

#### Amplitude Modulation

*Amplitude Modulation (AM) is the name given to the technique of varying the amplitude or loudness of one signal known as the carrier (typically an audio signal, swinging both above and below 0 volts) with a second signal called the modulator. In the typical amplitude modulation (AM) scenario, a low frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into the control input of a voltage controlled amplifier to add vibrato to an audio signal passing through it. Technically, this is known as a two-quadrant multiplier or modulator, as any negative swings in the modulation signal are ignored; when patching tremolo, you may need to make sure an offset voltage is being added to your LFO to make sure the sound doesn't cut out on the lower excursions of the LFO's waveform.*

#### Amplitude

*The height of a waveform above or below the zero line. In audio, this usually translates to the signal strength or the volume of the sound.*

#### Analog Recording

*A recording of the continuous changes of an audio waveform. The most common example of analog recording in a recording studio is recording on reel-to-reel magnetic tape.*

#### Analog Shift Register

*An Analog Shift Register (ASR) is a cross between a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how those work). When initially triggered, it samples the incoming voltage, and presents that at its first output. On the second trigger, the incoming voltage is sampled again with this new voltage presented at the first output, while the original voltage is now moved to a second output. This game of “telephone” is passed along for as many stages as the ASR has – traditionally three or four.*

### Analog To Digital Converter (A/D; or ADC)

*A device that translates a continuously changing signal (analog) into numeric values that approximate those changes (digital). In audio recording, this refers to converting recorded sound from electrical voltages to computerized data.*

### Analog

*The term analog implies a signal is continuously variable, compared to digital where a signal has been converted into discrete numbers. In the land of modular synthesizers, analog refers to a circuit design that has no digital (or at least, computer-based) components – instead, it does all of its processing using transistors, diodes, capacitors, and the such rather than CPUs and DSPs.*

### AND function

*One of the most common Boolean or binary logic functions, AND says only output a gate on signal if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all have gate ons). A NAND function has an inverted output: The output would be low if both inputs were high, but otherwise would be high.*

### AR

*The two-stage Attack/Release envelope raises from 0 volts to its maximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack parameter, and then stays at that value for as long as the gate signal fed into the envelope generator stays high. Then when the gate signal goes back to zero, the envelope’s output also falls back to zero at a rate set by its Release parameter. (There is a separate type of envelope known as an AHD – Attack/Hold/Decay – where you specify a fixed time for the level to stay at its maximum, rather than pay attention to the gate signal.)*

### Arpeggiator

*Putting on our music theory hat for a second, an arpeggio is a type of “broken chord” where the notes are played individually rather than all at once. An arpeggiator – usually built into a keyboard, or a device inserted between your keyboard and sound module – makes it easier for you to play arpeggios: You just hold down the notes of the chord, and it automatically plays the notes one at a time, over and over again, like a step sequencer you can program on the fly just by holding down a chord. Good arpeggiators have options for different patterns (up, down, back and forth, random, etc.), and even a latch or hold where it will keep doing this even after you’ve released the keys.*

### ASR

*An Analog Shift Register (ASR) is a cross between a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how those work). When initially triggered, it samples the incoming voltage, and presents that at its first output. On the second trigger, the incoming voltage is sampled again with this new voltage presented at the first output, while the original voltage is now moved to a second output. This game of “telephone” is passed along for as many stages as the ASR has – traditionally three or four.*

### Attack/Decay/Sustain/Release

*An envelope generator with four stages: Attack, Decay, Sustain, and Release. When this envelope generator receives a gate input, it typically starts at 0 volts (which is the equivalent of silence when connected to a Voltage Controlled Amplifier, or the lowest frequency when connected to a voltage*



*controlled filter or oscillator) and raises to the maximum voltage it can output (typically 5 to 10 volts depending on system; it can often be set with an output level control) over a time set by the Attack control. Once it reaches that level, the output voltage immediately starts dropping to speed set by the Decay control until it reaches the voltage set by the Sustain control. If the input gate is still active, this level is maintained until the gate goes back to 0 volts (usually because you released the key on a controlling keyboard, etc.). At that time, the output voltage then starts dropping back to 0 volts at the rate set by the Release control.*

#### Attack/Decay

*Shorthand for a two-stage Attack/Decay envelope. This simple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 volts) at a speed defined by its Attack parameter, and then immediately falls back to 0 volts at a rate defined by its Decay parameter. A variation on this is the AHD envelope: After finishing the Attack stage, it holds at the maximum level for a specified amount of time (in contrast to an AR envelope, which holds at the maximum level for as long as the note on gate is high), and then decays back to zero. I have heard there are some envelopes that a hybrid of AHD and AR in that they hold the maximum level for either the defined Hold time or the as long as the incoming gate is high.*

#### Attack/Hold/Decay/Sustain/Release

*This is a slightly fancier ADSR envelope that holds the voltage typically at its maximum value for a specified time after the attack is done rising and before the decay starts falling.*

#### Attack/Release

*The two-stage Attack/Release envelope raises from 0 volts to its maximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack parameter, and then stays at that value for as long as the gate signal fed into the envelope generator stays high. Then when the gate signal goes back to zero, the envelope's output also falls back to zero at a rate set by its Release parameter. (There is a separate type of envelope known as an AHD – Attack/Hold/Decay – where you specify a fixed time for the level to stay at its maximum, rather than pay attention to the gate signal.)*

#### Attack

*This usually refers to the first stage of an envelope that occurs at the onset of a note, as it rises from 0 volts (silence when if controlling an amplifier module) to typically the value of maximum loudness. Percussive and plucked sounds have very fast attacks; slow, languid wind or string instrument phrases may have long attacks.*

#### Attenuation

*The reduction of electrical or acoustic signal strength. In audio, attenuation is measured in decibels (dB) and is typically heard as a reduction in volume. Sound waves traveling through the air naturally attenuate as they travel away from the source of the sound. Engineers also purposefully attenuate signals in the studio through gain controls or pads to prevent overload.*

#### Attenuator

*A control that can reduce the strength of a signal or voltage going through it.*

#### Attenuverter

*A special version of an attenuator that can also invert the polarity of the signal or voltage going*

*through it. Most attenuverters use pass through no signal at their center position; as you turn them clockwise, you turn up the normal version of the signal; as you turn them counterclockwise, they turn up an inverted version of the signal. Some attenuverters are a normal attenuator with a polarity switch added on.*

#### Audio Frequency Generator

*The AFG (Audio Frequency Generator) is a very full-featured analog oscillator released by Livewire Electronics. It has since been discontinued, but refurbished B-stock units come up for sale every now and then. The expansion modules were, to the best of my knowledge, never released (at least not widely).*

#### Audio

*In its broadest sense, audio is the range of frequencies we humans can hear with our ears. In the technical sense, audio refers to the transmission, recording or reproduction of sound, whether digitally, electrically or acoustically.*

#### Automatic Dialogue Replacement (ADR)

*The process of re-recording dialogue for film in a controlled environment after the film is shot, for the purpose of replacing poorly recorded dialogue.*

#### Automatic Gain Control

*A compressor with a long release time, which is used to keep the volume of the audio at a consistent level.*

#### Automation

*Programming certain changes to occur automatically during recording and/or playback. In the studio, engineers use automation on their consoles or computers so various parameters will change automatically at different times during multitrack recording and playback. This pre-programming feature makes it easier to create those changes than attempting to perform them all manually in real time.*

#### Auxiliary Equipment

*External signal processing devices that work alongside the mixing console to modify the signal.*

#### Auxiliary Return

*(Abbreviated Aux Return or Return) The input on a console or DAW that returns the effected signal sent through the auxiliary send back into the channel mix.*

#### Auxiliary Send

*(Abbreviated Aux Send or Send) A control to adjust the signal level being sent from the input channel on a console or DAW to auxiliary equipment or plug-ins through the auxiliary bus. This is typically used for creating an effects loop that processes a portion of the signal, then returns it into the mix through the auxiliary return.*

#### Axis

*An imaginary line around which a device operates. For example: in microphone use, the axis is an imaginary line coming out from the front of the microphone in the direction of motion of the*

*diaphragm, delineating the optimum location for the mic to pick up the sound. Sounds that occur “off-axis” from the microphone will not be picked up as clearly.*

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### Background Noise

*Refers to either*

- \*1) The ambient noise in a room unrelated to the instrument(s) or vocal(s) being recorded; or \**
- 2) The system noise unrelated to the recorded signal. (All electronics emit a level of noise.)*

### Baffles

*Sound absorbing panels that are used to prevent sound waves from entering or leaving a space.*

### Balance

*1) The relative level of two or more instruments in a mix, or the relative level of audio signals in the channels of a stereo recording. 2) To even out the relative levels of audio signals in the channels of stereo recording.*

### Balanced Audio

*This refers to a system where three wires are used to carry an audio signal: one is the ground (the 0 volt reference), the second carries the audio signal as it varies above and below 0v, and the third carries an inverted copy of the audio signal that goes negative while the original is going positive. Balanced audio usually implies a reference signal level of +4dB (higher than line level; still lower than most modular synths), although microphone signals – much weaker by comparison, and therefore more susceptible to outside noise – are almost always balanced as well. Modular synths tend to use unbalanced audio for their internal signals. If you require a balanced output (or input), you need a special module that converts between balanced and unbalanced audio, plus does any necessary level matching.*

### Balanced Cable

*A cable consisting of three wires (two signal wires and a ground wire) and two connectors. The two signal wires carry the same signal in opposite polarities, providing protection against interference and noise in a balanced system. Examples of balanced cables include tip-ring-sleeve (TRS) stereo cables and XLR cables.*

### Balanced Mixer

*A circuit or device that generates the sum and difference frequencies of two input signals.*

### Balanced Modulator

*Balanced or ring modulation is a special type of amplitude modulation, where one bipolar (swinging both above and below 0 volts) signal – the modulator – is used to vary the amplitude of a second bipolar signal, known as the carrier. The modulator’s frequency is both added to and subtracted from the carrier’s frequency; the resulting harmonics replace the original carrier and modulator.*

### Banana

*An alternate type of connector ([https://en.wikipedia.org/wiki/Banana\\_connector](https://en.wikipedia.org/wiki/Banana_connector)) used by 4U systems*

*such as Buchla (control voltages) and Serge (both control and audio). These cables have only one wire, so they carry only the signal, relying on the module panels and chassis of the system to provide the ground reference. Banana connectors have an advantage in that they are usually “stackable” meaning you can plug a one jack into the back of another, providing a passive multiple.*

#### Band Pass Filter

*A device, circuit or plug-in that allows a narrow band of frequencies to pass through the circuit, rejecting or attenuating frequencies that are either higher or lower than the specified range.*

#### Band Stop Filter

*A device, circuit or plug-in that attenuates a narrow band of frequencies in the signal, allowing frequencies outside the band to pass. The exact opposite of a band pass filter.*

#### Band Track

*(Sometimes abbreviated “Track”) A mixdown of a song minus the lead vocal and/or background vocals. In other words, a mixed track containing only the instrumental parts of the song.*

#### Band

- 1) A range of frequencies, often identified by the center frequency of the range.*
- 2) A group of musicians playing together.*

#### Bandpass Filter

*A bandpass filter (BPF) leaves the harmonics around the center, corner or cutoff frequency untouched, and attenuates those above and below the center frequency. The further away you get from the center, the more they are attenuated, based on the number of poles in the filter, with each pole equalling 6 decibels of attenuation for each octave you get away from that center.*

#### Bandwidth

*In signal processing, bandwidth refers to the usable frequency range of a communication channel, measured by the difference between the device’s highest and lowest usable frequencies.*

#### Bank

- 1) A collection of sound patches, sequencer data and/or operating parameters of a synthesizer’s generators and modifiers in memory.*
- 2) A group of sound modules as a unit.*

#### Bar

*In music notation, bar is another term for measure a specified period of time containing a certain number of beats, and marked by bar lines on each side of the written measure.*

#### Bark Scale

*The human auditory (hearing) system can be thought of as consisting of a series of bandpass filters. Interestingly, the spacing of these filters do not strictly follow either a linear frequency scale or a logarithmic musical scale. The Bark Scale is an attempt to determine what the center frequency and bandwidth of those “hearing filters” are (known as critical bands).*

#### Barrier Miking

*A microphone placement technique in which a microphone is placed close to a reflective surface. When*

*done correctly, barrier miking ensures that both the direct and reflected sounds reach the microphone simultaneously, preventing phase cancellation between the two.*

#### Basic Session

*The first audio recording session for recording the basic tracks that serve as the song's foundation (for example, the drums and bass).*

#### Bass Reflex

*A type of loudspeaker cabinet design in which a port (opening) in the speaker cabinet enhances bass frequencies. The principle is that the sound pressure generated by the back of the speaker cone inside the cabinet is routed out the port at the front of the cabinet, mixed with the sound coming from the front of the woofer. Changing the port size and position will greatly change the character of the low frequencies.*

#### Bass

*The lower range of audio frequencies up to approximately 250 Hz. A reference value.*

#### BBD

*An early design for an echo or delay effect where the input audio would be sampled as an analog voltage, and held for a brief moment. Then at the next above-audio sample rate clock pulse, this voltage would get passed to the next sample and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade delays (BBDs) usually have numbers of stages or buckets that are powers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the number of stages multiplied by the time interval between samples.*

#### Beaming

*A phenomenon found in loudspeakers in which higher frequencies are projected straight out of the loudspeaker, rather than dispersing along with the lower frequencies. When you stand on-axis in front of the speaker, it sounds as though it is only reproducing the high frequencies, rather than the mids or lows. This phenomenon is alleviated by routing the high frequencies through horns in the loudspeaker.*

#### Beat Mapping

*The process of adjusting the tempo variations in a recorded piece of music to fit the set tempo of the project. In a DAW, this is done using time stretching tools and cuts to synchronize the transients to the appropriate tempo markers. This technique is often used, for example, to reconcile a drum or bass performance that was recorded without a click track.*

#### Beat

- 1) The steady, even pulse in music.*
- 2) The action of two sounds or audio signals of slightly different frequency interfering with one another and causing periodic increases and decreases in volume, heard to the ear as "beats."*

#### Beating

*When two oscillators are tuned to very nearly – but not quite – the same frequency, the difference between them causes an interference pattern known as beating. When the difference in frequency is below the audio rate, this can sound like a tremolo applied to the loudness of the combined sound.*

### Beatmatching

*A technique predominantly used by DJs to synchronize the tempos of two recorded tracks, generally through the use of time stretching and pitch shifting tools, to create a seamless transition from one song into another.*

### Beats Per Minute (B.P.M.)

*BPM (beats per minute) is the most common way of stating tempo: How many beats (typically, quarter notes) should be counted every minute. A tempo of 120 beats per minute means there would be two beats every second (120 beats/minute x 1 minute/60 seconds = 2).*

*The number of steady even pulses in music occurring in one minute, defining the tempo of the song.*

### Berlin School

*A particular style of electronic music popularized by the likes of Tangerine Dream and Klaus Schulze based on analog synthesizers, heavy on repetitive sequences and floating chords or drones with solos played on top. More recent versions of Berlin School music can be heard from Node and Red Shift.*

### Bi-amplification

*A technique in which high and low frequencies in a speaker or speaker system are driven by two separate amplifiers.*

### Bi-Directional Pattern

*A microphone pickup pattern which is most sensitive to picking up sounds directly in front and back of the mic, effectively rejecting sounds coming from the sides. Also called a "figure-8 pattern."*

### Binary

*A cornerstone of digital systems is the binary counting method, where each digit can have only two different values: 0 or 1; off or on; low or high. A binary signal can only have one of these two states. Therefore, a gate or trigger signal in a modular synth – even if generated by analog circuitry – could be referred to as a binary type signal. See the entry for Boolean for things you can do with binary signals like gates and divided clocks.*

### Bipolar

*A voltage that can range both above and below zero is referred to as bipolar. Some modulation signals inside a modular synth – such as vibrato (varying the pitch of an oscillator both above and below the note it is supposed to be playing) – are bipolar in nature.*

### Bit

*The smallest unit of digital information representing a single "0" or "1."*

### Bitrate (or Bit Depth)

*In digital recording, the number of computer bits used to describe each sample. The greater the bitrate, the greater the dynamic range of the sampled sound. The quality and resolution of an audio sample are described as a combination of sample rate and bitrate. (See also "Sample Rate.")*

### Blending

*The mixing of multiple sounds or channels together to form one sound, or mixing the left and right signals together.*

**Blue Noise**

*Technically, a type of noise whose power density (spectral loudness) increases 3 dB per octave with increasing frequency. It has a very “hissy” characteristic, lacking in bass.*

**Boolean**

*Boolean logic only can have two states: high or low; 1 or 0; on or off.*

**Boom Stand**

*A microphone stand equipped with a telescoping support arm to hold the microphone.*

**Boom**

*A telescoping support arm attached to a microphone stand holding the microphone.*

**Boost**

*To increase gain at specific frequencies with an equalizer.*

**Bouncing**

*(also called “Ping-Ponging” or “Ponging”) The technique of combining and mixing multiple tracks onto one or two tracks (mono or stereo). This can be done in real-time or analog by playing the tracks through the console and recording them onto separate tracks, or digitally through a digital audio workstation. Bouncing was once used frequently by engineers to free up additional tracks for recording, but in digital workstations where tracks are virtually unlimited, this practice is basically obsolete. Today, engineers typically bounce tracks for the purpose of creating a preliminary or final mix of a song.*

**Boundary Microphone**

*An omnidirectional microphone designed to be placed flush against a flat surface (or boundary), effectively creating a “half-Omni” pickup pattern while eliminating the danger of phase issues from reflected sounds. A popular type of boundary microphone is Crown Audio’s trademark Pressure Zone Microphone (PZM).*

**BPF**

*A bandpass filter (BPF) leaves the harmonics around the center, corner or cutoff frequency untouched, and attenuates those above and below the center frequency. The further away you get from the center, the more they are attenuated, based on the number of poles in the filter, with each pole equalling 6 decibels of attenuation for each octave you get away from that center.*

**BPM**

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**Breathing**

*Pumping and Breathing – In studio jargon, an effect created when a compressor is rapidly compressing and releasing the sound, creating audible changes in the signal level. “Pumping” generally refers to the audible increase of sound levels after compression has taken place; “breathing” refers to a similar effect with vocals, raising the signal volume just as the vocalist is inhaling. Pumping and*

*breathing is a sign of cheap compression or over-compression, and is usually undesirable, although some engineers and musicians use it on purpose occasionally to create a particular effect.*

#### Brickwall Filter

*A certain type of low-pass filter exhibiting a steep cutoff slope which resembles a “brick wall.” While these filters are often found in A/D converters to prevent aliasing, their steep cutoff can introduce unwanted side-effects to the audio signal, such as phase shift.*

#### Bridging

*A technique of feeding a single input to both channels of an amplifier, then summing them into one, thereby effectively doubling the amplifier power supplied to the signal.*

#### Brownian Noise

*Also referred to as brown noise, technically it’s a type of noise whose power density (spectral loudness) decreases 6 dB per octave with increasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distance. It can also be used as a slowly changing random control voltage or modulation signal, instead of as an audio source.*

#### Buchla Bongos

*This is a classic patch where a complex sound source – such as one oscillator frequency modulating another – is sent through a Low Pass Gate with either just a trigger to “strike” the vactrol inside or otherwise an instant attack/fast decay envelope to create a nice percussive sound. The fact that the low pass gate reduces the higher harmonics as its volume dies away helps tame the harmonics coming from the complex source, and give it a decay similar to a struck percussive instrument.*

#### Bucket Brigade Delay

*An early design for an echo or delay effect where the input audio would be sampled as an analog voltage, and held for a brief moment. Then at the next above-audio sample rate clock pulse, this voltage would get passed to the next sample and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade delays (BBDs) usually have numbers of stages or buckets that are powers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the number of stages multiplied by the time interval between samples.*

#### Bucking

*A type of phase cancellation in which two identical signals or frequencies, having the same amplitude but opposite polarity, cancel one another out. Most commonly used in the context of musical instrument frequencies. Example: a “Humbucker” guitar pickup is designed to remove or “buck” hum frequencies from the signal using this principle.*

#### Buffered Multiple

*Quite often you need to split or copy a signal to send to more than one destination. This is commonly done with a multiple, where you plug one source in, and then plug in additional patch cables to go off to multiple destinations. An active or buffered multiple is one that includes a buffer circuit between the input and output, making sure the signal does not lose its strength or integrity by being split too many times, and that no funny business happening on one of the outputs affects any of the other connections. Some modules have good buffering built into their outputs, and can drive multiple modules without issue. But if you try to use a passive mult to connect to, say, three oscillators, and*



*you realize the tracking isn't very good (they quickly go out of tune as you go up and down the scale), then you need a buffered mult instead.*

#### Bulk Dump

*Short for System Exclusive Bulk Dump, a method of transmitting data such as the internal parameters between MIDI devices.*

#### Burst Generator

*When you send this module a trigger, it outputs a stream or "burst" of triggers in response. You usually have control over the number of triggers, the spacing between them, and often the probability that individual trigger output will be sent or skipped (for random patterns). At its most tame, it can be used to create "double pluck" triggers in response to a normal note on; and its most extreme, it is used to trigger a high-energy, chaotic stream of drum hits that may or may not be in time with the music.*

#### Bus Board

*This simple circuit board takes the output of your modular system's power supply and creates multiple copies of it, routed to connectors that go to your individual modules.*

#### Bus

*An audio pathway by which one or more signals, usually from different sources, are routed to a designated place. Because busses are highly connected to signal flow, they serve a broad range of purposes in audio applications. 2) A shorthand term for the signals themselves that are routed through the bus (see also "Subgroup").*

#### Byte

*Information (data) bits in a grouping of eight. One byte = eight bits.*

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#### Cable Assembly

*Cable that is ready for installation in specific applications and usually terminated with connectors.*

#### Cable Harness

*A grouping of cables or wires used to interconnect electronic systems.*

#### Cable Sheath

*Conductive protective cover that is applied to cables.*

#### Cable

*A group of one or more insulated conductors, optical fibers, or a combination of both within an enveloping jacket, typically for transmitting electrical signals of different types.*

#### Capacitor

*An electronic device made of two plates separated by an insulator, designed to store electrostatic energy. The capacitor is a key component in condenser microphones, for example.*

#### Capstan

*A mechanical part of a magnetic tape recorder that controls the speed of the tape as it passes across the tape heads.*

### Capsule

*Space-travel definitions aside, this is the name given to the part of a microphone that contains the diaphragm and active element, the mechanical structure that converts acoustic sound waves into electrical current.*

### Carbon Microphone

*A microphone that uses carbon granules to convert sound waves to electrical impulses. The carbon element sits between two plates; as sound waves hit the carbon granules, it generates changes in resistance between the plates, affecting the electrical signal.*

### Cardioid Pattern

*A microphone pickup pattern which is most sensitive to sound coming from the front, less from the sides, and least from the back of the diaphragm. So named because the pickup pattern is in the shape of a heart (cardio).*

### Carrier

*There are a few different synthesis techniques where one usually audio-rate signal varies another audio signal. For example, in frequency modulation, a second signal (called the modulator) varies the frequency (pitch) of the main signal, called the carrier. More specifics are described in the entries on frequency modulation and amplitude modulation.*

### Cascade

*To connect or “daisy chain” two mixers so that the stereo mixing busses of the first mixer feed into the stereo busses of the second.*

### CCW

*Counter-clockwise, usually in the context of rotating a control the left (in the opposite direction of how a clock’s hands move).*

### CD

*An abbreviation for Compact Disc, or a small optical disk with digital audio recorded on it.*

### Cent

*When tuning instruments, a semitone is divided into 100 units called cents; there are 1200 cents per octave (100 x 12 semitones). When one oscillator is detuned compared to another, the difference in their frequencies is sometimes measured in cents.*

### Center Frequency

*The frequency of an audio signal that is most affected by an equalizer, either boosting or attenuating the frequency. Drawn graphically, this is the very top or bottom (the “peak”) of the frequency bell-shaped curve.*

### Channel Path

*The complete signal path from the sound source to the multitrack recorder (or DAW). For example, an audio signal that travels from the microphone to the preamplifier, then into a channel strip on the mixing console, then is sent through the outputs into the recorder. This is different from the monitor path, which feeds a mix of signals into monitor speakers or headphones without affecting the recorded signals. (See also “Monitor Path.”)*

### Channel

- 1) *An audio recording made on a portion of the width of a multitrack tape, or isolated within a digital audio workstation, usually for the purpose of combining with other channels.*
- 2) *A single path that an audio signal travels or can travel through a device from an input to an output.*

### Chaotic

*Believe it or not, chaotic does not mean completely random to mathematicians. Chaos theory deals with systems that are random within certain boundaries – such as the path of a wobbling wheel or the frequency of a dripping faucet. Although they are not out of control, neither are they completely predictable. In synthesis, a chaotic system usually refers to a modulation generator that is similar to a low frequency oscillator, but which has unpredictable wobbles or glitches in an otherwise loosely or occasionally repetitive pattern. It can also refer to bursts of triggers that do not follow musical divisions.*

### Chase

*The automatic adjusting of the speed of a recorder (or sequencer) to keep time with another recorder.*

### Chord Chart

*A shorthand form of musical notation that provides the basic chord changes and essential rhythmic information of a song. Most commonly used by studio session players, rhythm sections or jazz bands to provide the skeletal structure of the song while allowing players room to create their own parts and improvise. While lead sheets typically focus on melody line and chord structure, chord charts display mainly chord changes and rhythm. (See also “Lead Sheet.”)*

### Chord

*Three or more musical pitches sung or played together.*

### Chorus

- 1) *The part of a song that is repeated with the same music and lyrics each time, often containing the main point or hook of the song.*
- 2) *A musical singing group with many singers.*
- 3) *A delay effect that simulates a vocal chorus by adding several delays with a mild amount of feedback and a medium amount of depth.*

### Circuit

- 1) *One complete path of electric current.*
- 2) *Similar to definition 1, but including all audio signal paths and components to accomplish a particular audio function.*

### Class Compliant

*This refers to a device that is “plug and play” – it can be plugged directly into a computer or other host and immediately be recognized without additional drivers needing to be installed. This comes up in the modular world with MIDI to CV/Gate interfaces that use USB: If your converter is a USB Host, and you plug a class compliant USB Device such as a controller keyboard or fader panel into it, the converter will recognize it.*

### Click Track

*A metronome “click” fed into headphone monitors for the purpose of helping the musicians play in time with the song.*

### Clip

*All active electronic circuits have a limit on how strong of a signal can pass through them. These limits are often associated with the positive and negative power supply levels. If the signal attempts to go beyond these limits, they instead get chopped or clipped off at that limit. For example, an input voltage of +12 volts may get through without alteration, but +13 volts at the input would come out as 12 volts. This clipping causes distortion in the waveform, usually adding higher harmonics (such as a harsh buzz). Different circuits enter clipping in different ways – some may have a bit of rounding off before they reach that flat threshold; this is referred to as soft clipping and is often desirable as it can be less harsh. Clipping is so named because the resulting graphic waveform looks like the edges of the waveform have been “clipped.”*

### Clock Signal

*A signal sent by a device within the circuit that generates steady pulses or codes to keep other devices in sync with each other. An example in the music world is sequencing via MIDI. The sequencer sends a clock signal so connected devices will play in time.*

### Clock

*Usually refers to the main rhythmic pulse in a system. Often, the clock pulse is much faster than anything it might drive, such as a sequencer or LFO. The most common clock rate is 24 ppqn (pulses per quarter note), as is the case with MIDI clocks and DIN Sync. However, a trigger that drives a sequencer forward one note at a time may also be called the “clock” in a system. Indeed, there are modules that create divisions and multiplications of the main clock to generate new clock signals with a relationship to the main clock.*

### Clockwise

*Clockwise, as in rotating a control the the right – in the same direction as a clock’s hands move.*

### Close Miking

*A microphone placement technique that places the mic close to the sound source to pick up the direct sound and reject ambient sound.*

### Coaxial Cable

*(abbreviated “Coax”) A two-conductor cable that consists of one conductor surrounded by a shield.*

### Coincident Miking

*A stereo miking technique in which two microphones are placed with their heads as close to each other as possible. This prevents phase cancellation problems in the mix because the distance from the sound to either microphone is the same.*

### Compander

*A signal processor serving as a combination compressor and expander, primarily used for noise reduction purposes in analog systems. The audio signal is compressed prior to recording, then expanded at the reproduction stage. Comping is the principle behind Dolby noise reduction systems.*

### Comparator

*An electrical device that compares the level of one voltage to a second. That second voltage may be a second input on a comparator synth module, or may be set with a knob or internal reference voltage. Most often, a comparator outputs a gate signal that goes high when the first signal is higher than the second (or vice versa), and which goes low when the first signal is lower than the second. At audio rates, it converts an input waveform into a square or pulse wave, with the second signal setting when the new waveform goes high or low in voltage.*

### Comping

*1) In digital audio workstations (DAWs), the process of blending portions of multiple recorded takes to create a “compilation” track. (See also “Take,” “Playlist.”)*

*2) In jazz music performance, an abbreviation for “accompanying.”*

### Complex Oscillator

*This module typically has a pair of oscillators behind one panel that is prewired where one oscillator modulates the other’s frequency (known as Frequency Modulation or FM synthesis); some also allow you to quickly switch them so that the first modulates the amplitude of the second, or some other variation. They may also have waveshapers built in. They are based on a popular module created by Buchla, which is a standard of the “West Coast” approach to synthesis.*

### Compression Driver

*A diaphragm that feeds a sound pressure wave into a horn loudspeaker.*

### Compression Ratio

*The rate by which a compressor attenuates an incoming signal, measured in decibels. For example, a compression ratio of 4:1 means the compressor will only allow a 1 dB increase in the signal for every 4 dB increase in the signal above the threshold.*

### Compression

*1) In signal processing, the action performed by a compressor (see also “Compressor”).*

*2) In acoustics, the increased air pressure caused by the peak of a sound pressure wave, used in the context of “compression and rarefaction” (see also “Rarefaction”).*

### Compressor

*A signal processor or plug-in that reduces the dynamic range of an audio signal by amplifying its quieter sections and attenuating its louder ones.*

### Condenser Microphone

*A microphone in which sound is converted into electrical current through changes in a capacitor. The sound pressure waves move the diaphragm, producing changes in capacitance which are then changed into electrical voltage.*

### Contact Microphone

*A microphone designed to pick up vibrations from solid objects (as opposed to vibrations in the air). Also known as a “pickup” or “piezo,” this microphone is often used as an acoustic guitar pickup to pick up the vibrations from the soundboard, or by experimental musicians creating “noise music” from a variety of objects.*

### Control Voltage Processor

*CVP is the abbreviation for a module that allows processing of the voltage going through it – such as amplifying or attenuating it, offsetting it in a positive or negative direction, introducing slew (slurring of changes in voltage), and possibly other functions such as deriving a gate signal from an incoming voltage by running it through a comparator. Make Noise’s Maths is perhaps the most well known control voltage processor out there; you will also find some modules with CVP specifically in their name. Regardless, it’s good to have one or more of this type of module in your system to help massage voltages to get them to do what you want (or to teach them new tricks).*

### Control Voltage

*The concept of control voltage (CV) is at the very root of modular synthesizer. The general idea is that analog voltage levels are used control functions and parameters of a module. For example, one control voltage may determine the pitch played by an oscillator; a second control voltage may determine how loud that signal is after it’s passed through a voltage-controlled amplifier. CV is the most common shorthand to refer to control voltage – for example, when a synthesizer module says it features “CV over the filter’s resonance,” that means there is a control voltage input to control the amount of resonance (feedback) – not just the customary knob on the front panel.*

### Controller

*In the broadest sense, a controller is any device that is used to control another device. Most commonly used in the context of MIDI controllers, which send out MIDI signals to control other connected MIDI instruments and devices. Other examples of controllers in the recording studio can include monitor controllers, DAW controllers and DJ controllers.*

### Corner Frequency

*The cutoff or corner frequency of a filter is the point at which it starts filtering. For example, if a low-pass filter has a corner frequency of 500 Hz (cycles per second), all harmonics or other sound components below 500 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “filtered” – reduced in loudness – the further above 500Hz you go.*

### Counter Clockwise

*Counter-clockwise, usually in the context of rotating a control the left (in the opposite direction of how a clock’s hands move).*

### CPU

*Abbreviation for Central Processing Unit, the main “brain” chip in a computer (also known simply as “Processor”).*

### Critical Distance

*The distance from the sound source at which the direct sound and the reverberant sound are at equal volume. Critical distance varies according to the space; in a room with absorbent walls, the critical distance will be further from the source, and in a reverberant room, the distance will be closer to the source.*

### Crossfade

*An audio editing technique in which one sound is faded out as another sound is faded in, to create a seamless transition between the two. Audio engineers use crossfading, for example, to blend two*

*takes or more “takes” of a recorded track into a composite take. Club DJs also use crossfading to transition from one song to the next with no stops.*

#### Crossover Frequency

*The frequency at which the crossover stops sending the signal to one speaker and starts sending it to another.*

#### Crossover

*An audio filter component that splits an audio signal into two or more bands or signals, usually to be fed into different components of a loudspeaker system according to frequency range. (Also called a “crossover network.”)*

#### Crosstalk

*The unwanted leakage of an audio signal between two audio channels—for example, overlapping signals between channels on a mixing console, or overlapping audio between two tracks of audiotape.*

#### Cue

*In general terms, a cue is the starting point for a piece of music or section of music. Depending on the context, the word “cue” may describe:*

- 1) The point at which a musician or vocalist is supposed to start playing or singing;*
- 2) The audio fed to the musicians through headphones so they can determine when to start playing/singing;*
- 3) A specific location point on the music timeline within a DAW or on the tape; or*
- 4) To set the tape or disc to a certain starting point in the song (“cueing” the tape). A cue can even refer to an entire section of music being used for video production.*

#### Cutoff Frequency

*The cutoff or corner frequency of a filter is the point at which it starts filtering. For example, if a low-pass filter has a corner frequency of 500 Hz (cycles per second), all harmonics or other sound components below 500 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “filtered” – reduced in loudness – the further above 500Hz you go.*

#### Cutoff Slope

*The rate of reduction of the frequencies beyond the passband of a filter. The slope is described as the number of dB the filter reduces the signal for each octave past the cutoff frequency.*

#### CV/Gate

*This is the shorthand to say a synthesizer may be controlled by voltages – usually for pitch – and gate signals to indicate when a note is “on.” An increasing number of controller keyboards are including CV/Gate output in addition to the customary MIDI (Musical Instrument Digital Interface), making them much easier to connect to a modular synthesizer, as no additional MIDI to CV interface is required.*

#### CV

*The concept of control voltage (CV) is at the very root of modular synthesizer. The general idea is that analog voltage levels are used control functions and parameters of a module. For example, one control voltage may determine the pitch played by an oscillator; a second control voltage may determine how loud that signal is after it’s passed through a voltage-controlled amplifier. CV is the most common*

*shorthand to refer to control voltage – for example, when a synthesizer module says it features “CV over the filter’s resonance,” that means there is a control voltage input to control the amount of resonance (feedback) – not just the customary knob on the front panel.*

#### CVP

*CVP is the abbreviation for a module that allows processing of the voltage going through it – such as amplifying or attenuating it, offsetting it in a positive or negative direction, introducing slew (slurring of changes in voltage), and possibly other functions such as deriving a gate signal from an incoming voltage by running it through a comparator. Make Noise’s Maths is perhaps the most well known control voltage processor out there; you will also find some modules with CVP specifically in their name. Regardless, it’s good to have one or more of this type of module in your system to help massage voltages to get them to do what you want (or to teach them new tricks).*

#### CW

*Clockwise, as in rotating a control the the right – in the same direction as a clock’s hands move.*

#### Cycle

*One complete expression of a waveform beginning at a certain point, progressing through the zero line to the wave’s highest and lowest points, and returning to the same value as the starting point. One complete vibration or sound wave.*

#### D-Sub Connector

*Abbreviation for “D-subminiature connector,” a D-sub is a multipin connector that is most often used to connect a computer to a VGA monitor, but also used occasionally in digital audio applications in the recording studio.*

#### D/A

*Abbreviation for Digital to Analog conversion, which changes digital data numbers (digital audio signal) into discrete voltage level. The reverse process of A/D. Also known as DAC.*

#### DADSR

*This is a slightly fancier take on the standard ADSR envelope generator that introduces an initial timed delay before the initial attack stage (rising from 0 to a peak level) begins. One patch idea is to route this type of envelope to a low pass filter cutoff, so there’s initially a muted, filtered sound when the note starts, and then after a pause it starts to swell into a brighter, fuller sound.*

#### Daisy Chain

*The connection of three or more devices in a series, where the audio signal passes through one device to reach a second, and through the second to reach the third, etc.*

#### Damping Factor

*Describes an amplifier’s ability to restrain the pushback motion (back-EMF) of the loudspeaker cone when the audio signal stops.*

#### Damping

*The reduction of energy in a vibrating system, through friction. Can refer to the reduced amplitude*



*in an electrical signal, or the stifled vibrations of a musical instrument (for example, the damper pedal on an acoustic piano).*

#### DAW

*An abbreviation for Digital Audio Workstation, a device or software program designed for recording and mixing audio digitally.*

#### dB

*An abbreviation for decibel, a measurement ratio that compares signal strengths (usually audio levels).*

#### DBX

*A series of noise reduction systems, named for the company that developed them. DBX noise reduction has been less commercially successful than the more widely known Dolby systems, but is still found on occasion in recording studios.*

#### DC Coupled

*When a module says its inputs are DC Coupled, that means it can accept DC voltages (constant or slowly changing voltages) and pass them through unaltered. This is important if, for example, you want to use a VCA to control the amplitude of an envelope going through it: You would need one that was DC coupled, as an AC coupled input would try to remove the DC component of the signal (such as its sustain level) and return it to 0v.*

#### DC

*Electrical current that flows in a single direction, as opposed to Alternating Current (AC), which flows in alternating directions. Many electronic devices run on DC, which is usually provided by battery power, USB power or an AC adapter plugged into the wall.*

*In modular terms, DC refers to a voltage that tends to stay at one steady level for awhile, such as a gate output that switches between 0v when a note is off and 5 or 10v when a note is on. It can also refer to a slowly changing voltage, such as an envelope.*

#### DCO

*A DCO (Digitally Controlled Oscillator) is a hybrid design for an analog oscillator that – instead of using a voltage level to determine the pitch of the oscillator – uses a digital device such as a counter to determine the length of each waveform cycle and therefore the pitch. On the plus side, tuning is very stable, unlike some all-analog designs. On the minus side, there are no imperfections in pitch that cause subtle detuning (and therefore the perception of “fatness”) when using more than oscillator per voice.*

#### De-esser

*An audio compressor designed to reduce the volume of sibilant sounds and frequencies, especially those produced by pronouncing the letter “s.”*

#### Decay

*In general, decay refers to a voltage or overall level dropping down from some high point, such as the decay stage of an envelope generator. A real-world analogy is that after you initially strike a drum or pluck a string, it decays in volume from its initial loudness eventually all the way to silence. It can also refer to the tail of a reverb or echo effect where the sound dies away over time.*

### Decca Tree

*A stereo microphone placement technique involving three microphones (usually omnidirectional) placed in a “T” pattern. Commonly used in miking choirs, orchestras and other large ensembles, but variations of the Decca tree technique are also being used today in surround sound situations.*

### Decibel

*(abbreviated “dB”) The ratio measurement of two levels according to a scale where a certain percentage change comprises one unit. Most often used to describe audio levels.*

### Degaussing

*The process of demagnetizing an object. In the context of audio, degaussing essentially erases the recording on magnetic tape.*

### Delay/Attack/Decay/Sustain/Release

*This is a slightly fancier take on the standard ADSR envelope generator that introduces an initial timed delay before the initial attack stage (rising from 0 to a peak level) begins. One patch idea is to route this type of envelope to a low pass filter cutoff, so there’s initially a muted, filtered sound when the note starts, and then after a pause it starts to swell into a brighter, fuller sound.*

### Delay

*You all know what the word delay means in the normal world; it can appear in different forms inside a modular synth. For example, it can refer to the spacing between repeats in an echo; that’s why an echo device is often known as a “delay” effect. It can also refer to a programmable amount of time you delay a signal, such as a gate, trigger, or initial stage of an envelope so a note would start later than it was actually played.*

*\*Also, 1) An process by which an audio signal is recorded to a medium or device, reproduced at a time delay, then mixed with the original, non-delayed signal to create a variety of effects such as a fuller sound, echo, chorusing, flanging, etc. \**

*2) A signal processor that creates delay effects.*

### Demo

*A preliminary recording that is intended to give the listener an idea of how a song could sound in a final production. A demo usually involves minimal tracking or production, almost like a “rough draft” of a recording.*

### Detune

*If you have two oscillators tuned to exactly the same frequency – and I mean, exactly the same frequency – there’s not much point in having more than one oscillator. However, when you change the tuning of one ever so slightly – in other words, detune it – you will start to hear interesting interactions between the two, often referred to as chorusing or beating. The result tends to be more interesting and “full” – and a bit more natural, as two singers or instruments can rarely hit exactly the same note.*

*To purposely cause an instrument or signal to play out of tune (usually slightly). This effect can be used for a number of purposes in the studio, but is often used in “double-tracking,” blending the detuned instrument/track with the original to create a fuller sound.*

### DI

*The process of sending an electrical audio signal directly from an instrument to the mixing console through the use of electric pickups or direct boxes, as opposed to using a microphone.*

#### Dialogue

*The spoken word recorded in film/video sound, commercials and instructional recordings.*

#### Diaphragm

*The part of a microphone that moves in response to sound waves, converting them to electrical signals.*

#### Difference

*A fancy way of saying you subtracted on control voltage from another. It can also be applied to audio or harmonics.*

#### Digital Audio Workstation

*abbreviated DAW) A device or computer software that records and mixes audio digitally and creates digital audio files. A DAW can be a standalone unit or an integrated set of components, but today they are most commonly found as “in-the-box” software programs run from a computer. The most common DAW program found in recording studios is Pro Tools; other commonly used programs include Reason, Ableton and Logic.*

#### Digital Multimeter

*A small device that tests electrical voltage, current, and resistance. Multimeters are useful in recording studios for calibrating electrical systems and troubleshooting problems.*

#### Digital Recording

*The process of converting audio signals into numbers that represent the waveform, then storing these numbers as data.*

#### Digital Signal Processing

*(abbreviated “DSP”) Any signal processing done after an analog audio signal has been converted into digital audio.*

#### Digital to Analog Converter

*(abbreviated D/A) A device that converts the digital data of digital audio into voltage levels that approximate the original analog audio.*

#### Digital

*There was a time when digital (referring to circuitry based around binary logic, computers, and the such compared to the old-fashioned transistors, op amps, capacitors, and other bits that make up analog circuitry) was a dirty word among synthesists. The assumption was digital techniques created sounds that were more sterile, brittle, and abrasive – and just not as “authentic.” Today, digital circuitry is embraced in synthesizers, including modular systems. Although analog will always hold a special place in our hearts, a well-implemented digital circuit can sound just as good as an analog one, while digital signal processing and programming can create a wider range of sounds than most analog circuitry.*

#### Digitally Controlled Oscillator

*A DCO (Digitally Controlled Oscillator) is a hybrid design for an analog oscillator that – instead of using a voltage level to determine the pitch of the oscillator – uses a digital device such as a counter to*

*determine the length of each waveform cycle and therefore the pitch. On the plus side, tuning is very stable, unlike some all-analog designs. On the minus side, there are no imperfections in pitch that cause subtle detuning (and therefore the perception of “fatness”) when using more than oscillator per voice.*

#### DIN Stereo

*A stereo microphone placement technique that places two cardioid microphones about 20cm apart and set outward from each other at a 90-degree angle to create a stereo image. Particularly for stereo miking at close ranges. (See also “Near-Coincident Miking.”)*

#### DIN Sync

*A clock signal for controlling the tempo of sequencers, arpeggiators, and drum machines, distributed using cables with DIN-style connectors (yes, just like old-fashioned MIDI connectors, but DIN Sync is even older). Roland pioneered this standard, which included sending 24 pulses per quarter note (PPQN), giving rise to the alternate name Sync24. Korg equipment used a variation of this running at 48 pulses per quarter note, also known as Sync48. DIN Sync is still a popular way of sending a clock signal to a modular synth today, especially when interfacing with other vintage synthesizers, sequencers, and drum machines.*

#### Diode Ladder Filter

*This is a filter design most often associated with the Roland TB-303 Bass Line, which is known for its rubbery sound with eager resonance.*

#### Diode

*An electrical component that enables easy electrical current flow in one direction but not the other. In the recording studio, these are commonly found in the vacuum tubes of tube amplifiers.*

#### Direct Box

*A small device that converts an unbalanced, high-impedance speaker or instrument-level output to a balanced, low-impedance mic-level output. Frequently used in the signal path connecting electric instruments “directly” to the mixing console, as opposed to miking them acoustically. Also called “direct injection box” or “DI box.”*

#### Direct Current

*In modular terms, DC refers to a voltage that tends to stay at one steady level for awhile, such as a gate output that switches between 0v when a note is off and 5 or 10v when a note is on. It can also refer to a slowly changing voltage, such as an envelope.*

*(abbreviated “DC”) Electrical current that flows in a single direction, as opposed to Alternating Current (AC), which flows in alternating directions. Many electronic devices run on DC, which is usually provided by battery power, USB power or an AC adapter plugged into the wall.*

#### Direct Injection

*(abbreviated “DI”) The process of sending an electrical audio signal directly from an instrument to the mixing console through the use of electric pickups or direct boxes, as opposed to using a microphone.*

#### Direct Out

*An output available on some consoles which is fed directly from the preamplifier stage of the input,*

*bypassing the channel strips and faders. This feature is often used to send a “dry” signal to a monitor mix or a recording device.*

#### **Direct Sound**

*The sound that reaches a microphone or a listener’s ear without hitting or bouncing off any obstacles (as opposed to reflected or ambient sound).*

#### **Directional Pattern**

*1) In microphones, a term meaning the same thing as “Pick Up Pattern,” a description of the area in which a microphone is most sensitive to sounds.*

*2) In loudspeakers, it is the pattern of dispersion, the area that the sound from a speaker will evenly cover in a listening area.*

#### **Dispersion (also Dispersion Angle)**

*The area that is effectively covered by the sound coming from a loudspeaker; specifically, the imaginary boundaries on either side of the speaker at which the sound level is 6 dB lower than if you were standing directly in front of the speaker. Each speaker has both a horizontal and vertical dispersion angle.*

#### **Distant Miking**

*The technique of placing a microphone far from the sound source in order to pick up a combination of the direct and reflected sounds.*

#### **Distortion**

*Refers to the deforming of a waveform at the output of a device as compared with the input, usually due to overload, creating a distorted or “dirty” signal. While electrical or audio distortion is typically unwanted and avoided, it is frequently used in controlled situations in audio to create certain desirable effects, particularly with electric guitars and amplifiers.*

#### **Diversity**

*1) In audio settings: the use of two or more antennas in a wireless receiver system to prevent dropouts in the audio from a wireless microphone.*

*2) In other settings: the embracing of the uniqueness of all individuals.*

#### **Dolby**

*The brand name of a manufacturer of noise reduction systems and other audio systems, to improve performance and fidelity of audio recording, playback, and transmission.*

#### **Doppler Effect**

*The phenomenon in which the human ear perceives a change in the frequency (pitch) of a sound while the sound source is in motion. As the sound source approaches, the sound waves travel a shorter distance to the ear, increasing the frequency of the waves and the pitch of the sound; as the sound source moves away, the sound waves must travel farther and farther, resulting in lower frequencies. A common example of this effect is an approaching emergency vehicle whose siren sounds higher as it approaches and lower after it passes. The Doppler Effect can be utilized in audio settings, for example, in the Leslie speaker in which an electric motor rotates the speakers inside the cabinet, constantly changing the distance between the sound source and the listener (or microphone) and creating its signature warbling vibrato effect.*

### Double

- 1) To record a second performance closely matching the first performance, for the purpose of blending the two tracks.
- 2) To use a delay line with medium delay to simulate double tracking.

### Driver

- 1) A transducer in a loudspeaker that converts electrical signals into sound pressure waves.
- 2) A computer program that controls an attached device or piece of hardware.

### Dropout

A brief loss of audio signal on tape, or a brief loss of data in a digital audio file (often due to a dropped sample), that can result in an unwanted dip in audio, a crackle or a pop.

### Drum Machine

An electronic device containing synthesized and/or sampled drum sounds in its memory, along with an internal sequencer that can be programmed to play drum patterns or loops.

### Drum Pattern

A specific sequence of drum sounds played by a drummer or sequenced into a drum machine for use in a song.

### Dry

A sound with no effects is referred to as “dry”; a sound with effects (such as reverb) mixed is referred to as “wet.” Effects units or mixers often have wet/dry mix amounts that set the ratio between the original, unprocessed sound and the fully-effected sound.

### DSP

Any signal processing done after an analog audio signal has been converted into digital audio.

### Dub (or Dubbing)

- 1) To copy a recording.
- 2) To record in real time with another recording with the intent of mixing the two recordings (see also “Overdub/Overdubbing”).
- 3) “Dub” is an abbreviation for “dubstep,” a style or subgenre of electronic music.

### Ducking

A compression-based audio effect in which an audio signal is reduced proportionately by the presence of another audio signal, sometimes accomplished through a “sidechain” connection with the signal processor. A notable example is a spoken-word voice-over track recorded over a musical track, where the music drops in volume when the speaker begins to speak. A more subtle example is when an audio engineer “ducks” specific sounds to make room for others in the track; for example, when a bass guitar signal triggers a slight reduction in the level of drums or guitars. (See also “Sidechain.”)

### Duophonic

Duophonic means two “voices.” Most early synths (including modular systems) are monophonic, which means they can play only one note at a time; some instruments have enough oscillators, filters, envelopes, and amplifiers that they could play two separate notes as once. Some MIDI interfaces for modular synths include duophonic modes so you can patch up and control two separate voices

*from your keyboard. Some users play fast and loose with terms such as duophonic, monophonic, and polyphonic.*

#### Duration

*Duration is another way of saying length. A clock pulse or a gate signal that is “high” for a certain amount of time – say, 100 msec – is said to have a duration of 100 msec. The length of time you hold a note down, or the length of a step in a sequence, is also called its duration.*

#### Dynamic Microphone

*(Also called Moving Coil Microphone) A microphone in which sound pressure waves are converted to an electrical audio signal by an induction coil moving within a magnetic field—a process often compared to a loudspeaker working in reverse. Dynamic microphones are less sensitive than condenser microphones, but can be effective for miking louder sound sources or for close-miking applications.*

#### Dynamic Processing/Dynamic Signal Processing

*The process of automatically changing the level (or gain) to alter the level relationship of the loudest audio to the softest audio. Dynamic processors include compressors, limiters, expanders and gates.*

#### Dynamic Range

- 1) *The ratio (in dB) between the loudest peak and the softest level of a song or recording.*
- 2) *The ratio (in dB) between the softest and loudest possible levels a device or system can provide without distortion.*

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#### Early Reflections

*The first sound waves that reach a listener’s ear after bouncing off a surface in the room, usually heard almost immediately after the initial sound. The first stage of reverberation.*

#### East Coast Synthesis

*This blanket term is applied to most common synthesizer configuration pioneered by East Coast based companies such as Moog, Arp, and EML (as well as “Far East” companies such as Roland and Korg) where one or more oscillators producing waveforms with rich harmonic content (such as a sawtooth or square wave) are fed into a filter that removes some of those harmonics, and then onto an amplifier to shape the loudness of a note. This approach is also often known as subtractive synthesis, as the filter reduces (subtracts) harmonics that came from the oscillators. East Coast synthesizers also regularly have organ-style black & white keyboards, and four stage ADSR type envelopes. Today it’s common to mix both East Coast and West Coast approaches in the same system.*

#### Echo Chamber

*An enclosed room designed with reflective, non-parallel surfaces for the purpose of creating acoustic echoes (reverberation).*

#### Echo

*The distinct repetition of an initial sound, caused by the reflection of the sound waves upon a surface.*

*We recognize a sound as an echo when the distance between the source and the reflection is far enough apart that we can detect the time delay between one and the other. Essentially, reverberation is the combination of many echoes occurring too rapidly to hear each individually. In the studio, echoes can be reproduced acoustically or simulated by a digital signal processor.*

### Edit

*To change one or more parameters of a recorded sound after the fact. This can take many forms, including “punching in” a section of the music that is re-recorded to replace the original version; altering the shape/size of waveforms graphically; changing the sequence of playback; and many others. Analog editing would typically involve splicing the magnetic tape on which the audio signals were recorded. These days, almost all editing in the studio is done via computer using a digital audio workstation (DAW).*

### Effect Loop

*Sometimes you might want to send a signal outside your modular system, process it through an external effects device, and bring it back into your modular for more processing. This going out/coming back in is referred to as an effect loop. The trick with modular synths is that their internal signal levels tend to be much higher than those used by external effect equipment, so a modular effect loop will usually have level matching circuitry as well.*

### Effects Processor

*(Also called Guitar Processor) A device that adds audio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, etc. Effects processors can occur as individual effects boxes or multi-sound pedal boards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between the guitar and the console. They can also be found as presets in guitar amplifiers, or even as digital plug-ins within a DAW.*

### Effects Track

- 1) In film production audio, a recording of the mixdown of all the sound effects ready to be mixed with the dialogue and music.*
- 2) In music recording, one track with a recording of effects to be added to another track of a multitrack recording.*

### Effects

- 1) Various ways an audio signal can be modified by adding something to the signal to change the sound.*
- 2) Short for the term Sound Effects (sounds other than dialogue, narration or music like door closings, wind, etc.) added to film or video.*

### EG

*The envelope generator (EG) module is used to shape the loudness or dynamics of a note when connected to a VCA (Voltage Controlled Amplifier), as well as how its frequency content or timbre changes over time when connected to a VCF (Voltage Controlled Filter). To do this, an envelope generator creates a voltage that typically rises from zero volts to some maximum level, and back down again. You control how long this takes, usually in various stages: an attack stage as it goes from zero to max, a decay stage as it falls back down from maximum to either zero (in the case of an*



*AD, or Attack/Decay envelope) or an intermediate level known as the sustain, and then (usually after a key has been released and the corresponding gate signal has gone back to zero) from the sustain level back to zero over a duration known as its release.*

#### Electret Microphone

*A variation of condenser microphone that uses an electret instead of a capacitor. (Also called “Electret Condenser Microphone.”) Because the electret is permanently polarized, an electret microphone does not require an external power source as a standard condenser microphone does.*

#### Electret

*A dielectric plate that is designed with permanent polarity, allowing it to function similarly to a magnet. (“Electret” comes from the words “electricity” and “magnet.”) Used in some microphone types in place of a capacitor (condenser).*

#### Electromagnetic Field

*(Abbreviated EMF) A field of magnetic energy put out because of current traveling through a conductor.*

#### Electromagnetic Interference (EMI)

*The bane of audio professionals everywhere, EMI is a type of interference caused by nearby electromagnetic activity, which can be picked up by audio cables and equipment, causing unwanted noise, hum or buzz in audio systems. Common causes of EMI in audio systems may include high-current power lines, fluorescent lighting, dimmer switches, computers, video monitors and radio transmitters.*

#### Electrons

*Negatively charged particles revolving around the nucleus of an atom. Electrical current is generated by electrons moving along a conductor, like a metallic wire.*

#### Emphasis

*This word can have two meanings. In a normal audio context, it usually means some form of high frequency boost, as emphasizing the higher harmonics can add clarity to a tone and help distinguish it from another. In synthesizers, emphasis usually means the Q or resonance setting on a filter, as increasing this setting boosts (emphasizes) the harmonics at the cutoff or corner frequency.*

#### Envelope Follower

*This module follows the loudness contour of a sound, and outputs a voltage that corresponds to how that loudness changes. They tend to perform some smoothing on this signal so that it’s not too nervous or jumpy in nature. Envelope followers often also have a gate output that goes high when the loudness of the input signal went over a certain level, and low when it falls back below that level.*

#### Envelope Generator

*The envelope generator (EG) module is used to shape the loudness or dynamics of a note when connected to a VCA (Voltage Controlled Amplifier), as well as how its frequency content or timbre changes over time when connected to a VCF (Voltage Controlled Filter). To do this, an envelope generator creates a voltage that typically rises from zero volts to some maximum level, and back down again. You control how long this takes, usually in various stages: an attack stage as it goes from zero to max, a decay stage as it falls back down from maximum to either zero (in the case of an*

*AD, or Attack/Decay envelope) or an intermediate level known as the sustain, and then (usually after a key has been released and the corresponding gate signal has gone back to zero) from the sustain level back to zero over a duration known as its release.*

#### Envelope Tracking

*This describes the main action of an envelope follower: a module or section of a module that follows the loudness of a signal and outputs a voltage that corresponds to – tracks – that input.*

#### Envelope

*The collective term for the four elements of the lifespan of a sound: Attack, Decay, Sustain and Release (ASDR). The envelope of a sound describes how a sound or audio signal varies in intensity over a period of time.*

#### Equal Loudness Contours

*A drawing of several curves showing how loud the tones of different frequencies would have to be played for a person to say they were of equal loudness. (See also “Fletcher-Munson Curves.”)*

#### Equalizer

*An audio signal processor that uses one or more filters to boost or cut the amplitude (volume) of certain frequencies within the sound. The underlying principle is to balance or “equalize” the frequency response of the audio system, or to create balance between multiple signals in a sonic space. However, audio engineers may use equalizers to alter or “color” the sound in many different ways.*

#### Eurorack

*Eurorack is arguably the most popular format of modular synthesizer today, with over 100 manufacturers and over 1000 modules available. It was created by Doepfer Musikelektronik in 1995, basing its size off the Eurorack format for lab equipment. Some users will try to tell you that Eurorack doesn’t “sound” as good as other formats, but that’s just based on a few substandard manufacturers or modules; there’s nothing inherent to the standard that makes a huge difference in the final sound (no; the difference between 12 and 15 volt power supplies is not enough to most ears).*

#### Expander

*A signal processor (or plug-in) that performs the opposite function of a compressor, expanding the dynamic range of an audio signal rather than compressing it. It accomplishes this by further reducing the amplitude of signals that drop below a set threshold.*

#### Expansion Ratio

*The rate by which an expander attenuates an incoming signal, measured in decibels. For example, an expansion ratio of 2:1 means the expander will reduce the signal by 2dB for every 1dB it drops below the threshold. If the signal falls 3dB below the threshold, the expander attenuates it by 6 dB, and so on.*

#### Exponential

*In general terms, this is a mathematical curve that starts out relatively flat and then bends to climb steeply. In synthesizer terms, it most often refers to the control voltage scheme where a change of 1 volt corresponds to an increased pitch of one octave, which is doubling in cycles (vibrations) per*

*second. This is in contrast to a linear system where 1 volt increase would always result in the same increase of cycles per second.*

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#### Fade

*A gradual reduction of the level of the audio signal, or a gradual change of level from one pre-set level to another.*

#### Fader

*A control which adjusts the level (gain or attenuation) of an incoming signal to a channel or grouping of channels on a console.*

#### Far Field

*The region away from a loudspeaker at which the sound drops 6dB for each doubling of the distance, up to the critical distance. The beginning of the far field varies according to the size of the speaker, but in most cases the far field begins around 3 feet from the sound source. Audio engineers often use both near field and far field monitoring when fine-tuning a mix. (See also "Critical Distance," "Near Field.")*

#### Feed

*To send an audio or control signal to.*

#### Feedback Control

*The control on a delay line or delay effects device that controls the amount of feedback into the system.*

#### Feedback

*The return of a portion of the output signal back into the input of a system. This can be done in a controlled manner through a feedback circuit to alter the sound of an instrument (most commonly electric guitars or analog synths). It can also describe the unwanted feedback loop created when an open microphone is picking up the sound from a nearby speaker, generating a loud, oscillating frequency that increases in intensity until the feedback loop is broken by turning off the mic or speaker, or by use of an equalizer to attenuate the frequency.*

#### Fidelity

*A term describing how accurately a sound is reproduced from its original source.*

#### Figure-8 Pattern

*A microphone pickup pattern which is most sensitive to picking up sounds directly in front and back of the mic, effectively rejecting sounds coming from the sides.*

#### Filter

*A module that reduced or removes certain frequencies and harmonics from the sound that is passed through it. In a synthesizer, the most typical filter types are low pass (passes all of the harmonics below its cutoff or corner frequency untouched, and then reduces the level of higher harmonics the further you go above that cutoff frequency), high pass (passes all harmonics above its cutoff frequency*

*untouched, and reduces the level of progressively lower harmonics below the cutoff), bandpass (harmonics right around the cutoff are passed intact, and then reduced more in level the further away they are above or below the cutoff frequency), and notch (harmonics right around the cutoff frequency are reduced or cut out entirely; others above or below are allowed to live).*

### Flanger

*A signal processor often identified as the one that creates a “jet taking off” whoosh. What’s going on behind the panel is that a copy of the input signal is delayed by a very small amount (longer than a chorus effect; shorter than an echo effect) and mixed in with the original. When the delay is constant, the result is a “comb filter” where certain harmonics are cancelled out as they are mixed back on top of themselves out of phase. When the delay is varied over time, you get swooshes and sweeps. The effect was originally created by playing two tape reels of the same song, starting them in time with each other, and dragging your finger on the flange of one of the tape reels to delay it.*

### Flanging

*An audio effect caused by blending the signal with a copy of that signal at a slight time delay, then modifying the delayed copy, creating a “swirling” sound. This was originally accomplished in analog tape recording by playing the original tape and the copy on two tape machines simultaneously, then physically pressing on the flange of one of the machines to alter the timing of the duplicate track. These days, most flanging is done through delay boxes or digital plug-ins.*

### Flat

- 1) A term used to describe an even frequency response in a device or speaker, meaning that the device/speaker treats all frequencies the same without the need for EQ. When displayed graphically, the frequency response is shown as a “flat” line with no peaks or valleys.*
- 2) In music, describes a note or pitch that is out of tune, sounding at a slightly lower frequency than it should.*
- 3) In music notation, an “accidental” mark that instructs the player to play/sing the note one-half step lower.*

### Fletcher-Munson Curves

*Also known as “Equal Loudness Contours,” a set of graphical curves plotted to illustrate how the human ear responds to different frequencies at different volume levels. Named after the two researchers who first plotted the curves. (See also “Equal Loudness Contours.”)*

### Flip-Flop

*In binary logic terms, a flip-flop toggles between high and low every time it receives an input trigger (i.e. the first trigger would set the output high, the second trigger sets it low again, and so on). In clock or audio terms, it divides the speed of an input clock or square wave by 2.*

### Floating Unbalanced Line

*A connection “workaround” in which an unbalanced output is connected to a balanced input by modifying the connections in the line to resemble a balanced line, alleviating unwanted hum or buzz.*

### Fly In

*To add sounds into a mix or recording that have no synchronization.*

### Flying Bus

*This is a very simple type of power distribution or bus board that typically uses a ribbon cable with multiple connectors along its length to take the output of your power supply and distribute it to your individual modules. They're cheap and easy to install and use, but in a few cases might be a cause of noise being shared between modules.*

### FM

*Frequency modulation (FM for short) refers to a synthesis technique where the pitch of an oscillator is varied (modulated) very quickly – at audio rates – by another oscillator. The result is a complex side of harmonics that may either be nicely in tune or clangorous and “out of tune” with the fundamental pitch of the main oscillator.*

### FOH

*In live audio settings, the location in a venue opposite the stage, where live audio for the show is controlled and mixed.*

### Foldback

*A stage monitoring system used in live audio. A set of on-stage speakers called monitors or wedges (or “foldback speakers” in British countries) are fed a special mix of audio signals for the onstage performers to hear in order to play. This mix is usually different from the FOH (front-of-house) mix that the audience hears, and is sometimes controlled by a second engineer through amplifiers and speakers separate from the main sound system. This type of stage monitoring is frequently susceptible to feedback from the microphones, and in certain venues can cause unwanted reflective noise that makes it difficult for FOH engineers to create a good mix for the audience. For this reason, many live audio systems now use in-ear monitoring as an alternative to stage monitors to control the onstage noise and reduce the risk of feedback.*

### Foot Pedal

*An effects device controlled by a musician with his foot.*

### Foot Switch

*A switch placed on the floor and pressed by a musician to do various functions.*

### Force-Sensing Resistor

*In modular systems, an FSR (Force-Sensing or -Sensitive Resistor) usually takes the form of a circular pad that you press on to vary a parameter. It acts as a resistor that decreases in resistance the harder you press.*

### Formant

*Many instruments based on vibrating tubes – including our own vocal tract – have certain frequencies that they like to vibrate or “resonate” at. When you send a sound down these tubes, they will accentuate the frequency of that sound (or some of its harmonics) to match these resonate frequencies. Each of these resonant frequencies is known as a formant of that instrument. A common way of synthesizing vocal-like sounds is to pass an oscillator through a filter or equalizer that has several formant peaks, spaced apart in ways that mimic certain vowels.*

*Formant is an element in the sound of a voice or instrument that does not change frequency as different pitches are sounded. Formants are essentially “fixed” frequencies or resonances that occur*

*as a result of the physical structure of the sound source. These frequencies are what create timbre, that element of sound that creates the specific sound of a guitar, a flute, a male or female voice, etc.*

#### Format

*1) One of many different media used to store and reproduce audio, whether in the recording studio or for listening purposes. Examples include currently used physical formats such as vinyl records and compact discs; obsolete formats such as cassette tape, 8-track tape and DAT; analog recording staples such as reel-to-reel multitrack tape; and many different digital audio file formats such as mp3, WAV, WMA, AIFF and others.*

*2) Format can also describe specific parameters when recording to analog tape, such as number of tracks, width, spacing and order.*

*3) To prepare a hard drive or memory card for use, usually erasing all existing data in the process.*

#### Four Quadrant Multiplier

*A Four-Quadrant Multiplier is a special case of Amplitude Modulation (AM). It is also referred to as ring or balanced modulation. One signal changes the level of – “multiplies” – the level of a second signal. A typical use is two VCOs running at audio rates fed into a ring modulator (a four-quadrant multiplier). The output is a complex set of component tones that don’t follow typical “musical” spacing based on octaves above the fundamental that harmonics usually follow. Namely, the modulation frequency is both added to and subtracted from the carrier’s frequency; the resulting harmonics replace the original carrier and modulator. Say the carrier was a sine wave (only the fundamental harmonic present) at 600Hz, and the modulator was a sine wave at 100Hz. The result would be a tone that had frequency components at 500 and 700Hz.*

#### FracRack

*A less-common format of modular synthesizers put forward by PAiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5” (3.8 cm) wide by 3U, or 5.25” (13.3 cm) high.*

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#### Frequency Modulation (FM) Synthesis

*A method of sound synthesis in which the frequencies generated by one oscillator (the carrier) are altered by the output of one or more additional oscillators (operators) to create a diversity of harmonically rich sounds.*

#### Frequency Range

*1) The range of frequencies over which an electronic device puts out a useful signal (see also “Bandwidth”).*

*2) The range of frequencies that can be substantially transmitted or received in relation to a sound source. Each instrument has a certain frequency range in which it can play; the human ear can also hear within a certain frequency range.*

#### Frequency Response

*The range between high and low frequencies that a component of an audio system can adequately handle, transmit or receive.*

### Frequency-Agile

*In wireless microphone systems, frequency-agile describes the ability of the system to operate on a choice of different RF frequencies within a certain bandwidth. Frequency-agile systems are preferred for live touring and in areas with high concentrations of radio signals (like large cities) because the RF frequency of the device can be changed to avoid interference.*

### Frequency-Shift Key (FSK)

*A now out-of-date protocol in which a sync tone is recorded onto a spare track of a multi-track tape recorder to enable electronic devices (mainly drum machines) to perform in sync with the tape. While some older devices still read FSK, an updated protocol (Smart FSK) is now more commonly used. (See also "Smart FSK.")*

### Frequency

*The number of occurrences of a particular event within a certain amount of time. In audio and acoustics, frequency specifically refers to the number of complete cycles a vibration or waveform makes in a second, measured in cycles per second, or Hertz (Hz). In sound, frequency determines what we hear as pitch. The longer the wavelength, the fewer the cycles per second, and the lower the pitch.*

### Front-of-House

*(Abbreviated FOH) In live audio settings, the location in a venue opposite the stage, where live audio for the show is controlled and mixed.*

### FSR

*In modular systems, an FSR (Force-Sensing or -Sensitive Resistor) usually takes the form of a circular pad that you press on to vary a parameter. It acts as a resistor that decreases in resistance the harder you press.*

### Full-Normalled

*Describes the configuration within a patch bay in which the jacks form a connected pathway until a patch cord is inserted to change the path. When a patch bay is "full-normalled," the connection is altered by inserting a cord into either the input or output side; when it is "half-normalled," the path changes only when a cord is plugged into the input. "Non-normalled" or "open" means there are no internal connections, and each input sends the signal through its corresponding output.*

### Full-Wave Rectifier

*A full-wave rectifier takes any negative voltages and inverts them so they become positive. This effectively doubles the frequency of many simple waveforms, like the triangle and sine.*

### Function Generator

*The term function generator can have two meanings in the world of synthesis. One, test equipment that generates waveforms such as sine or square waves are often called "function generators." Two, envelope generators are sometimes referred to as "function generators." In both cases, "function" means to execute an equation of some sort, such as creating a periodic waveform such as a sine or creating a rise & fall in response to a trigger.*

### Fundamental

*(Also called fundamental frequency or first harmonic) The lowest frequency present in the sounding*

*of a note by musical instrument or voice.*

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#### Gain Control

*A device that changes the gain of an amplifier or circuit, often a knob (potentiometer) that can be turned. In a mixing console, each channel usually has its own gain control to regulate the gain of the signal coming into the board—not to be confused with the channel “fader,” which regulates the output of an already-amplified signal.*

#### Gain Reduction

*The action of a compressor or limiter in regulating the amplitude of the audio signal.*

#### Gain Structure

*A term that describes the interconnection of multiple components in an audio system, and the amount of gain increase or reduction that occurs at each point. A configuration with a good gain structure means that the components are working properly together to provide optimal gain with minimal distortion or noise.*

#### Gain

*The amount of increase in audio signal strength, often expressed in dB.*

#### Gate Detector

*This is one of the main signal types that are passed around inside a modular synthesizer. It jumps to high level – typically 5 volts – when a new note is supposed to start (such as when you press a key on a keyboard controller), or when a sequencer jumps to the next “stage” or note. A gate typically stays at that level for the duration of the note (i.e. while the key is being held down), and suddenly drops or “goes low” to its resting level – typically 0 volts, but sometimes –5 volts or another number – when the note ends (i.e. when the key is released). In practice, when a gate signal is sent to a typical envelope generator, the start of the gate (when it “goes high”) tells the envelope to go through its Attack and Decay stages; while the gate remains high, the envelope stays at its Sustain level, and when the gate goes low again, the envelope moves onto its Release stage.*

#### Generation Loss

*The amount of clarity lost when recorded audio is copied, due to added noise and distortion.*

#### Generation

*A term used to describe the number of times that the recorded audio signal has been copied.*

#### Glide

*Refers to a note that glides from one pitch to another while it is still audible. The music term for this effect is portamento, which is a slurring between notes. In a synthesizer, this effect is created by causing the control voltage for the pitch of a note to slide from the pitch of the previous note rather than make a discrete jump. The module that creates this effect is sometimes known as a slew generator, slew limiter, slope generator, or lag. Some use the terms glide, glissando, and portamento interchangeably, but if you want to split musical hairs, a glissando (gliss) is a different effect where*



*the intermediate notes are more distinct – such as played rapidly in order – rather than slurred through.*

#### Golden Section

*(also called Golden Ratio) A ratio of height to width to length, where the width is approximately 1.6 times the height, and the length approximately 2.6 times the height. First calculated by the ancient Greeks, this ratio (known mathematically as “phi”) is used as an optimal ratio in many applications, including room dimensions and studio design (to achieve “optimal acoustics” in the room), and even in the design of certain acoustic instruments.*

#### Granular Synthesis

*Granular synthesis can be thought of as particle theory applied to sound. The concept is that a sound can be broken down into very small “grains” – typically 1-50 or 100 msec in duration. These tiny snippets are then played back to reproduce the original sound, or to create new sounds by changing the speed, pitch, volume, playback order, and direction of the individual grains. You can crossfade between these modified grains, or layer more grains on top. The result can range from audio processing tricks such as changing speed without changing pitch and vice versa, to creating psychedelic “clouds” of sound (and indeed, there is a popular module called Clouds).*

#### Graphic Equalizer

*A type of equalizer that can adjust various frequencies of the incoming signal using sliders that are assigned to specific frequency bands. (See also “Equalizer.”)*

#### Ground Lift Plug

*An adapter that enables a three-prong power cord to plug into two-prong outlet. Some engineers wrongly use this plug to interrupt the ground connection and prevent buzz, but it is a VERY unsafe practice to break the ground connection using this plug without grounding the unit by another means.*

#### Ground Lift Switch

*A switch that breaks the connection between the ground point in one circuit and the ground point in another circuit, for the purpose of eliminating hum or buzz caused by ground loops.*

#### Ground Loop

*A situation caused when one or more electronic devices are connected to the same ground at different points. The devices operate at different ground potentials, which creates voltage along the ground, resulting in a low-frequency hum that can be annoying at best and cause damage to gear at worst. The best resolution for ground loops is to ground all devices at the same point using a central power source. An alternative solution is to break the loop via ground lift switches or plugs, but this should be avoided when possible as it is considered an unsafe management of electricity.*

#### Group (or Grouping)

*A number of input channels on a console that can be controlled and adjusted as a single set before sending the combined signal to the master output. Sometimes also called “Submix,” “Bus” or just “Group.”*

#### Group Delay

*In audio, group delay is a phenomenon within all electronic audio devices (e.g., speakers, amplifiers) in which different frequencies in the signal are output at slight delays from one another. In simpler*

*terms, lower frequencies are delivered slightly more slowly than higher ones. In all devices, there is an inherent delay between input and output of the signal, but group delay specifically deals with the time delays between specific frequencies of the sound. The goal in any configuration is to keep the group delay as small as possible; in cases of extremely poor configurations, the delays between highs and lows can be audible.*

#### Guitar Controller

*An electric guitar (or device played like a guitar) that transmits MIDI data that can be used to control synthesizers and sound modules.*

#### Guitar Processor

*A device that adds audio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, etc. Effects processors can occur as individual effects boxes or multi-sound pedal boards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between the guitar and the console. They can also be found as presets in guitar amplifiers, or even as digital plug-ins within a DAW.*

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#### Haas Effect

*(Also called Precedence Effect) Simply stated, a factor in human hearing in which we perceive the source of a sound by its timing rather than its sound level. In his research, Helmut Haas determined that the first sound waves to reach our ears help our brains determine where the sound is coming from, rather than its reflection or reproduction from another source. The reflection of the sound must be at least 10dB louder than the original source, or delayed by more than 30ms (where we can perceive it as an echo), before it affects our perception of the direction of the sound. This is what helps us distinguish the original sound source without being confused by reflections and reverberations off of nearby surfaces. Understanding the Haas effect is particularly useful in live audio settings, especially in large venues where loudspeakers are time-delayed to match the initial sound waves coming from the source.*

#### Half Step

*A change in pitch equivalent to adjacent keys on a piano. Also known as a “semitone.”*

#### Half-Normalled

*Describes the configuration within a patch bay in which the jacks form a connected pathway until a patch cord is inserted to change the path. When a patch bay is “full-normalled,” the connection is altered by inserting a cord into either the input or output side; when it is “half-normalled,” the path changes only when a cord is plugged into the input. “Non-normalled” or “open” means there are no internal connections, and each input sends the signal through its corresponding output.*

#### Half-Wave Rectifier

*A half-wave rectifier passes only positive voltages, and replaces anything negative with 0v. In other words, anything “below zero” is clipped off.*

### Hall Program

*A setting of a digital delay/reverb effects unit that approximates concert halls. Hall programs are characterized by pre-delay of up to 25 ms.*

### Hard Knee

*In compression, refers to a more abrupt introduction of compression of the signal once the sound level crosses the threshold. (See also "Knee.")*

### Hard Sync

*This is the most common type of oscillator sync where the slave oscillator will reset its waveform whenever it receives a sync pulse. If the type of sync is not specified, then it's probably hard sync.*

### Harmonic Distortion

*The presence of harmonics in the output signal of a device which were not present in the input signal, usually for the purpose of changing the instrument's timbre.*

### Harmonic

*A single harmonic is the purest sound possible: It contains no overtones or other identifying characteristics aside from its pitch and loudness. The shape of its vibration – whether it be vibrating the air so you can hear it, or causing the electrical vibrations of a voltage going up and down – is a sine wave. Most of the time, overtones have a very specific pitch relationship to each other. The first or lowest harmonic – known as the 'fundamental' – is the pitch of the sound, just as the lowest note of a chord is its 'root.' The other harmonics are higher, and spaced out as integer multiples of the fundamental: two times its frequency, three times, four times, and so forth. The first few harmonics happen to have a nice musical spacing: an octave; an octave and a fifth; two octaves. But the higher they get, the less musical they may seem.*

### Harmonics

*Whole number multiples of the fundamental frequency that occur naturally within the playing of a tone. Mathematically, if the fundamental frequency is  $x$ , the harmonics would be  $2x$ ,  $3x$ ,  $4x$ , etc. For example, if the fundamental frequency of the note played is 440Hz (or A-440), the harmonics would be 880Hz, 1320Hz, 1760Hz, and so on. The presence of harmonics in the tone is what creates the timbre of an instrument or voice.*

### Head

*In tape recording, an electromagnetic transducer that magnetically affects the tape passing over it. Recording/playback heads change the audio signal from electrical energy to magnetic energy and back, for recording and playback purposes. An erase head creates a powerful electromagnetic field to the tape to erase previous signals from the tape.*

### Headroom

*The difference in dB between normal operating level and clipping level in an amplifier or audio device. Also describes the difference in dB between the peak levels of a recording and the point at which the signal distorts. (Also called "Margin.")*

### Hertz/Volt

*A system where a change of 1 volt at the input results in a change in pitch of a fixed number of hertz (cycles per second), rather than a fixed musical interval.*

## Hertz

*(Abbreviated Hz)*

1) *The unit of measurement for frequency, specifically, the number of complete wave cycles that occur in a second (cycles per second). 1 Hz = 1 complete wave per second.*

2) *A popular rental car company (not typically used in recording except for transport to the studio).*

## Hi-Hat

*In drum sets, double cymbal on a stand, usually positioned next to the snare, which can be played with a foot pedal and/or by the top cymbal being hit with a stick.*

## Hi-Z

*(abbreviated Hi-Z) Described as an impedance or resistance of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more ohms. (See also “Impedance.”)*

## High (gate)

*When a gate signal is at the voltage level (typically 5 volts, although it can be more) that indicates it is “on” – such as when a note is being held down on a keyboard controller – it is said that the gate is high.*

## High Impedance

*(abbreviated Hi-Z) Described as an impedance or resistance of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more ohms. (See also “Impedance.”)*

## High Pass Filter

*An audio filter that attenuates signals below a certain frequency (the cut-off frequency) and passes signals with frequencies that are higher.*

## High-End

*Highs or High-End – Short for “high frequencies,” loosely the frequencies above 4000 Hz. Usually meant in the context of “highs, mids and lows” in an audio signal.*

## High-Pass Filter

*The high pass filter (HPF) design passes harmonics above its cutoff or corner frequency untouched, and reduces the level of lower harmonics depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) high pass filter, harmonics one octave below the cutoff frequency (in other words, one half the cutoff frequency) are reduced in level by 12 dB; harmonics two octaves below the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pass filters are typically used to create bright sounds where the higher harmonics are much stronger than the fundamental and lower harmonics – for example, the sound of a harpsichord.*

## Horizontal Pitch

*HP = Horizontal Pitch. In the Eurorack format for synthesizer modules, the width of a module is defined as the number of hp (horizontal pitch) units. Each hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, although some are odd numbers. Also, modules tend to be ever so slightly less than exactly some multiple of 0.2” wide, just to make sure you don’t run into problems with ever so slightly too wide modules overlapping.*

## Horn

- 1) *A speaker or speaker enclosure where sound waves are sent by a speaker cone or driver into a narrow opening which flares out to a larger opening.*
- 2) *One of several different types of brass musical instruments.*

#### House Sync

*A reference signal such as SMPTE time code that is used to keep all devices in the room in sync.*

#### HP

*HP = Horizontal Pitch. In the Eurorack format for synthesizer modules, the width of a module is defined as the number of hp (horizontal pitch) units. Each hp is 0.2" (0.5 cm). Most modules are even numbers of hp wide, although some are odd numbers. Also, modules tend to be ever so slightly less than exactly some multiple of 0.2" wide, just to make sure you don't run into problems with ever so slightly too wide modules overlapping.*

#### HPF

*The high pass filter (HPF) design passes harmonics above its cutoff or corner frequency untouched, and reduces the level of lower harmonics depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) high pass filter, harmonics one octave below the cutoff frequency (in other words, one half the cutoff frequency) are reduced in level by 12 dB; harmonics two octaves below the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pass filters are typically used to create bright sounds where the higher harmonics are much stronger than the fundamental and lower harmonics – for example, the sound of a harpsichord.*

#### Hum

- 1) *The low-frequency pitch that occurs when power line current is accidentally induced or fed into electronic equipment. The hum reflects the fundamental frequency of the current (60 Hz in the U.S., and 50 Hz in many European countries).*
- 2) *To vocalize a pitch without opening one's mouth.*

#### Hybrid Power Supply

*A hybrid power supply uses a lower weight, more efficient switching power supply to perform most of the drop in voltage – say, from 120v AC to 15v DC – and then uses a linear power supply for the remaining much smaller drop, such as from 15v to 12v. These are becoming the preferred design in many modular synthesizer enclosures. Shortcomings with the power supply – too noisy, or not enough – tend to be at the cause of many unexpected problems in modular synthesizers.*

#### Hypercardioid

*A variation of the cardioid microphone pickup sensitivity pattern in which the shape of the optimal pickup area is tighter and more directional than cardioid. Hypercardioid microphones are most sensitive directly on-axis in front of the microphone, and begins rejecting sounds between 90-150 degrees off-axis, depending on the tightness of the pattern.*

#### Hz/V

*A system where a change of 1 volt at the input results in a change in pitch of a fixed number of hertz (cycles per second), rather than a fixed musical interval.*

#### Hz

*An abbreviation for the term Hertz, or the unit of frequency.*

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### IADSR

*This is an Attack/Decay/Sustain/Release (ADSR) envelope generator that allows you to start the attack phase at an initial level – the “T” – rather than the customary 0 volts. The envelopes in the Prophet VS, as well as a module from Ladik, have this capability.*

### IC

*Integrated Circuit – A miniature circuit of many components set on semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”*

### Imaging

*Refers to the ability to localize a specific sound within the sound space. In recording environment, it refers to “placing” instruments within the stereo or surround field so that it when the sound is played through speakers, it fools our ears into thinking the sound source is emanating from a specific point instead of from the speakers. In live audio and sound reinforcement, the principle of imaging is the same, the goal being to make the audience perceive the sounds as coming from performers on the stage, rather than from the speakers.*

### Impedance

*Refers to the resistance of a circuit or device to alternating current, which can be mathematically described as the ratio of voltage to current. Differences in impedance between devices in the studio can affect how they work together. Impedance is abbreviated by the letter Z, and measured in ohms (W).*

### In Line Console

*An audio mixing console that is designed and configured so each channel strip can be used for both recording and monitoring functions during multitrack recording. This configuration is in contrast to split mixing consoles, which requires separate channels on the board for recording and monitoring functions.*

### In Phase

*The desirable situation in which two or more devices (and their respective audio signals) are on the same side of the polarity spectrum, producing waveforms that do not conflict or cancel each other out.*

### In Port

*A jack on a MIDI device or computer that will accept an incoming data signal.*

### Inductance

*A characteristic of electrical conductors in which electrical charge (voltage) is produced or stored magnetically due to the natural resistance to change in the electrical current. Inductance is an electromagnetic principle that can either assist in audio applications (as in loudspeakers) or cause resistance (as in using speaker wire whose gauge is too low for the application).*

### Inductor

*A device (usually a coil of wire) that converts electrical energy into stored magnetic energy as*

*electrical current passes through it. Commonly found in a variety of audio applications such as guitar pickups and loudspeakers.*

#### Infinite Baffle

*A loudspeaker mount or enclosure designed so that sound waves coming from the front theoretically do not reach the back, preventing the sound waves from cancelling each other out. The term “infinite” comes from the idea that mounting the speaker on a wall with no end points would not allow sound waves to migrate behind it. Of course, this is physically impossible, so infinite baffles are designed to replicate this as much as possible. Examples of infinite baffles are mounting the speaker on a wall of an enclosed room, or building it inside a sealed cabinet large enough to prevent rear sounds from affecting the cone from the back.*

#### Initial/Attack/Decay/Sustain/Release

*This is an Attack/Decay/Sustain/Release (ADSR) envelope generator that allows you to start the attack phase at an initial level – the “I” – rather than the customary 0 volts. The envelopes in the Prophet VS, as well as a module from Ladik, have this capability.*

#### Input / Output (I/O)

*I/O – An abbreviation for “Input/Output.” In audio, it refers to any device, program or system involving the transferring of electrical/audio signals or data.*

#### Input Impedance

*The opposition to current flow by the first circuits of a device.*

#### Input Monitoring

*A setting on many DAWs that allows you to monitor the live input signal coming into the DAW (as opposed to the recorded signal).*

#### Input

*The jack or physical location where a device receives a signal. Also refers to the incoming signal itself.*

#### Insert

*An access in the signal chain (usually in the mixing console or virtually within a DAW) in which a device, signal processor or digital plug-in can be “inserted” into the circuit between pre-amplification and the channel or bus output. Commonly used to add processing such as reverb, compression or EQ to a channel or group of channels.*

#### Instrument Amplifier

*A device that has a power amplifier and speaker to reproduce the signal put out by an electric instrument.*

#### Instrument Out Direct

*Feeding the output of an electric instrument (like an electric guitar) directly to the recording console or tape recorder, as opposed to miking the amplifier.*

#### Insulator

*A substance such as glass, air, plastic, etc., that will (for all practical purposes) not conduct electricity.*

#### Integrated Circuit

*Integrated Circuit (Abbreviated “IC”) – A miniature circuit of many components set on semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”*

#### Integrator

*This function smoothens out an incoming signal so that the change in voltage level. “Integrator” is the technical name for this math function; you are more likely to see this module called a slew limiter (where I go into more detail on its uses) or less often as a lag generator or processor.*

#### Interface

*Any device or connection point that allows one unit to work, drive or communicate with another unit, or that allows a human to interact with a computer or other electronics. There are many examples of interfaces in professional audio situations, including MIDI (Musical Instrument Digital Interface); audio interfaces which connect audio inputs to your computer; and even your DAW program, which displays a screen that enables you to assign instruments, adjust settings, record, mix and playback. Even the mixing console is an interface of sorts, connecting the many elements of the control room.*

#### Intermodulation (IM) Distortion

*Distortion caused by two or more audio signals of different frequencies interacting with one another. The sum and difference of the frequencies produce new (usually unwanted frequencies) that didn’t exist in any of the original frequencies.*

#### Inverse Square Law

*A mathematical rule that describes an inverse relationship between one quantity and the square of another quantity. In plain English, one number goes down by a certain amount each time the other number doubles. In audio and acoustics, the inverse square law says that in an open sound field with no obstructions, the sound pressure level will drop by half (6dB) each time the distance from the sound source is doubled. (This equation is quite useful to audio engineers trying to provide sound in open-air settings, for example.)*

#### Inverter

*An inverter multiplies an incoming control voltage by  $-1$ . In the case of a gate or logic inverter, it reverses the high and low states so that (for example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a polarizer, as it changes the polarity (+ versus  $-$ ) of a signal. A control voltage inverter is often combined with an offset voltage to adjust the output voltage into the desired range. For example, if you had an envelope generator that had an output range of 0 to +8 volts, and you just inverted it, the result would be 0 to  $-8$  volts. Since some modules such as voltage controlled amplifiers usually expect only positive voltages, you would then need to add 8 volts to that result to get an upside-down (inverted) envelope that still had an overall range of 0 to +8v.*

#### Inverting Mixer

*Most signal mixers make an effort to keep the same polarity of a signal as it passes through the mixer. However, some mixers may invert the polarity or “phase” of a signal (as it’s a simpler design); other mixers may allow you to invert a signal on purpose so that you can experiment with tricks like adding one waveform or filter mode output out of phase with another coming from the same oscillator or filter.*

#### Isolation



*The process of containing sound within a certain area so that it doesn't interact with other sounds. For example, acoustically treated isolation booths are often used to record vocals or instruments in the studio to keep outside noises from bleeding into the recording microphone, or likewise to keep vocals or other sounds away from instrument mics during live recording sessions.*

#### IV Cable

*You often need to send one signal to multiple destinations. Options for doing this include using dedicated multiples, free-floating widgets with multiple jacks wired together, or fancy cables that allow you plug one or two extra cables into them. The IV cable is one the latter: Made by Erthenvar, it has an extra 3.5mm jack molded into the mid-point of the cable (loosely resembling an intravenous or "IV" drip), in addition to having 3.5mm plugs at either end.*

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#### Jack

*That hole you plug your patch cables into on the face of your synthesizer modules? That's called a jack. The size and type of jack – 3.5mm, banana, or 1/4" – often is one of the defining features of different synth module formats: 3U/Eurorack, 4U, and 5U/MU respectively. (No, a plug is not called a jill. Actually, it's the other way around: A plug is sometimes referred to as a male connector, and a jack is referred to as a female connector.)*

#### Jam Sync

*A process available on some clock or syncing devices which reads an external time code and recreates (or "jams") a new time code identical to the original external code for the syncing of devices. This function is mainly used for replacing code that has become degraded.*

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#### Karplus Strong

*This is a physical modeling synthesis algorithm designed to replicate the sound of plucked, vibrating strings – although it has also proven useful for some percussion sounds as well. A short sample – originally noise, although it can be a high frequency chirp or other sound – is sent to both the output, and to a delay line. The output of a delay line is connected to a filter – originally a one-pole low pass filter; changing the filter has a huge effect on the character of the sound – and then back to both the main output and the input of the delay line. A few modules implement Karplus Strong synthesis, although it is an interesting challenge to patch yourself and play with the results.*

#### Key

1) *In music, the note scale in which a piece of music is written or played, identified by the first note (tonic) of the scale, as in, "Key of C."*

\*2) *The control of a dynamics processing device by an external audio signal through the use of a side chain. \**

3) *A digital or data code that unlocks the use of a device or software. Example: Pro Tools is licensed through an iLok ID via the use of a physical USB key.*

### Keyboard Controller

*A piano-styled keyboard that sends out MIDI signals to control other MIDI devices. Most keyboard instruments are equipped with MIDI control capabilities, but dedicated MIDI keyboard controllers emit no audio signals, only MIDI data.*

### Keyboard Tracking

*Most modular synths follow a strict relationship between voltage and pitch, such as 1 volt per octave; any deviation would cause tuning errors. Because of this sensitivity, 1v/oct and similar signals and connections are sometimes specifically distinguished as keyboard tracking rather than just “CV” (control voltage) to make it clear they are not attenuated or otherwise modified when controlling a function on a module.*

### Keyboard

*Any musical instrument or computer controlled by pressing a key.*

### Keytar

*Lightweight, portable keyboard meant to allow keyboardists the same freedom (not to mention posturing opportunities) as guitarists.*

### Kick Drum

*The bass drum on a trap drum set, so called because it is played with a kick pedal.*

### Kilohertz (kHz)

*kHz – An abbreviation for kilohertz (1000 Hz, or 1000 cycles per second). Example: 2000 Hz = 2 kHz. Most commonly used in the studio for describing audio frequency ranges or digital sampling rates.*

### Knee

*A function on a compressor that determines how abruptly or gradually compression begins once the sound level crosses the threshold. So-called because the graphic “bend” in the response curve is reminiscent of a knee. “Hard knee” refers to an abrupt activation of the compressor, while “soft knee” refers to a more gradual change.*

### Krell Patch

*Recreating this patch is a challenge many modular musicians like to tackle. It is based on the 1959 movie *Forbidden Planet*, in a segment where they supposedly play the music of the ancient Krell race. In general terms, each note has a random pitch, envelope, and duration.*

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### Lag Generator

*This function smoothes out an incoming signal so that the change in voltage level cannot exceed a certain number of volts per second. This causes the result to “lag behind” changes in the input. It is sometimes called a slew limiter or technically as an integrator.*

### Layering

*Refers to almost any blending of similar multiple musical parts or sounds at once, often combined on one channel or assigned to one controller. In audio recording, layering usually involves recording*

*similar takes of the same instrument or vocal (or duplicating parts with slight delays or chorusing effects) to create a fuller, richer sound than the vocal/instrument by itself. In sound design, it also refers to blending multiple samples (example: two or more drum sounds) to create a fuller sound.*

#### Lead Sheet

*A shorthand form of music notation (similar to a chord chart) that displays the basic essential elements of a song so musicians can follow along without the full notation of every note or expression. Lead sheets most commonly include a melody line written in music notation with chord changes above the staff, and lyrics below it. (See also “Chord Chart.”)*

#### Leakage

*Sounds from other instruments and sound sources that were not intended to be picked up by the microphone.*

#### Level

*The amount of signal strength; the amplitude, especially the average amplitude.*

#### LFO

*This module produces repetitive, cycling waves ranging in frequency from the low end of the audio spectrum to as slow as many seconds or even minutes per cycle. They are used to produce effects such as tremolo (when controlling the loudness of a signal), vibrato (when controlling the pitch of a signal), repetitive filter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an oscillator, and more.*

#### Limiter

*A type of compressor that sharply reduces (limits) the gain of the signal when the audio level reaches a certain threshold, typically used to prevent overload and signal peaking. A compressor effectively becomes a limiter when its ratio is 10:1 or higher. (See also “Compressor.”)*

#### Line Input

*Line Input (“Line In”) – An input designed to take a line level signal.*

#### Line Level

*Most consumer and lower-cost professional audio equipment use a signal level reference known as line level or  $-10\text{dBV}$  (decibel volts). The most common connectors are RCA (phono) or 3.5mm, although  $1/4"$  is also used; the signal is “unbalanced” (it uses two wires: signal and ground). In the line level standard, a sine wave that varies between  $\pm 0.447$  volts is considered to be at  $-10\text{dBV}$ . By contrast, a typical oscillator signal in a modular synthesizer is  $\pm 5$  to  $\pm 8$  volts. As a result, you will need either an output module in your modular synth or one heckuva input attenuator on your mixer or recorder to plug your synth into equipment that runs at line level. Similarly, you will need to substantially boost a line level signal to get it up to modular standards to process in your modular synth.*

#### Line Output

*Line Output (“Line Out”) – Any output that sends out a line level signal, such as the output of a console that feeds a recorder.*

#### Linear FM

*This is often the preferred input response for frequency modulating (FM’ing) an oscillator, as the*

*result stays in tune while you change the modulator.*

#### Linear Power Supply

*A linear power supply design takes a higher incoming voltage and reduces it to a lower voltage using components such as transformers. In very general terms, they tend to introduce less noise into the output power signal, at the cost of increased heat and weight (they're not very efficient). Many are moving to a hybrid power supply that combines a switcher with a small linear supply or regulator to get the best of both worlds.*

#### Linear VCA

*A linear voltage-controlled amplifier (VCA) uses a simple mathematical relationship between control voltage input and signal level output – for example, 50% of nominal control voltage in would result in the output signal being at 50% of the level of the input signal. This, however, is not how our ears perceive loudness; a sound must be amplified by 10x in order to be perceived as twice as loud. This makes a linear VCA desirable for scaling control voltages, but perhaps less so for scaling audio signals. If you connect an envelope generator with an exponential output to a linear VCA, then you will get the desired aural result. Confusing? That's why it's great when an envelope generator or VCA has a switch or control to vary it between linear and exponential response. A linear mixer is similar to a linear VCA: "half" on the input level control equals the output having half the voltage swing as the input. Again, this is fine for altering control voltages, but not for mixing audio signals; in that case you want a mixer with exponential controls.*

#### Linear VCO

*A linear voltage-controlled oscillator (VCO) follows the volts/hertz (v/Hz) standard; more common is the exponential volts/octave (v/oct) standard. The exception is frequency modulation (FM), where a linear control voltage input is often preferred to recreate classic style FM as it does not change the fundamental pitch of the carrier oscillator.*

#### Live Recording

*A recording session where all the musicians are playing at once with no overdubbing.*

#### Live Room

*The large, main room of the recording studio where most of the instruments and/or vocalists perform. So called, not just because there is room for live performances, but because the room has been acoustically treated to produce a pleasing amount of live reverberation.*

#### Live

- 1) A term describing a space with a reverberant or reflected sound. In a "live" space, the sound waves are active or "live."*
- 2) Occurring in real time, as opposed to previously recorded.*

#### Local On/Off

*Local On/Off – A MIDI message that controls the internal sound module of a synthesizer or MIDI controller. "Local On" triggers the internal module when the keyboard is played; "Local Off" disconnects it. "Local Off" is frequently used to prevent unwanted looping of MIDI messages in some configurations, or when controlling the internal module via another controller.*

## Logic Functions

*In a modular synth, control voltages tend to be continuous in nature, while gate and trigger signals are binary: on or off; high or low. This is the same as logic signals in digital circuitry. Therefore, some make digital logic modules. A common logic function is OR: If either signal A or signal B is high (on), then output a high gate signal (on); otherwise output a low gate (off). Another is AND: If and only if signal A and signal B are both, then output a high gate (on); otherwise, output a low gate (off). These are great functions for combining beat triggers from different timing sources.*

## Logic

*Binary or Boolean logic is a way of combining gate signals (on or off voltages) to create new outputs. Each section of a logic module typically includes 1 to 3 inputs, with 2 being the most common. An OR function says if there is a gate on (or “high”) signal at any of the inputs (i.e. input 1 or input 2 or input 3, etc.), to output a gate on signal. An AND function says only output a gate on signal if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all have gate ons). Adding an “N” to the front of a function’s name says “not” this function – in other words, a NOR function would only output a high signal if all inputs were low (not input 1 nor input 2 are high).*

## Loop

*1) Effectively, any piece of music or data that repeats endlessly. Before digital audio and sampling, loops were created by looping tape. Today, loops are used in samples to sustain a sampled note for as long as the note is triggered, while drum loops and other music loops are common in modern music production.*

*2) Another term for antinode, or the points of maximum displacement of motion in a vibrating stretched string or a sound wave. (See also “Standing Wave.”)*

## Looping

*Sometimes it’s useful to have a module loop or repeat its functions. For example, an envelope generator that can be set to loop becomes a low frequency oscillator: as it attacks to a maximum value and decays back to zero, it starts that attack phase again. Quite often you want a note sequencer to loop: When it reaches the last note in the sequence, it would be useful for it to then look back to or return to the first note and start over. Audio recorders with looping features are also popular for live performance.*

## Loudness

*A term referring to how the human ear perceives incoming sound waves. This term seems self-explanatory, but it’s deceptive. We commonly think of loudness as it relates to the volume of a sound, but this is an indirect relationship. In acoustic terms, volume is more about the amplitude of the sound waves, while loudness describes how our ears hear the intensity of those waves.*

## Low (gate)

*Most often, this is shorthand for saying a gate or trigger signal is in its “off” condition (typically 0 or –5 volts, in contrast to a “high” or “on” signal of +5 volts).*

## Low Frequency Oscillator

*This module produces repetitive, cycling waves ranging in frequency from the low end of the audio spectrum to as slow as many seconds or even minutes per cycle. They are used to produce effects*

*such as tremolo (when controlling the loudness of a signal), vibrato (when controlling the pitch of a signal), repetitive filter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an oscillator, and more.*

#### Low Impedance

*(abbreviated Lo-Z) Described as impedance of 500 ohms or less. (See also “Impedance.”)*

#### Low Pass Filter

*The low pass filter (LPF) design passes harmonics below its cutoff or corner frequency untouched, and reduces the level of lower harmonics depending on how far above the cutoff they are. In a 12dB/oct (decibel/octave) low pass filter, harmonics one octave above the cutoff frequency (in other words, double cutoff frequency) are reduced in level by 12 dB; harmonics two octaves above the cutoff (four times the frequency) are reduced by 24dB, and so forth. This is the most common type of filter used, as most natural sounds have stronger low harmonics and weaker high harmonics – especially as a note fades to silence.*

#### Low Pass Gate

*By strict definition, a low pass gate (LPG) is a low pass filter whose cutoff frequency goes down into the subsonic range as its control voltage goes towards 0 volts, resulting in the input signal being filtered almost into silence. Some replicate this by combining a low pass filter and a voltage controlled amplifier into the same module, with both following the same control voltage. In either case, as an input envelope falls from a high level to 0 volts, the output gets duller (higher harmonics are filtered more) as it falls to silence. This mimics the way many natural sounds work.*

#### Low-Frequency Oscillator (LFO)

*A circuit that emits low-frequency electronic waveforms below the audible level of human hearing (20 Hz or less). This low-frequency waveform creates a rhythmic pulse that is used to modulate various parameters in the audio signal, such as pitch or volume. LFOs are frequently used in samplers, synthesizers and signal processors to create such effects as vibrato, tremolo, and phasing.*

#### low-pass-filter

*An audio filter or device that attenuates signals above a certain frequency (the cut-off frequency) and passes signals with frequencies that are lower than the cut-off.*

#### Lows or Low-End

*Short for “low frequencies,” loosely referring to bass-frequency signals below 250 Hz. Usually meant in the context of “highs, mids and lows” in an audio signal.*

#### LPF

*The low pass filter (LPF) design passes harmonics below its cutoff or corner frequency untouched, and reduces the level of lower harmonics depending on how far above the cutoff they are. In a 12dB/oct (decibel/octave) low pass filter, harmonics one octave above the cutoff frequency (in other words, double cutoff frequency) are reduced in level by 12 dB; harmonics two octaves above the cutoff (four times the frequency) are reduced by 24dB, and so forth. This is the most common type of filter used, as most natural sounds have stronger low harmonics and weaker high harmonics – especially as a note fades to silence.*

## LPG

*By strict definition, a low pass gate (LPG) is a low pass filter whose cutoff frequency goes down into the subsonic range as its control voltage goes towards 0 volts, resulting in the input signal being filtered almost into silence. Some replicate this by combining a low pass filter and a voltage controlled amplifier into the same module, with both following the same control voltage. In either case, as an input envelope falls from a high level to 0 volts, the output gets duller (higher harmonics are filtered more) as it falls to silence. This mimics the way many natural sounds work.*

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## M2.5

*A common screw thread size used to mount Eurorack modules. This size is most common when using a system of loose nuts that slide along the rails that the modules are attached to.*

## M3

*A common screw thread size used to mount Eurorack modules. This size is most common when using module mounting rails that have been pre-drilled.*

## Magnetic Tape

*Recording tape consisting of a plastic strip coated by magnetic materials, finely ground iron oxide (rust) particles. Commonly used for analog recording.*

## Magnetism

*A natural attractive energy of iron based-materials toward other iron-based materials.*

## MARF

*The rare Buchla Model 248 MARF (Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelope generator (both described elsewhere in this glossary) in that it typically contains 16 or 32 stages (sometimes referred to as “segments”), and a rate control to interpolate between these stages. This means very complex envelope shapes and other control voltage sequences can be created. Later on, Buchla used the term MARF to describe the multi-step envelopes in instruments such as the Buchla 400.*

## Margin

*See “Headroom.”*

## Masking

*The characteristic of hearing by which loud sounds prevent the ear from hearing softer sounds of similar frequency. Also refers to the obscuring of softer sounds by louder ones.*

## Master

- 1) The main output control of a console or DAW, setting the level of the mixed signal as it leaves the console. (Also called “master fader.”)*
- 2) The final-mixed original recording from which copies are made.*

## Mastering

*The final process of fine-tuning and “sweetening” the mix on a song or collection of songs, from which the master will be created.*

### Measure

*The grouping of a number of beats in music. (See also "Bar.")*

### Meg

*A slang abbreviation based on the prefix "Mega-, meaning 1,000,000. Often used as shorthand for megahertz (1,000,000 Hertz, Mhz) or megabytes (1,000,000 bytes, MB).*

### Meter

*1) A device that measures and displays the signal level in audio or digital equipment. Meters usually measure peak values or RMS values. (See also "Peak Value," "RMS Value.")*

*2) The rhythmic structure of music, typically describing the number of beats in a measure.*

### Mic / Line Switch

*Mic, Mike – Abbreviations for "microphone."*

### Microphone (Mic) Input

*The input of a console or other device designated for a microphone signal.*

### Microphone (Mic) Level

*The very low audio voltage level emitted by a studio microphone. The signal must go through a preamplifier to be increased to line level before entering the console. (See also "Line Level," "Preamplifier.")*

### Microphone (Mic) Pad

*A setting on a microphone or preamp, or a separate adapter/connector, that reduces the level of the microphone signal before it enters the preamplifier to prevent overload.*

### Microphone

*A transducer which converts sound pressure waves into electrical signals.*

### Mid-Side Miking (M/S)

*(Abbreviated M/S) A stereo coincident microphone placement technique in which one cardioid pattern microphone is aimed directly at the sound source, and a bi-directional microphone placed sideways and as close as possible to the first mic.*

### MIDI Clock

*A clock signal conveyed by MIDI that is used by the connected sequencers and musical devices to stay in sync with one another. Not to be confused with MIDI time code (MTC), MIDI clock is tied to the Beats-Per-Minute (BPM) tempo, advancing 24 steps per quarter note.*

### MIDI Controller

*\*Can refer to two different elements of MIDI, depending on the context. \**

*1) A device or software that sends MIDI data to connected devices, either through pre-programmed sequencing or through live performance by a musician.*

*2) Any of a number of smaller controls on a MIDI device that is assigned to control specific parameters of the sound or performance.*

### MIDI Interface

*A device that converts a MIDI signal into the digital format of a computer so it can store and use the MIDI signal.*



### MIDI over Bluetooth

*Bluetooth Low Energy (BLE) is a wireless connection specification supported by the majority of mobile computing devices. BLE (also called Bluetooth SMART) can extend battery life for mobile devices using connected accessories (such as MIDI keyboards and controllers) that don't continuously stream data.*

*An MMA Working Group evaluated Bluetooth LE MIDI performance (latency and jitter) and decided on a specification for MIDI over Bluetooth which would enable products from different manufacturers to interoperate. The Specification for MIDI over Bluetooth Low Energy (BLE-MIDI) is based on Apple's implementation which appeared in iOS8 and OSX 10.10, so that products from early adopters would remain compatible with the industry standard.*

### MIDI Sample Dump Standard (SDS)

*A sub-protocol that was added into MIDI to enable the transfer of digitally recorded samples between instruments, storage units or sound modules without converting them to analog.*

### MIDI Sequencer

*A device or software that can record and play back MIDI data, controlling the performance of MIDI musical instruments or devices in a series of timed steps. MIDI sequencers can exist on board MIDI controllers, keyboards or workstations, as standalone devices, or as computer software.*

### MIDI Thru Box

*A unit with one MIDI In Port and several MIDI Thru Ports to relay the MIDI signal to multiple devices. MIDI users often prefer this as an alternative to "daisy chaining" devices, which can cause slight delays in the MIDI signal.*

### MIDI Thru

*A port that puts out a MIDI signal that is the same as the incoming MIDI signal, effectively relaying the signal to another device without altering or changing it. (Many MIDI devices have three MIDI ports: In, Out and Thru.)*

### MIDI Time Code (MTC)

*The translation of the information in SMPTE time code into MIDI data, enabling MIDI sequencers and connected devices to sync with SMPTE code (usually in relation to video). (See also "SMPTE Time Code.")*

### MIDI

*Short for Musical Instrument Digital Interface. MIDI is a common language to connect one synthesizer to another, and synthesizers to a computer. Although it is a digital language, it is easy to buy a MIDI to CV/Gate (control voltage and gate) converter module that handles both note events and MIDI clocks for driving sequencers and the such. The biggest thing to watch out for is what type of connector is required: the traditional 5-pin DIN, or a USB computer-style connection.*

### Mids

*Abbreviation for "mid-range frequencies," the audio frequencies from about 250 Hz through 6000 Hz. Meant in the context of "highs, mids and lows" in an audio signal.*

### Mini Keys

*A number of keyboard controllers and even keyboard synths use a key size that is much smaller than*

*a typical piano key. Mini keys is the term commonly used (sometimes derisively, although the space and cost savings can be quite significant) to refer to this hardware choice.*

#### Mix Down

*Mixdown or Mix Down – The processes of creating a final mix by combining multiple audio tracks into a single track (or two-channel stereo track) prior to the mastering stage. This can include the traditional method of mixing the multiple channels of analog tape into a two-track master, or the more modern method of creating a digital mixdown using a DAW (which is often referred to as “rendering”).*

#### Mix

- 1) The blending of audio signals together into one composite signal.*
- 2) Can also refer to the blending of a portion of an effected audio signal back into the direct signal.*

#### Mixer

*This module combines signals together. You may use a mixer to combine audio signals, in which case you may want one with exponential level controls and perhaps stereo panning, or to combine control voltages, in which case you may want linear level controls plus additional functions to invert and offset the voltages going through it.*

#### Modular

*A modular synth breaks down the main components of a synthesizer – the tone-generating oscillators, the tone-modifying filters, the amplitude-shaping VCAs, and the modulation sources that create envelopes, tremolos, and more – into individual modules you can purchase and install. At the most basic level, this allows you to play mix-and-match in building your own custom synth.*

#### Modulation Noise

*Noise that is present only when the audio signal is present.*

#### Modulation

*When you vary a parameter of a synthesizer module using voltage control, it is said that you’re modulating that parameter. For example, when a low frequency oscillator (LFO) varies the cutoff frequency of a filter to create a wah-wah effect, it is said that the LFO is modulating the cutoff. When an envelope generator causes a voltage controlled amplifier (VCA) to open up to allow a sound to become suddenly loud, and then fades it back down to silence, you can also say the envelope is modulating the amp (although some like to restrict the term “modulate” to a repetitive action). Therefore, we call the sources of these changes modulators.*

#### Modulator

*We touched on the general subject of modulation and modulators in the definition above. However, quite often when someone uses the term modulator, they’re usually discussing a synthesis techniques where one usually audio-rate signal “modulates” (varies) another audio signal. For example, in frequency modulation (FM) synthesis, the modulator (or modulating oscillator) varies the frequency (pitch) of the main signal generator (oscillator), called the carrier. In ring, balanced, or amplitude modulation, the modulator is varying the loudness of the carrier signal. So the term modulator is a way to make it clear which component you’re talking about in one of these patches: not the main tone generator, but the module that is driving that generator crazy.*

## Module

*A self-contained group of circuits and controls. In the recording studio, modules are often contained in interchangeable housing for installation on rack mounts, and can include amplifiers, equalizers, effects processors and sound modules (MIDI instruments to be activated by an external controller). In the digital space, plug-ins, software synths, samplers and plug-ins are also described as modules.*

## Monaural (Mono)

*(Abbreviated “Mono”) Describing an audio signal coming through a single, as opposed to stereo, which is two channels. (See also “Monophonic.”)*

## Monitor Mix

*A mix of the live and/or recorded audio signals that is fed to the musicians so they can hear the music while performing, whether live onstage or in the studio. Monitor mixes are on a separate signal path from the main mix (often controlled by a separate, smaller console) and do not affect the FOH mix (in live audio) or the signal going into the multitrack recorder/DAW. In live performance settings, the monitor mix is often controlled by a separate audio engineer running a separate sound board.*

## Monitor Mixer Section

*Monitor Section/Monitor Mixer Section – The section of the console that is used to create a rough mix so the engineer can hear what is being recorded without affecting the levels being fed to the multitrack recorder or DAW.*

## Monitor Path

*A signal path separate from the channel path that allows the engineer to listen to what is being recorded without affecting the signal being fed to the multitrack recorder or DAW. (See also “Channel Path.”)*

## Monitor

- 1) To listen to the music for the purpose of checking quality or avoiding peaks.*
- 2) A speaker in the studio (usually one of a pair) that is used to listen to the audio signals. This can include studio monitors in the control room for listening to the mix, and headphones in the booths or live room for the performers to hear a mix of the tracks while they are performing.*

## Monophonic

*(Abbreviated “Mono”)*

- 1) A single sound source or single-channel transmission (as opposed to stereo).*
- 2) A melody line in which only one note at a time is played.*
- 3) Describing an instrument or synthesizer setting that only plays one pitch (or “voice”) at a time. (See also “Voice.”)*

## Morphing

*In the context of a modular synth, morphing refers to an oscillator that can more or less smoothly change the shape of its output waveform – and therefore, the resulting sound – as you play it. This is usually the domain of digital oscillators which internally crossfade (or in some cases, switch) from one waveshape to another, although it is sometimes applied to analog oscillators that give you real time control over waveshapes.*

### Mother-32

*A very popular semi-modular synthesizer by Moog. It comes in its own case, but can be mounted in a Eurorack-format case. It comes with one VCO (sawtooth and pulse waveforms), one LFO (triangle and square waveforms), one Moog-style transistor ladder filter that can be low pass or high pass, and one AD or AR envelope generator. It also has a very capable step sequencer plus a miniature one-octave keyboard. What makes it a semi-modular is a nice patch panel that allows alternate routings for the way the synth voice is internally wired, and for it to be patched to external modules. As so many of these were sold, I'm using it as a representative of a typical semi-modular or "starter" synthesizer voice when discussing how to expand a basic modular system. I have an online introductory course to the Mother-32 coming out this spring, and will have a course plus ongoing weekly series on adding different modules to this starter system.*

### Moving Coil Microphone

*A microphone in which sound pressure waves are converted to an electrical audio signal by an induction coil moving within a magnetic field—a process often compared to a loudspeaker working in reverse. Dynamic microphones are less sensitive than condenser microphones, but can be effective for miking louder sound sources or for close-miking applications.*

### Moving Fader Automation

*A feature in some consoles in which fader changes can be pre-programmed to occur automatically during playback of a multitrack recording.*

### MU

*Refers to modules that are 5U (rack units) or 8.75" (22.2 cm) high, which is most often associated with the vintage Moog standard and those who have followed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You will sometimes hear this used interchangeably with MU for Moog Units, which also refers to a standardized width of 2.125" (5.4 cm) wide per MU. Given that this standard is both historical and physically large, some users "5U" as a badge of honor that they're traditional and cool. (And they are.) There was also a briefly popular 5U format from MOTM that used a different width and power connection. It has since been discontinued, but there are still diehard MOTM format users today.*

### Multi-Tap Delay

*A delay works by in essence putting audio in one end of a pipe and grabbing it again when it comes out the other. A multi-tap delay says "Why wait until the audio snapshots go all the way through the pipe? Let's grab it when it's only part way through the pipe." Those points where it's prematurely grabbed are the "taps" – kind of like additional water taps added along a long pipe.*

### Multimeter

*A small device that tests electrical voltage, current, and resistance. Multimeters are useful in recording studios for calibrating electrical systems and troubleshooting problems.*

### Multiple Arbitrary Function Generator

*The rare Buchla Model 248 MArF (Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelope generator (both described elsewhere in this glossary) in that it typically contains 16 or 32 stages (sometimes referred to as "segments"), and a rate control to interpolate*

*between these stages. This means very complex envelope shapes and other control voltage sequences can be created. Later on, Buchla used the term MARF to describe the multi-step envelopes in instruments such as the Buchla 400.*

#### Multiple

*Quite often you need to split or copy a signal to send to more than one destination. This is commonly done with a multiple (“mult” for short) where you plug one source in, and then plug in additional patch cables to go off to multiple destinations.*

#### Multiplexer

*Multiplexing is a technical way to describe signal routing, where multiple signals may be routed to one destination. In synth modules, this is usually extended to include the possibility of one input being switched between multiple outputs. A sequential switch is a type of multiplexer, as it chooses among multiple inputs to decide which one to send to the output (or the other way around). There are some modules that do this at audio rate, using an oscillator’s output to switch between variations of another waveshape to create complex, chopped mixtures of those waveforms.*

#### Multitimbral

*Refers to the ability of a synthesizer or module to play several different sounds, patches or “timbres” at once.*

#### Multitrack Recording

*Also called tracking or multitracking) The heartbeat of the recording studio, multitrack recording is process of recording a collective of sound sources onto separate tracks, each with its own audio channel, then combining the tracks to play back simultaneously. Recording can be done either one track or instrument at a time (to be combined later) or by recording the performers onto separate tracks as they play together live. These signals were originally recorded onto multitrack analog tape, but today they can also be recorded digitally as separate audio files into a digital audio workstation (DAW).*

#### Multitrack Tape

*A piece/reel of magnetic tape which can be used to store two or more discrete signals in sync with each other.*

#### Musical Instrument Digital Interface (MIDI)

*Short for Musical Instrument Digital Interface. MIDI is a common language to connect one synthesizer to another, and synthesizers to a computer. Although it is a digital language, it is easy to buy a MIDI to CV/Gate (control voltage and gate) converter module that handles both note events and MIDI clocks for driving sequencers and the such. The biggest thing to watch out for is what type of connector is required: the traditional 5-pin DIN, or a USB computer-style connection.*

#### Mute Switch

*A switch on a console or other piece of audio equipment that turns off the input or output, or a matching button on the virtual audio control space of a DAW. The individual channels on a console each have a mute switch that can cut the signal for that channel.*

#### Mute

*Sometimes you need to silence or disconnect a signal. A circuit that allows you to do so is called a*

*mute.*

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#### Nanowebers per Meter (NW/m)

*The standard unit in measuring the amount of magnetic strength on analog tape. A Weber is a unit of magnetic strength, but it is too large a unit to apply to the magnetism in tape recorders, so nanowebers is used instead. Nanowebers per meter of tape effectively describes the signal strength that is being recorded to tape.*

#### Narrowband Noise

*Noise (random energy) that occurs over a limited frequency range.*

#### Near Field

*The area between 1-5 feet from the sound source. Studio monitors are generally considered “near-field” speakers because they are meant to be listened to at close range. (See also “Far Field.”)*

#### Near-Coincident Miking

*A stereo miking technique in which two microphones are placed near each other at an outward angle to create a stereo image (as opposed to “Coincident Miking” which angles the microphones toward each other). Common versions of near-coincident miking include DIN stereo (90-degree angle, 20cm apart), NOS stereo (90-degree angle, 30 cm apart) and ORTF (110-degree angle, 17 cm apart).*

#### Negative Feedback

*A portion of the output signal that is fed back to the input of an amplifier with its phase inverted from the original output signal. This has a dampening effect on the output, effectively cancelling out a portion of the volume.*

#### Noise Floor

*The level of the noise present below the audio signal, measured in dB. Every electronic device emits a minimum level of noise, even when no audio is traveling through it; this is described as its noise floor. Generally speaking, the lower the noise floor in these devices, the higher the quality of the device. The noise floor also translates to the recorded signal; the noise floor of a recording is the sum of all the noise generated by connected devices. The objective is always to keep the noise floor as low as possible.*

#### Noise Gate

*A gate that is used reduce audible noise by automatically turning off an audio channel when the signal is not present.*

#### Noise Reduction

*Any of a number of processes to remove noise from a signal, device or system.*

#### Noise

*Describes any unpleasant, objectionable or unintended sound frequencies present in the audio signal. All electronic equipment produces some type of noise, which may be described as a hiss or buzz that can be heard during quiet or otherwise silent passages. (See also “Noise Floor.”) Bad connections,*

*improper grounding, radio interference and other issues can also cause introduce noise into the signal. Engineers may also deliberately run a noise signal through a sound system for testing purposes. (See also “White Noise, “Pink Noise.”)*

#### Non-destructive Editing

*A feature in recording systems (most common in Digital Audio Workstations, or DAWs) in which the original signal or content stays intact while edits are performed, allowing the engineer to revert to the original version at any time. (Sometimes also called “Nonlinear editing.”)*

#### Nondirectional

*In microphones, picking up evenly from all directions.*

#### Normalize

*To apply a fixed amount of gain to audio so that the highest peak is set at the highest acceptable recording level.*

#### Normalled

*The power of modular synthesizers is that you can patch a signal to flow the way you prefer through your system. This can also be a time-consuming bummer when you’re just trying to patch a “typical” signal flow. Therefore, some manufacturers have created “semi-modular” synths that have all of these typical connections pre-wired for you, with the important feature that many of these wirings can be overridden by inserting patch cables into the correct jacks. These pre-wired connections are often referred to as being normalled. For example: An internal noise source may normally be connected to one channel of a mixer that appears before the filter, but if you insert a patch cable into a jack usually labeled external input, this “normalled” connection is broken and replaced by your external connection.*

#### Notch Filter

*This is a particular type of filter mode where audio frequencies or harmonics around the corner or cutoff frequency setting are removed, nor “notched out” of the overall spectrum. It is the opposite of a bandpass filter, which only passes harmonics around the cutoff frequency. Notch filters tend to have a subtle effect on the sound; moving (modulating) the cutoff frequency can result in a weak phasing sort of sound. Notch filters are often used in sound systems to weaken or remove a problematic frequency, such as ground loop hum, a resonance in a room, or other annoying peak in the harmonic spectrum of a sound. Think of using a notch filter in a patch to hollow out a sound, leaving room in the harmonic spectrum for other sounds to exist with less competition, or just to create a sound more likely to catch the ear because something that is expected is instead missing.*

#### Notch

*A narrow band of audio frequencies.*

#### NW/m

*The standard unit in measuring the amount of magnetic strength on analog tape. A Weber is a unit of magnetic strength, but it is too large a unit to apply to the magnetism in tape recorders, so nanowebers is used instead. Nanowebers per meter of tape effectively describes the signal strength that is being recorded to tape.*

### Nybble

*Nybble (or Nibble) – One half byte of computer data, or 4 bits.*

### Nyquist Frequency

*In digital recording, the highest frequency that can be recorded and reproduced properly, equivalent to a one-half the sampling rate. (For example, with the common sampling rate of 44,100 kHz per second, the Nyquist frequency would be 22,050 kHz.) Aliasing begins to occur with frequencies that exceed this threshold. (See also “Aliasing.”)*

### Nyquist Rate

*The lowest sampling rate that can be used to record and reproduce a given audio signal, equivalent to twice the highest frequency. If the highest frequency found in an analog signal or sound is 18,000 kHz, theoretically the signal must be sampled at a minimum of 36,000 kHz per second—otherwise, the signal is considered to be undersampled and aliasing will occur. This is essentially the inverse principle of the Nyquist Frequency. (NOTE: the sample rate of 44,100 kHz/second is considered the standard sample rate because it easily covers the upper range of human hearing, which is about 20,000 kHz.)*

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### Octave Divider

*A module that creates a new tone one or two octaves below the fundamental harmonic – the “pitch” – of the sound coming into it, to emphasize the bass. Sometimes also known as a suboctave or sub bass function.*

### Octave

*An octave is a typical musical interval. For example, all of the “C” notes on a keyboard are octaves apart from each other. To play a note that is one octave higher in tuning, you need to double its pitch; to play an octave lower, you need to cut the pitch in half. In patch terms, this typically means adding or subtracting 1 volt to get a one octave change in pitch; some oscillators also have octave switches on their front panels that add or subtract these voltages for you (all they are not always perfectly accurate; you often need to re-tune after switching octaves). Suboctave or subharmonic generators divide the input pitch by 2 or 4 to create new waveforms that are one or two octaves lower in pitch, which adds bass.*

### Off Axis

*Veering away from the imaginary line (axis) directly in front of the receiving end of a microphone. Measured as degrees of an angle. (For example, a sound coming from directly behind the microphone is said to be 180 degrees off-axis.)*

### Offset Time

- 1) The SMPTE time that will trigger a MIDI sequencer to begin.*
- 2) The amount of position difference needed to get two reels to play the music in time.*

### Offset

*In simple terms, Offset modules usually add or subtract a voltage from a signal passing through – such as shifting a 0 to +10v signal to instead vary between -5 and +5 volts.*



**Ohm's Law**

*The mathematical relationship between voltage, current and resistance.*

**Ohm**

*The unit used to measure the amount of opposition (impedance) to electrical current flow in a signal or device. (See also "Impedance.")*

**Omni Mode**

*A setting that enables a MIDI device to recognize and respond to all MIDI channels at once.*

**Omni**

*A prefix meaning "all."*

**Omnidirectional Pattern**

*In microphones, picking up evenly from all directions (sometimes also called "Nondirectional"). 2) In speakers, sending out the signal evenly in all directions.*

**On Axis**

*The position directly in front of the diaphragm of a microphone, in line with its movement.*

**Open Circuit**

*An electrical circuit that is disconnected, interrupted or incomplete, preventing the flow of electricity.*

**Operating Level**

*(Sometimes called "Reference Level") The maximum level that should not be exceeded in normal operation.*

**Operational Amplifier**

*(Abbreviated "Op Amp") An amplifying circuit used in most audio and electronic devices.*

**Operational Transconductance Amplifier**

*An OTA (operational transconductance amplifier) circuit is one that converts an input voltage to an output current. This is a popular amplifier design as it can be less prone to going into saturation (clipping), has good bandwidth, and is also known for a "warm" sound. Therefore, you may find it in VCAs (voltage controlled amplifiers). Current can be thought of as the inverse of resistance, so what you have in an OTA circuit is in essence a voltage to resistance device that makes it possible to add voltage control to circuits such as filters. In general, when someone touts they have an OTA based filter, they usually mean it has a "warm" sound...unless it's an MS-20 filter clone, in which case it's thinner and more edgy. In reality, using an OTA is more about convenience of design than creating a specific sound.*

**Operator**

*There are a few different synthesis techniques where one usually audio-rate signal does something to another audio signal. For example, in frequency modulation (FM), a second signal (called the modulator) varies the frequency (pitch) of the main signal, called the carrier. These two signals or oscillators are often referred to as operators, particularly in FM patches. You're more likely to hear this term used when working with a dedicated FM synthesizer like a Yamaha DX-7 and its descendants, than with a modular system.*

### OR function

*One of the most common Boolean or binary logic functions, OR says if there is a gate on (or “high”) signal at any of the inputs (i.e. input 1 or input 2 or input 3, etc.), to output a gate on signal. A NOR function has an inverted output: it would only be on (high) if all inputs were low (not input 1 nor input 2 are high). An XOR (Exclusive OR) would only output a high signal if one of the inputs was high, but not if both inputs were high (or low). Finally, an XNOR is the invert of an XOR function.*

### Oscillator

*At its core, to oscillate means to vary back and forth in a repeating pattern. The main sound generator in a modular system is called an oscillator because its output varies up and down (oscillates) in voltage in a repeating pattern. This pattern is referred to as its waveshape (such as a square wave, that alternates between high and low voltages); how fast this pattern repeats is called its frequency or pitch. An acoustic instrument equivalent of an oscillator is a string that vibrates back and forth on a guitar, a drum head that vibrates up and down, or the vibrations in the reed of a woodwind instrument. The vibrations of a modular synth’s oscillator just happen with electricity going down a wire rather than a physical object vibrating in air. (Eventually this electricity is routed to a speaker, which then vibrates the air with the same pattern sent to it over a wire.)*

### Oscilloscope

*This is a piece of test equipment that displays voltage fluctuations as graphical waveforms. A ‘scope can run at a wide range of frequencies, displaying slowly changing voltages like LFOs or envelopes, or quickly changing voltages like oscillators and noise. Oscilloscopes used to be bulky pieces of external equipment, but now you can get USB scopes that offload the display portion of the job to your computer, or scopes as modules.*

### OTA

*An OTA (operational transconductance amplifier) circuit is one that converts an input voltage to an output current. This is a popular amplifier design as it can be less prone to going into saturation (clipping), has good bandwidth, and is also known for a “warm” sound. Therefore, you may find it in VCAs (voltage controlled amplifiers). Current can be thought of as the inverse of resistance, so what you have in an OTA circuit is in essence a voltage to resistance device that makes it possible to add voltage control to circuits such as filters. In general, when someone touts they have an OTA based filter, they usually mean it has a “warm” sound...unless it’s an MS-20 filter clone, in which case it’s thinner and more edgy. In reality, using an OTA is more about convenience of design than creating a specific sound.*

### Out of Phase

- 1) Being similar to another signal in amplitude, frequency and wave shape but being offset in time by part of a cycle.*
- 2) Having the opposite polarity.*

### Outboard Equipment

*Equipment that is used with, but is not a part of, a console.*

### Output Impedance

*The opposition to the flow of electrical current by the output circuits of an amplifier (or other device).*

### Output Level

*The signal level at the output of a device.*

### Output

- 1) *The jack or physical location of where a device sends out a signal.*
- 2) *The signal put out by a device.*

### Overdubbing

*The process of recording an additional musical performance over an existing recording, usually on its own track. Overdubbing has become a common recording technique with the advent of multitrack recording, first on multitrack analog tape, and more recently via computers and Digital Audio Workstations (DAWs).*

### Overtone

*Any harmonic in a tone except the fundamental frequency. (See also “Partial.”)*

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### Pad

- 1) *A device or circuit that attenuates an incoming signal, usually to prevent overload of an amplifier that follows along the signal path. (Also sometimes called “Attenuator pad.”)*
- 2) *A device with a surface that can be hit by a drum stick; hitting the pad produces an output signal pulse (or MIDI command) that causes a drum machine or synthesizer to sound a drum sound.*
- 3) *A type of synthesizer patch/program used to create sustained background or atmospheric sounds.*

### Pan (Panning)

*The process of “placing” a particular sound within the stereo field. This is accomplished by controlling the balance of the signal between the left and right speakers so the ear hears the sound as coming from a particular point in the sonic space between left and right. This sonic space is sometimes called the “stereo panorama,” from which the word “panning” is derived. In surround sound, panning occurs in a 360° sound space, not just left-right.*

### Panpot (or Pan Pot)

*Short for “Panoramic Potentiometer,” a panpot is a knob in the channel strip that controls the panning of the audio signal in the stereo (or surround) space by controlling how much of the signal is sent to each speaker or channel.*

### Parallel Jacks

*Several jacks that are wired so that each connection is wired to the corresponding connection of other jacks.*

### Parallel Port

*A connector that is able to transmit and receive digital data at the same time through different pins.*

### Parameter

*Parameter is the fancy name given to any value or property or control of a synthesizer module that you’re trying to change. For example, an oscillator’s parameters typically include its pitch and the*

*width of its pulse wave. A filter's parameter will include its cutoff frequency (pitch), the amount of resonance (feedback), and possibly other controls such as a blend between its different outputs. Parameter was a popular term to describe a value you could change in software, and it's been carried over by some to hardware modular synths.*

#### Parametric Equalization

*An equalizer in which all parameters of equalization can be adjusted to any amount, including the center frequency, the amount of boost or cut, and the bandwidth.*

#### Paraphonic

*A paraphonic synth is one where all of the notes being played go through a single filter (VCF) and amplifier (VCA). This was a popular scheme in the early days of polyphonic synths in that a separate oscillator (or organ-like frequency divider, in the case of "string synths" and the such) was used for each note played, but they were mixed before all going to the filter and amp to articulate the note(s). It was not uncommon for some monophonic synths to allow two to four independent notes to independently control the pitch of its oscillators, while still going through a single filter. This works great for chords; it doesn't always work all that great for when a new note is played while others are being held as all of the notes will be re-articulated together.*

#### Partial

*1) Another word for overtone.*

*2) One of a number of sine waves that makes up a complex sound, helping to define the timbre. This concept is a key part of creating sounds in synthesizers: in additive synthesis, a number of partials are combined to create a certain tone.*

#### Pass Band

*The frequency range of signals that will be "passed" by a filter, rather than reduced.*

#### Passive Device

*A component that does not generate or control electrical current (as opposed to an "Active Device"). In audio applications, this usually refers to a piece of gear that does not include an amplifier as part of its design. For example, active speakers are self-powered, while passive speakers require an external amplifier in order to reproduce sound. (See also "Active Device.")*

#### Passive

*Means no active (i.e. connected to a power supply) electronics are involved – such as sending a signal straight through a potentiometer control, instead of using op amps and other electronics to create a mixer circuit around it. Passive is cheap and easy, and does not add noise to a signal. But passive electronics cannot buffer one signal from another (meaning they might interact in undesirable ways), and cannot boost, offset, or invert a signal.*

#### Patch Bay

*Patch Bay (or Patchbay, Patch Field, Patch Panel) – A panel or component containing a series of jacks with connections for most of the inputs and outputs of the console and components in the studio, used for the purpose of organizing, managing and regulating signal flow.*

#### Patch Cable

*The cables used to connect together the different inputs and outputs in a modular synthesizer,*

*carrying electrical control voltages and audio. The term came from the old telephone patch boards where an operator had to physically connect two callers together using electrical cables. As different modular formats often use different connector standards, you need to make sure the connectors at the ends of the wire in a patch cord are the size you need (3.5mm for Eurorack, 1/4" for 5U/Moog Unit, or banana for Serge or Buchla control voltages).*

#### Patch Cord (or Patch Cable)

*An insulated cable with plugs on each end used to route audio signals. Patch cords are typically thought of as short cables used to make connections in the patch bay (hence the name); however, patch cords facilitate almost any kind of audio connection between devices, can come in a wide range of lengths, and can include a number of different types of connectors.*

#### Patch Field

*A panel or component containing a series of jacks with connections for most of the inputs and outputs of the console and components in the studio, used for the purpose of organizing, managing and regulating signal flow.*

#### Patch Librarian

*A computer program allowing for the storing of sound patches outside of a synthesizer via MIDI.*

#### Patch Panel

*A panel or component containing a series of jacks with connections for most of the inputs and outputs of the console and components in the studio, used for the purpose of organizing, managing and regulating signal flow.*

#### Patch

*The shorthand term used to refer how a series of modules are interconnected to create a sound, derived from the fact that patch cords are used to connect the modules together.*

*1) To route or reroute the signal in an audio system (such as a console) by using short cables with plugs inserted into jacks.*

*2) A sound setting or program on a synthesizer.*

#### Path

*Short for Signal Path, the way in which current does or may travel in a circuit or through a device.*

#### PCM

*Pulse Code Modulation - A process by which analog signals are translated to digital code. This is done by taking samples of the amplitude of the analog signal at regular rapid intervals, then translating it into binary numbers as a digital representation of the original signal. The faster the sample rate, the better the digital reproduction. PCM is the most common form of A/D conversion in digital audio.*

#### PD

*Phase Distortion synthesis was used by Casio originally in the 80s in the CZ line of synths. It is related to FM (frequency modulation), with enough differences to avoid problems with the patent used by Yamaha's FM synths of the era. Intriguingly, it did a good job at mimicking many "analog" synth effects including the sound of a resonant filter.*

#### Peak Filter

*An EQ circuit/filter that boosts or cuts the middle (center frequencies in an audio signal, as opposed to high-pass or low-pass filters. (NOT to be confused with amplitude peaks.)*

#### Peak Meter

*A meter which detects the absolute peak value of a waveform, as opposed to the RMS value. (See also "Peak Value," "Root-Mean-Square," "RMS Meter.")*

#### Peak to Peak Value

*The measure of the total amplitude between positive and negative peaks in an audio signal. Equal to twice the peak value for a sine wave. (See also "Peak Value.")*

#### Peak Value

*The measure of the maximum positive or negative value (amplitude) of a waveform at any moment. In audio, this is visually depicted as the farthest point of the waveform above or below the zero axis.*

#### Pedal Board

*A board with several guitar pedals attached and inter-connected so that a guitar player can conveniently activate a number of different effects.*

#### Phantom Power

*A system used to supply DC voltage to condenser mics and other components through the audio cables, eliminating the need for external power supplies.*

#### Phase Addition

*The increased audio energy that happens when waveforms are in similar phase relationships, resulting in an increase in volume up to twice what it should be.*

#### Phase Cancellation

*The opposite of phase addition, this is the reduction of energy that occurs when two similar waveforms that are out of phase with one another and begin cancelling each other out, either greatly reducing or eliminating the volume. When two identical wave forms are completely out of phase (by 180 degrees), the result in theory is a total silencing or cancellation of the signal.*

#### Phase Distortion Synthesis

*Phase Distortion synthesis was used by Casio originally in the 80s in the CZ line of synths. It is related to FM (frequency modulation), with enough differences to avoid problems with the patent used by Yamaha's FM synths of the era. Intriguingly, it did a good job at mimicking many "analog" synth effects including the sound of a resonant filter.*

#### Phase Distortion

*A change in the sound because of a phase shift in the signal. Sometimes used in synthesizers as a method of altering the wave shape or adding harmonics to the sound.*

#### Phase Lock

*Any of a number of processes used to help synchronize signals or devices by correcting phase differences. For example, in analog tape machines, phase locking helps to keep multiple machines synced together by sensing phase differences in the playback of pilot tones by the two machines and adjusting the speed to eliminate the phase difference. In synthesizers, phase locking controls one*

*tone generator so that it begins its waveform in phase with the signal from another tone generator. Phase-locked loops (PLL) are reference signals used in the clock functions of electronic devices.*

#### Phase Locked Loop

*A phase locked loop is, in essence, an oscillator that tries to match the frequency of – or more importantly, a division or multiple of the frequency of – another signal. This is most commonly used to create a frequency that is much higher than the incoming reference signal – such as a timing module that can create an output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a very high frequency oscillator that is locked to a multiple of an incoming pitch – perhaps to drive a special circuit such as a switched-capacitor filter.*

#### Phase Modulation

*Some would say this is the pedantically correct term for frequency modulation (FM), as the act of causing a carrier oscillator to play back faster and slower (quickly changing its frequency to be higher and lower) is the same as advancing and retarding position (phase) of the normal playback of a waveform. But don't get bogged down by terminology when creating an FM patch; just connect the output of one oscillator to the pitch input of another and go for it.*

#### Phase Reversal

*A change in a circuit to get the waveform to shift by 180 degrees.*

#### Phase Shift

*A delay introduced into an audio signal measured in degrees delayed.*

#### Phase Shifter

*This effect splits a signal into two copies. One copy is fed through an “all pass filter” which does not attenuate any of the original harmonics like a low pass or high pass filter does, but which does alter the phase of the signal, causing those harmonics to have varying amounts of phase shift in relation to the original depending on their frequency. Mix these two copies back together, and different harmonic components of the original sound cancel each other out (see Phase), resulting in a notch filter effect. Each “stage” – all-pass filter section – of a phase shifter creates one of these notches. More stages create more notches, and a deeper effect.*

#### Phase-Locked Loop

*PLL - Any of a number of processes used to help synchronize signals or devices by correcting phase differences. For example, in analog tape machines, phase locking helps to keep multiple machines synced together by sensing phase differences in the playback of pilot tones by the two machines and adjusting the speed to eliminate the phase difference. In synthesizers, phase locking controls one tone generator so that it begins its waveform in phase with the signal from another tone generator. Phase-locked loops (PLL) are reference signals used in the clock functions of electronic devices.*

#### Phase

*A measurement (expressed in degrees) of the time difference between two similar waveforms. One cycle of a waveform is considered to have 360 degrees, just like a circle. How far you move around the circle (or through the waveform) can be defined by the phase. For example, if you are one-quarter of the way through a waveform's cycle, your phase is 90°.*

### Phasing

*An effects sound created by varying the phase shift of an audio signal, then mixing it with the direct signal.*

### Phon

*A unit of apparent loudness, numerically equal to the same number of dB as a tone playing at 1000 Hz. For example, a sound is said to be 60 phon if it is perceived to be as loud as a 1000-Hz tone playing at 60dB.*

### Phone Plug

*A plug (or its mating jack) with a diameter of 1/4 inch and a length of 1 1/4 inches used for interconnecting audio.*

### Phono Plug

*A common audio connector found on most stereo systems with a center pin as one connection and an outer shell as the second connection.*

### Physical Modeling

*One approach to (often digital) synthesis is to recreate the components of actual instruments – such as a vibrating string or tube, or a resonating body such as the shell of a guitar or drum – and string those together to create sounds. There are a handful of modules available which perform this modeling to create their sounds.*

### Pickup Pattern

*The shape of the area in front of or around the microphone from where it evenly picks up sound. Many use this term interchangeably with “polar pattern,” but a polar pattern gives more detail about microphone sensitivity. (See also “Polar Pattern.”)*

### Pickup

- 1) *A device on an electric guitar or other instrument that puts out an audio signal according to the string motion on the instrument.*
- 2) *See “Contact Microphone.”*

### Pinch Roller

*A rubber (or plastic) wheel on a tape recorder that pinches the tape between it and the capstan, allowing the capstan to pull the tape.*

### Ping-Ponging (Bouncing)

*The technique of combining and mixing multiple tracks onto one or two tracks (mono or stereo). This can be done in real-time or analog by playing the tracks through the console and recording them onto separate tracks, or digitally through a digital audio workstation. Bouncing was once used frequently by engineers to free up additional tracks for recording, but in digital workstations where tracks are virtually unlimited, this practice is basically obsolete. Today, engineers typically bounce tracks for the purpose of creating a preliminary or final mix of a song.*

### Pink Noise

*A noise signal similar to white noise, containing all audible frequencies, but with equal energy per octave as opposed to all frequency bands. Engineers frequently use pink noise as a tool to tune and*



*calibrate audio equipment. (See also “White Noise.”)*

*Noise is a random, unpitched signal that, at audio rates, can sound like hissing or the wind. Pink noise has equal energy (sound level) per octave. As each higher octave has double the frequency of the octave below it which spreads out the energy over a wider range of frequencies, pink noise tends have a more natural, less electronic sound with more bass and less high end – especially when compared to white noise, which has an equal energy per number of hertz (frequency) and therefore tends to sound very bright.*

#### Pitch Bend

*A mechanism on a synth, keyboard or controller that can cause the pitch of the note to move up or down by a small amount.*

#### Pitch to Voltage Converter

*A device that detects the frequency of an audio waveform and changes it into a control voltage, which is in turn fed to an oscillator that produces a pitch at the same frequency.*

#### Pitch-to-MIDI Converter

*A device that detects pitch in an analog audio signal and translates it into MIDI information. (Also called “Audio-to-MIDI-Converter.”)*

#### Pitch-to-Voltage Converter

*A device that detects the frequency of an audio waveform and changes it into a control voltage, which is in turn fed to an oscillator that produces a pitch at the same frequency.*

#### pitch

- 1) The perception of frequency by the ear (a higher or lower tone of music).*
- 2) A control on a tape transport which adjusts the speed slightly up or down, changing the pitch and time of the music.*

#### Plate Reverb

*A device that produces artificial reverberation by sending vibrations across a metal plate via a transducer similar to a speaker driver. Physical plate reverbs today are considered a vintage form of artificial reverb; nowadays, most plate reverb effects are emulated digitally by plugins or reverb units.*

#### Playback Head

*A transducer that converts magnetic flux recorded on tape into an audio signal for playback.*

#### Playback Mode

*A configuration on a console that allows quick playback of the signal previously recorded on tape or via DAW via the monitor mixer.*

#### Playback

- 1) The reproduction of recorded audio.*
- 2) In motion picture or video production, the reproduction of the music over loudspeakers so the performers/musicians can perform in time to the music for the camera.*

#### Playlist

- 1) See “Take.”*

2) *A user-defined selection of songs; a feature available on most streaming and digital media players.*

#### PLL

*A phase locked loop is, in essence, an oscillator that tries to match the frequency of – or more importantly, a division or multiple of the frequency of – another signal. This is most commonly used to create a frequency that is much higher than the incoming reference signal – such as a timing module that can create an output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a very high frequency oscillator that is locked to a multiple of an incoming pitch – perhaps to drive a special circuit such as a switched-capacitor filter.*

#### Plug

*A connector, usually on a cable, that mates with a jack.*

#### Polar Pattern

1) *In microphones, a graphic display of the area around the microphone that is sensitive to sound waves, detailing the audio output levels in dB of sound arriving from different directions. Similar to “Pickup pattern,” but more specific.*

2) *In speakers, a graphic display of the speaker’s dispersion of sound.*

#### Polarity

*The direction of current flow or magnetizing force.*

#### Polarizer

*An inverter multiplies an incoming control voltage by  $-1$ . In the case of a gate or logic inverter, it reverses the high and low states so that (for example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a polarizer, as it changes the polarity (+ versus  $-$ ) of a signal. A control voltage inverter is often combined with an offset voltage to adjust the output voltage into the desired range. For example, if you had an envelope generator that had an output range of 0 to +8 volts, and you just inverted it, the result would be 0 to  $-8$  volts. Since some modules such as voltage controlled amplifiers usually expect only positive voltages, you would then need to add 8 volts to that result to get an upside-down (inverted) envelope that still had an overall range of 0 to +8v.*

#### Polarizing Voltage

*In condenser and electret microphones, the introduction of a small amount of electrical current to create the magnetism by which the capacitor converts audio signals to electrical current. In condenser microphones, polarizing voltage is provided externally (see also “Phantom Power”); in electret microphones, the polarizing voltage is permanently impressed on the condenser during manufacturing.*

#### Pole Pieces

*Iron or other magnetic material that conducts magnetic force for use in transducers like record heads, playback heads, microphones, speakers, etc.*

#### Pole

*This is a technical term that helps describe the design of a filter. Each pole of a filter attenuates frequencies beyond its cutoff or corner frequency by 6 decibels (dB)/octave; the more poles, the stronger the filtering effect. A 4-pole low pass filter, for example, attenuates frequencies one octave*

*above its cutoff frequency by 24 dB; frequencies two octaves above the cutoff are attenuated by 48 dB and so forth.*

### Polyphonic

*The term “polyphonic” refers to a synthesizer that can play more than one individually articulated note at a time; in most cases, those notes all play a similar sound or patch.*

*Able to play more than one pitch or “voice” at the same time. A term commonly used to describe synths and keyboards. (See also “Voice.”)*

### Ponging (Bouncing)

*The technique of combining and mixing multiple tracks onto one or two tracks (mono or stereo). This can be done in real-time or analog by playing the tracks through the console and recording them onto separate tracks, or digitally through a digital audio workstation. Bouncing was once used frequently by engineers to free up additional tracks for recording, but in digital workstations where tracks are virtually unlimited, this practice is basically obsolete. Today, engineers typically bounce tracks for the purpose of creating a preliminary or final mix of a song.*

### Pop Filter

*A device that is placed over a microphone or between the microphone and vocalist to prevent loud “pop” sounds created by the vocalist’s breath directed toward the microphone.*

### Port

*1) A connection point in computer or electronic device for transmitting and receiving digital data, similarly to how a jack receives and transmits audio signals.*

*2) An opening or vent in a speaker case that resonates with air movement in the speaker, used in bass reflex speakers and woofers to enhance low frequencies.*

### Portamento

*A pitch change that smoothly glides from one pitch to another. Also refers to the synthesizer mode or MIDI command that allows or causes this to happen.*

### Post Production

*Refers to the work of adding tracks, editing and other fine tuning after primary recording or filming has taken place. Post-production in recording includes such things as additional overdubs, editing, mixing and mastering. Post-production in film includes a wide range of additional audio and visual effects. NOTE: We mention film in this context because film post-production includes a lot of audio work (e.g., voiceovers, foley, audio mixing and editing) to the point that many audio engineers are involved in film post-production as a full-time career.*

### Post Roll

*A segment of blank tape (or track silence, on a DAW) that runs past the end of the recording. (See also “Pre-Roll.”)*

### Post-Fader

*Refers to an aux send position or setting that places the send after the channel fader within the signal path. Sending a signal post-fader means the fader itself affects the level of the send signal, as opposed to pre-fader. (See also Pre-Fader.)*

### Post

*Refers to an aux send position or setting that places the send after the channel fader within the signal path. Sending a signal post-fader means the fader itself affects the level of the send signal, as opposed to pre-fader. (See also Pre-Fader.)*

### Pot

*Often thought of as a fancy word for “knob,” a potentiometer is basically any mechanism that controls input or output voltage by varying amounts (for example, panning a signal left/right, volume control, or the amount of signal sent to an aux send or bus. Potentiometers can be knobs or faders, meaning that almost every control on a console that isn’t a button or switch is a potentiometer. However, many engineers commonly refer to faders as “faders” and knobs as “pots.”*

### Potentiometer

*(Abbreviated “Pot”) Often thought of as a fancy word for “knob,” a potentiometer is basically any mechanism that controls input or output voltage by varying amounts (for example, panning a signal left/right, volume control, or the amount of signal sent to an aux send or bus. Potentiometers can be knobs or faders, meaning that almost every control on a console that isn’t a button or switch is a potentiometer. However, many engineers commonly refer to faders as “faders” and knobs as “pots.”*

### Power Amplifier

*(abbreviated “Power Amp”) A device that amplifies a line level signal to drive a speaker or set of speakers. (See also “Line Level.”)*

### Power Distribution Board

*This simple circuit board takes the output of your modular system’s power supply and creates multiple copies of it, routed to connectors that go to your individual modules.*

### PPQN

*When you send a clock signal (usually a gate signal or other electrical pulse) around a modular synth to move sequencers through their steps and the such, it’s good to know how fast that clock is pulsing. This is usually defined in terms of how many pulses there are per quarter note – PPQ or PPQN for short. If the clock is just happening every quarter note, then the clock speed is 1 PPQN; in the case of DIN Sync (a popular standard among early Roland synths, with DIN being the type of electrical connector used) or MIDI clocks, the standard is 24 PPQN. This means the master pulse can define a triplet for every 8th note (8 x 3).*

### Pre / Post Switch

*A switch on the input module that determines whether the send control comes before or after the main channel fader in the signal path (See also “Pre-Fader,” “Post-Fader.”)*

### Pre Emphasis

*A boosting of high frequencies during the recording process to keep the audible signal above the noise floor.*

### Pre Fader

*Refers to an aux send position or setting that places the send before the channel fader within the signal path. Sending a signal pre-fader means the fader does not affect the level of the send signal, as opposed to pre-fader.*

### Pre-Delay

*A parameter on a reverb unit or plugin that determines the amount of time (delay) between the original dry sound and the early reflections of reverberation. This feature is often used to simulate the natural acoustic properties of a room, but can also be used to create interesting unnatural effects.*

### Pre-Echo

*(Also called “Forward Echo”) A compression artifact that often occurs in digital audio in which an “echo” of a sound (or part of a sound) is heard ahead of the sound itself, often due to the data inconsistencies in certain compressed digital formats. A type of pre-echo can also sometimes occur in the end product of a recording, occurring on tape as a result of low-level leakage caused by print-through, and also on vinyl records due to physical differences and/or deformities in the grooves between silence and a loud transient. In digital formats, pre-echo is generally an unwanted problem that requires additional signal processing to resolve—but in some cases it can also be used on purpose as a sound effect (not to be confused with “Reverse Echo”).*

### Pre-Fade Listen (PFL)

*A function on the channel strip of a mixer or DAW that allows a channel signal to be heard and often metered before the channel fader.*

### Preamplifier (Preamp)

*A low-noise amplifier designed to take a low-level signal (for example, from a microphone) and bring it up to normal line level before sending it into the mixing console.*

### Precedence Effect (Haas Effect)

*Simply stated, a factor in human hearing in which we perceive the source of a sound by its timing rather than its sound level. In his research, Helmut Haas determined that the first sound waves to reach our ears help our brains determine where the sound is coming from, rather than its reflection or reproduction from another source. The reflection of the sound must be at least 10dB louder than the original source, or delayed by more than 30ms (where we can perceive it as an echo), before it affects our perception of the direction of the sound. This is what helps us distinguish the original sound source without being confused by reflections and reverberations off of nearby surfaces. Understanding the Haas effect is particularly useful in live audio settings, especially in large venues where loudspeakers are time-delayed to match the initial sound waves coming from the source.*

### Precision Adder

*Synthesizers are very sensitive to unintentional variations in pitch control voltage – any error can result in the oscillators under control going out of tune. Therefore, whenever you add together pitch control voltages inside a modular synth, you really should be using a precision adder that precisely adds together the pitch voltages without introducing an error. Ordinary mixers might slightly attenuate or amplify a voltage passed through them, which in most cases would create tuning errors.*

### Premix

*1) The process of mixing a set of tracks as group, then managing the mixed group in the context of the other tracks by routing them to an auxiliary channel. Consolidating tracks by bouncing is a form of premixing, but a premix is not necessarily pre-recorded. (See also “Bouncing.”)*

2) *An important part of film post-production in which the process of mixing a section of audio for combination with the others. Dialogue, Foley, SFX and music may all be premixed before being combined together under the video.*

#### Presence Frequencies

*The range of audio frequencies between 4 kHz and 6 kHz that when boosted, can increase the sense of presence, especially on voices.*

#### Presence

1) *In amplification and mixing, the boosting of upper-mid frequencies to cause a sound or instrument to cut through, creating the impression that the sound source is more “present,” right next to the listener.*

2) *See “Room Tone.”*

#### Preset

*A factory programmed set of parameters on a synth, signal processor, plug-in or other electronic device.*

#### Pressure Microphone

*(Also called “pressure operative microphone”) – A microphone whose diaphragm responds to incoming sound wave pressure as it works against the normal or controlled air pressure inside the microphone case. This design makes the diaphragm sensitive to pressure regardless of direction, giving it an omnidirectional pickup pattern. (See also “Omnidirectional Pattern.”)*

#### Pressure Sensitivity (Aftertouch)

*A feature in some keyboard instruments by which applying additional pressure to a key after it has been pressed can activate an additional MIDI control command. a synthesizer or Keyboard Controller of After Touch (a control or operational function of a synthesizer where pressing a key after it has been pressed, and before it is released, will activate a control command that can be set by the player).*

#### Pressure Zone Microphone (Boundary Microphone)

*An omnidirectional microphone designed to be placed flush against a flat surface (or boundary), effectively creating a “half-Omni” pickup pattern while eliminating the danger of phase issues from reflected sounds. A popular type of boundary microphone is Crown Audio’s trademark Pressure Zone Microphone (PZM).*

#### Pressure-Gradient Microphone

*(Also called “Velocity Microphone”) A microphone whose diaphragm is exposed front and back, with diaphragm movement being caused by the pressure difference between its front and back. This creates a bi-directional or “figure-8” pickup pattern (See also “Bi-Directional Pattern.”)*

#### Pressure

*Some keyboards measure how hard you press down on the keys, and convert this to a voltage (or other control signal such as MIDI, which can then be converted into a control voltage) that you can use to add expression to a note, such as adding vibrato or opening the filter wider. Monophonic aftertouch measures one pressure value for the entire keyboard, regardless of which key(s) you are pressing; polyphonic aftertouch produces a signal for each individual key. Important trivia: Touch plate keyboards actually measure the surface area of the skin touching them rather than pressure or*

*force – so you can increase or decrease the aftertouch amount by rolling between the tip and length of your finger.*

#### Print Through

*The unwanted transfer of magnetic flux from one layer of analog tape to another.*

#### Pro Tools

*Avid's trade name for its digital audio workstation (DAW) that has become an industry standard in professional recording studios.*

#### Producer

*In music, the producer is the director of an audio recording project; the person responsible for getting a final product of desired quality within a budget.*

#### Production Studio

*Broadly speaking, any space dedicated to production within the arts, for example, film/video, animation or post production. In the context of audio, a production studio is effectively a recording studio that specializes in the assembly and mixing of commercials and radio programs from pre recorded music and effects with newly recorded dialogue.*

#### Production

- 1) *The collective actions that go into producing music.*
- 2) *Describing the quality of a recording—the end result of production decisions during the recording and mixing process.*

#### Program Change

*A MIDI message that tells the receiving device to change presets.*

#### Programmable

*Able to have the parameters changed by the user, especially in a computer controlled device.*

#### Prompt

*A set of instructions for the user to follow, which appears on a computer screen.*

#### Protocol

*In digital and information technology, a set of rules governing the structuring and transmitting of data in a standardized format so all related devices can properly interpret the data.*

#### Proximity Effect

*The natural boost in the microphone's output for bass frequencies as the mic is placed closer to the sound source.*

#### Psychoacoustics

*The study of how humans perceive and respond to sound, not just in the context of interpreting the physical sound waves, but also taking psychological and emotional factors into account. This branch of science is helpful to audio engineers in understanding how the brain interprets various sounds and frequencies.*

#### Puck

*Any circular piece of metal, fiber, rubber, etc., which drives something from a rotating power source.*

*A common example in the recording studio is the puck in a rotating Leslie speaker.*

#### Pulse Code Modulation (PCM)

*A process by which analog signals are translated to digital code. This is done by taking samples of the amplitude of the analog signal at regular rapid intervals, then translating it into binary numbers as a digital representation of the original signal. The faster the sample rate, the better the digital reproduction. PCM is the most common form of A/D conversion in digital audio.*

#### Pulse Per Quarter Note

*When you send a clock signal (usually a gate signal or other electrical pulse) around a modular synth to move sequencers through their steps and the such, it's good to know how fast that clock is pulsing. This is usually defined in terms of how many pulses there are per quarter note – PPQ or PPQN for short. If the clock is just happening every quarter note, then the clock speed is 1 PPQN; in the case of DIN Sync (a popular standard among early Roland synths, with DIN being the type of electrical connector used) or MIDI clocks, the standard is 24 PPQN. This means the master pulse can define a triplet for every 8th note (8 x 3).*

#### Pulse Width Modulation

*Most oscillators that output a square waveform also have an additional control voltage input that sets the width of the top portion of the “square” wave (obviously, making the top portion wider makes the bottom portion narrower and vice versa). The act of varying the width of the resulting pulse wave creates a sort of Doppler shift; varying the width back and forth – for example, by modulating the pulse width with a low frequency oscillator – creates a chorusing effect that can sound like a detuned pair of oscillators. The resulting effect is referred to as pulse width modulation.*

*The process of using a control voltage to vary the width of a pulse wave form, essentially switching between square waves and pulse waves. This has the effect of creating richer timbres, giving sounds a thicker, more lush feel, or of giving a digital sound more analog properties.*

#### Pulse

*Pulse has a couple of different meanings in a modular synth. When you alter the shape of a square wave so that one portion is narrower than the other, it is referred to a pulse wave (see Pulse Wave Modulation below). Also, a narrow gate or trigger used as a clocking signal for sequencers and the such is often referred to as a pulse.*

*1) The steady beat in music based on its tempo, whether audible or perceived.*

*2) A type of sound wave commonly created and manipulated by synthesizers whose waveform is characterized by sharp rises and drops in amplitude like a square wave, but whose peaks are shorter than its troughs, giving the wave a pulse-like feel. Also called “Pulse Wave.”*

#### Pumping and Breathing

*In studio jargon, an effect created when a compressor is rapidly compressing and releasing the sound, creating audible changes in the signal level. “Pumping” generally refers to the audible increase of sound levels after compression has taken place; “breathing” refers to a similar effect with vocals, raising the signal volume just as the vocalist is inhaling. Pumping and breathing is a sign of cheap compression or over-compression, and is usually undesirable, although some engineers and musicians use it on purpose occasionally to create a particular effect.*



### Punch In / Punch Out Recording

*The process of activating and/or deactivating the record function on tape or DAW during playback of a passage, usually as the performer plays/sings along. This can be used either as a method of doing quick overdubs, or as a way of getting a better take on a certain passage without having to start the track from the beginning.*

### Pure Tone

*A tone consisting of only the fundamental frequency with no overtones or harmonics, graphically represented as a simple sine wave.*

### PVC

*PVC stands for pitch to voltage conversion. In the quest to play a voltage-controlled synthesizer with something other than a keyboard-like thingy (touch plates included), some have designed modules or other equipment that attempt to detect the pitch of an audio signal – say, from a guitar, flute, or singer – and convert that pitch to a corresponding voltage that can drive a VCO in unison with the original sound.*

### PWM

*Most oscillators that output a square waveform also have an additional control voltage input that sets the width of the top portion of the “square” wave (obviously, making the top portion wider makes the bottom portion narrower and vice versa). The act of varying the width of the resulting pulse wave creates a sort of Doppler shift; varying the width back and forth – for example, by modulating the pulse width with a low frequency oscillator – creates a chorusing effect that can sound like a detuned pair of oscillators. The resulting effect is referred to as pulse width modulation.*

### PZM

*Abbreviation for Crown Audio’s Pressure Zone Microphone. (See also “Boundary Microphone.”)*

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### Q – (Also called “Q Factor”)

*Stands for “Quality Factor,” defining the bandwidth of frequencies that will be affected by an equalizer. The lower the Q, the broader the bandwidth curve of frequencies that will be boosted or cut.*

*If you come from the pro audio world, you may be used to Q referring to the width or narrowness of a peak or notch filter. In a synthesizer filter, when you increase the resonance (feedback), a peak forms around the cutoff frequency of the filter’s curve or shape. The higher the resonance, the higher and narrower this peak. As a result, some used to use the audio term Q to refer to the resonance amount, although you don’t hear that term used nearly as much today.*

### Quadraphonic

*A now rarely-used system of four-channel sound where the channels are designated as left front, left back, right front, right back, intended to deliver sound from all four corners of a room. Quadraphonic sound was a precursor to the surround-sound systems of today.*

### Quadrature

*You can define a full cycle of a waveform as consisting of 360 degrees, akin to a circle. One quarter*

*of the way around this circle – or moving to a point that is one quarter of the way through a cyclical wave – is 90°. A sine and cosine wave are shifted 90° degrees or a quarter cycle out of alignment (phase) with each other. Since this is a quarter of a cycle, this is often referred to as a quadrature relationship.*

#### Quantization Distortion

*Quantization Distortion/Quantization Error – The effective “error in translation” between an analog signal and its sampled counterpart due to the rounding of a large number of analog values to the nearest digital quantity. This often results in additional random frequencies in the sound, often heard as noise.*

#### Quantization Noise

*\*The modulation noise in a signal resulting from quantization error. \**

#### Quantization

*1) In digital music, the process of adjusting the rhythmic performance of music by moving the notes to precise locations on the time line, effectively “rounding” the note occurrences to the nearest defined increment.*

*2) In analog-to-digital conversion, the use of the same mathematical quantization principles to convert an analog signal into a smaller set of steps (a digital quantity).*

#### Quantizer

*A quantizer auto-corrects the input voltage to the nearest desired target, such as the voltage that corresponds to a semitone or other note in a scale. These are occasionally built into modules like sequencers or oscillators, but quite often they are standalone modules.*

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#### Rack Ears

*Rack Ears/Rack Flanges – Mounting brackets that can be attached to equipment so it can be mounted in a standard equipment rack.*

#### Rack Mounted

*Describing outboard gear that can be housed in an equipment rack.*

#### Rack Rash

*When you mount a module into a case, the head of the screw or bolt used to mount the module can scratch the faceplate of the module. These scratches are referred to as rack rash. You can almost never see it when you mount a module, as the scratches are behind the screw or bolt head, but nonetheless some will pay more for a used module that is unscratched. So buy a bag of plastic washers and put them behind the screw or bolt head just to remove another reason for someone to not buy your used module.*

#### Rack Unit

*Rack-mounted equipment usually follows a standard set of dimensions, including 19” (48.3 cm) for width, and a “rack unit” (or U for short) for height equaling 1.75” (4.4 cm) per U. Many common*

*modular synthesizer formats follow the rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25” or 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometimes referred to as MU for Moog Unit).*

#### Radiation Pattern

*A graphic depiction of speaker coverage. This is not unlike the polar pattern of a microphone, with the exception that a polar pattern describes the area where sound arrives at the microphone, while a radiation pattern describes how sound is dispersed from the loudspeaker.*

#### Radiation

*The angle and pattern of coverage of a speaker.*

#### Ramp

*In general, a ramp refers to any voltage that is steadily raising or falling; quite often it resets when it reaches a target voltage and starts over again. A sawtooth oscillator waveform is sometime referred to as a ramp. Sometimes, the individual stages of an envelope generator are also referred to a ramp as it raises from 0 volts to a maximum level such as 5v for the attack stage, then falls from this peak to the sustain level for the decay stage.*

#### Random Access Memory (RAM)

*The “short-term” memory in a computer that is used in tandem with the processor for performing immediate tasks (as opposed to hard-drive storage memory where projects are saved and recalled). In the recording studios, the more RAM a computer has, the more ability it has to handle large amounts of data at a time (for example, in multi-track recording or working with virtual MIDI instruments).*

#### Random Note Generator

*A device that generates random pitches at a set rate, used in synthesizers.*

#### Random

*Most voltages moving around inside a modular synth are very purposeful in their variations: the repeating waveforms of an audio rate or low frequency oscillator; the rising then falling voltages of an envelope generator. However, it can also be useful to have randomly wandering voltages to create everything from subtle variations in pitch to wildly varying volumes or filterings. Noise is an example of an audio-rate random signal.*

#### Rap

*To perform a spoken rhythmic part to a music or percussion performance.*

#### Rarefaction

*The reduced density of air particles during the trough of a sound wave; in the context of “compression and rarefaction,” it is the opposite of compression. (See also “Compression.”)*

#### Ratcheting

*This is a trick used with sequencers where one stage of the sequence may be triggered quickly multiple times, rather than just once as you step to that stage. For example, the result may be a series of quarter notes, with a burst of four sixteenth notes appearing instead for one or more stages.*

#### Rate

*This word is used sometimes to refer to the speed or frequency of a low frequency oscillator or similar*

*repetitive function, such a sequencer's tempo clock.*

#### Rated Load Impedance

*The input impedance, or opposition to current flow by an input of a device, that a piece of equipment is designed to feed.*

#### RCA Plug

*(Also called Phono Plug) A common audio connector found on most stereo systems with a center pin as one connection and an outer shell as the second connection.*

#### Read Only Memory (ROM)

*A type of data storage that cannot be erased or reprogrammed by the user. The most common form of ROM in audio/video settings today is optical storage media (i.e, CD, DVD, CD-ROM and DVD-ROM).*

#### Read

*To retrieve information bits from a storage device; in digital audio, the reproduction of digital signals.*

#### Reason

*Popular music software program from Propellerhead Software. It offers the digital equivalent of hardware synthesizers, samplers, signal processors, sequencers and mixers. Reason works as a virtual music studio, or as a set of virtual musical instruments which can be played live or used with other sequencing software.*

#### Recapping

*Electronic components can age. Certain types of capacitors – namely, electrolytic and tantalum, often used in the power supply section – are the most likely to deteriorate over time; some put the maximum safe life of an electrolytic capacitor to be 25 years. Therefore, serious vintage synth owners “recap” (replace the age-sensitive capacitors in) their older equipment.*

#### Record Head

*A device on an analog tape machine that changes electrical current to magnetic energy; the changes of the magnetism match the waveshape of the audio signal fed to the head.*

#### Record Level

*A control on a tape machine that determines the amount of magnetic flux recorded on the tape, or the DAW control that determines the level of the digital signal recorded to the sound file.*

#### Record Monitor

*On some tape machines, a switch position that allows the VU meter and sound output of the tape machine electronics to monitor the input signal to the tape machine.*

#### Record Ready

*A control state of a multitrack tape recorder where the designated track will begin recording when the record function of the tape recorder is activated.*

#### Recording Bus

*A bus that sends a mix signals from the console channels to the multitrack recorder or DAW. (See also “Bus.”)*

### Recording Session

*A bloc of time in which music is being recorded in the studio.*

### Rectifier

*A circuit that makes sure a voltage stays only positive or negative. In power supplies, it is used to remove the negative component of AC voltage, or to protect you from plugging in module's power connector backwards. As a module, a half-wave rectifier passes only positive voltages and replaces anything negative with 0v; a full-wave rectifier takes any negative voltages and inverts them so they become positive. This effectively doubles the frequency of many simple waveforms, like the triangle and sine.*

### Red Noise

*Also referred to as brown noise, technically it's a type of noise whose power density (spectral loudness) decreases 6 dB per octave with increasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distance. It can also be used a slowly changing random control voltage or modulation signal, instead of as an audio source.*

### Reel

- 1) The hub and flanges onto which analog tape is spooled; recording and playback involves unspooling the tape from one reel and onto another.*
- 2) Sometimes also called "demo reel," a compilation of audio or video that demonstrates the abilities of a musician, audio engineer, actor, or other audio/visual professional. Unlike a demo, which is intended to pitch one or more songs, a reel is a demo intended to promote the abilities of the professional rather than the product itself. The term itself is a holdover from the days when this promotional material was delivered on reels.*

### Reference Level

- 1) A standard baseline level of volume used to measure how much level is present in dB above or below the baseline.*
- 2) See "Operating Level."*

### Reference Tone

*A single-frequency tone (often at 1000 kHz) used to calibrate the levels of sound equipment; the tone used to set reference level. (See also "Test Tones.")*

### Reflected Sound

*Sound that reaches a microphone or listener after one or more reflections from surrounding surfaces.*

### Reflection

*In acoustics, the bouncing of sound waves off of a flat surface, as opposed to absorption. Reflection can have a great impact on how we perceive the collective sound; reflected sounds from a distance is perceived as echo, while reverberation is created from thousands of reflections. (See also "Absorption," "Early Reflection," "Echo," "Reverberation.")*

### Regeneration

*Regeneration can have a couple of different meanings inside a synth, both meaning feedback. An echo unit can feed some of its output back into its input, causing the delayed signal to be repeated*

*again; this is sometimes referred to as regeneration. Also, very rarely you will hear resonance in a filter referred to as regeneration.*

#### Regulated Power Supply

*A device to supply power to electronic equipment whose output voltage will not fluctuate when more equipment is turned on, or if there is a change in voltage of the power line. A regulated power supply is designed to protect sensitive electronics from destructive power surges.*

#### Relay

*An electromagnetically activated switch that connects or disconnects two terminals when a control voltage is applied.*

#### Release Time

*In dynamics signal processors, the time it takes for the output signal to return to original levels when the input signal crosses the designated threshold.*

#### Release

*This refers to the final stage of an envelope that typically falls back to zero volts, usually resulting in silence. It is often used in the context of talking about an Attack/Release (AR) or Attack/Decay/-Sustain/Release (ADSR) envelope generator, but can refer to any final stage of an envelope.*

#### Remote

- 1) A device that controls the functions of another device wirelessly.*
- 2) Describing on-site recording, as opposed to recording in the studio.*

#### Reset

*The Reset input on a module accepts a trigger or gate signal, and tells the module to go back the beginning of whatever it was doing. In the case of a clock divider, this means pretend the next clock is the first clock you should be counting in the division (more on that in the full definition). In the case of a sequencer, it means go back to the first stage. In the case of an envelope, it means go back to the start of the attack. In the case of a gate delay, it means to re-start the timer for the delay.*

#### Residual Magnetization

*The amount of magnetism left in a magnetic material after the magnetizing force is removed. Residual magnetism can accumulate in tape machines over time, either creating distortions and noise in the sound output or partially erasing the tape.*

#### Residual Noise

*The noise level left on recording tape after it has been erased.*

#### Resistance

*The opposition of a substance to the flow of electrical current, measured in ohms.*

#### Resistor

*An electrical component with a specific amount of resistance to electrical current, used within the circuit to regulate the flow of current.*

#### Resonance

- 1) The natural tendency of physical substances to vibrate with more energy at certain frequencies. The*

*principle of resonance is a key element in the design of acoustic instruments; for example, the hollow chamber of a guitar or violin is designed to resonate with the vibrations of the string. Resonance also plays a role the acoustic design of a space, and even in developing good vocal technique to project the voice.*

*When the output of a filter is fed back into its input, the result is an increased boost in the harmonics right around the filter's cutoff or corner frequency. The audible result is similar to playing a sound in a room that has a resonance – sympathetic, reinforcing echo or vibration – at a certain frequency. Therefore, the term resonance is often used to refer to a filter's feedback amount.*

#### Resonant Frequency

*A frequency at which a physical item vibrates naturally.*

#### Resonate

*To vibrate at the resonant frequency. Also refers to the lingering reverberation that causes a sound to continue after the sound source has stopped. This continuing sound is due to the sympathetic resonance of nearby objects.*

#### Resonator

*Many acoustic instruments include a body or sound chamber that “resonates” – sympathetically vibrates at, or reinforces – one or more frequencies. To simulate this effect in modular synths, you can get a specialized filter or equalization module that boosts the sound at typically three or so user-definable frequencies, each usually within a narrow band. This is one of the secrets of synthesizing real-world sounds or spaces.*

#### Reverb (Reverberation)

- 1) Short for “Reverberation.” (See “Reverberation.”)*
- 2) A signal processor or plug-in that creates artificial reverb to a signal.*

#### Reverb Time (RT)

*The time it takes for the reverberation or echoes of a sound source to die out after the direct sound has stopped. Specifically, the reverb time is measured between the point at which the sound source stops and the point at which the reverberation levels fall by 60 dB.*

#### Reverb

*Short for reverberation. This is an effect device that mimics being in a room where you can hear the original sound reflect off the walls multiple times, bouncing around in a wash of sound until it eventually decays into silence. A reverb can greatly enhance the sound of a synthesizer, adding lushness and dimension to what might otherwise be a stark sound. There are relatively few modules that implement a reverb effect, and even fewer that allow you to voltage control some of its parameters (the ErbeVerb being the most famous); many just use an external reverb effect.*

#### Reverberant Field

*Describes the space that is far enough from the sound source that the reverberations are louder than the direct sound.*

#### Reverberation Chamber

*A device built to simulate room reflections.*

### Reverberation Envelope

*The attack, decay, sustain and release of the reverberation volume; or how fast the reverberation reaches peak level and its rate of decay.*

### Reverberation

*The persistence of a sound after the source stops emitting it, caused by many discrete echoes arriving at the ear so closely spaced in time that the ear cannot separate them.*

### RF Interference

*The unwanted noise introduced into electronics, circuits and/or audio systems by the presence of RF signals. RF interference in a system can result in humming, buzzing, static or even the reproduction of radio transmissions.*

### RF Signals

*RF Signals (or RF) – Short for Radio Frequency Signals, electromagnetic waves that carry wireless radio and television signals. The vast majority of RF signals exist at frequencies higher than 100 kHz.*

### Rhythm Section

*The musical instruments in a band or ensemble that are responsible for playing rhythmic parts rather than melody parts. In contemporary music, rhythm sections typically consist of drums and bass, along with some combination of percussion, piano/keyboard and/or guitars.*

### Ribbon Controller

*This is a long strip that is capable of measuring the position where you press it along its length, and the pressure used to press it. It can be used as an alternate keyboard or as a pitch bend controller, with the position determining pitch. Shorter versions also appeared sometimes as alternate controllers on synthesizers, such as the Yamaha CS-80.*

### Ribbon Microphone

*A microphone that converts sound waves to electrical current via a thin conductive ribbon set between magnetic poles. Ribbon microphones are almost always responsive to sound on both sides of the ribbon, creating a bi-directional or figure-8 pattern.*

### Riff

*A short melody repeatedly played in a tune often with variation between vocal lines.*

### Ring Modulator

*Balanced or ring modulation is a special type of amplitude modulation, where one bipolar (swinging both above and below 0 volts) signal – the modulator – is used to vary the amplitude of a second bipolar signal, known as the carrier. The modulator's frequency is both added to and subtracted from the carrier's frequency; the resulting harmonics replace the original carrier and modulator.*

### Ring Out a Room

*The process of identifying and compensating for problem frequencies within a room for the purpose of optimizing live audio within that space. This is typically done by sending pink noise through the speakers, turning up the microphones to the point of feedback, and using EQ to notch out the offending frequencies.*



**Rise Time**

*The rate at which an audio waveform makes a sudden increase to a higher amplitude.*

**RMS Meter**

*A meter that recognizes and responds to the effective average, the RMS level, or the effective average value of an AC waveform, rather than to the peak level. (See also "Root-Mean-Square," "Peak Meter.")*

**Roll Off**

*The reduction of signal level as the frequency of the signal moves away from the cut-off frequency, especially when the cut-off rate is mild.*

**Room Equalization**

*In live audio, an equalizer inserted in the monitor system that attempts to compensate for frequency response changes caused by room acoustics.*

**Room Sound**

*The natural ambience of a room, including the reverberation and background noise.*

**Room Tone**

*The natural background noise occurring in a room without music playing or people speaking. In recording audio for film and TV, on-set sound mixers capture a take of room tone for the purpose of providing continuity between clips of dialogue during post-production.*

**Root Mean Square (RMS)**

*The effective average value of an AC waveform. Used as a measure of the overall level of the sound rather than just measuring by the peaks. (See also "RMS Metering," "Peak Metering.")*

**Rotating Head**

*A circular head with two (or more) gaps that rotates against the direction of tape motion at a slight angle to the tape travel.*

**Rumble**

*A low-frequency noise, typically caused by earth/floor vibration or by uneven surfaces in the drive mechanism of a tape recorder or playback unit.*

**Rythm Tracks**

*The recording of the rhythm instruments in a music production.*

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**S-trig**

*Some systems – such as the original Moog modular – use an s-trigger (switch or shorting trigger) instead of a normal gate, which was a wire that was shorted to 0 volts ground, like the closing of a switch wired to ground. You cannot interconnect these two systems without some form of conversion between the two, which can be as simple as a special cable.*

**S/H**

*A sample and hold (S/H) module has two inputs: a signal that is being sampled, and a trigger input*

*that indicates when the first input should be sampled. When a trigger is received, the current voltage at the first input is sampled (measured) and held (stored), and presented at the output. This stable voltage is held until a new trigger is received. Sample and holds are most often associated with creating stepped random voltages. To do this, noise is fed to the main input; whenever a trigger is received, the voltage present at that input is some random value, which is then dutifully sent to the output.*

#### S/PDIF

*Abbreviation for “Sony/Phillips Digital Interface,” a protocol for sending and receiving digital audio signals using a common RCA connector.*

#### Safety Take (ST)

*An additional take of audio captured for good measure after a take of acceptable quality has been recorded.*

#### Sallen-Key

*The Sallen-Key filter topology or design creates a “second order” or two-pole low, high, or bandpass filter and is capable of high resonance or  $Q$ . This is the design used in the Korg MS-20 filter and its clones, among others.*

#### Sample & Hold

*A sample and hold (S/H) module has two inputs: a signal that is being sampled, and a trigger input that indicates when the first input should be sampled. When a trigger is received, the current voltage at the first input is sampled (measured) and held (stored), and presented at the output. This stable voltage is held until a new trigger is received. Sample and holds are most often associated with creating stepped random voltages. To do this, noise is fed to the main input; whenever a trigger is received, the voltage present at that input is some random value, which is then dutifully sent to the output.*

#### Sample Dump Standard (SDS)

*Sample Dump Standard (SDS) – See “MIDI Sample Dump Standard.”*

#### Sample Rate Conversion

*The conversion of digital audio taken at one sample rate to a different sample rate without first converting the signal to analog.*

#### Sample Rate

*This is a specification of digital audio: How fast the individual measurements (samples) that reconstruct a sound are recorded or played back. The bandwidth of that audio file (which corresponds to the highest frequency that can be reproduced) is in practice a bit less than half of the sample rate. In digital recording, the number of times per second that samples are taken. The higher the sample rate, the more realistic the digital reproduction of the sound, and the higher frequencies of the sound can be reproduced. In digital audio, the quality and resolution of a digitally reproduced sound are described as a combination of sample rate and bitrate. (See also “Bitrate.”)*

#### Sample

*1) In digital recording, the numerical measure of the level of a waveform at a given instant of time. Analog music is represented digitally by many samples taken in rapid succession.*

2) *A short segment of audio recorded for the purpose of reproducing and manipulating the sound digitally.*

### Sampler

*A device that records and plays samples, often with features for editing, manipulating and storing the samples.*

### Saturation

*On a simple level, saturation is a fancy word for clipping: Once the input voltage goes higher (or lower) than a circuit can handle, it is instead held at that limit. However, saturation usually implies a more rounded, shaped approach to that clipping limit, resulting in a more pleasing (or at least less annoying) form of distortion. Tubes circuits are often associated with this soft clipping behavior, although it can be emulated in other circuits or even digital signal processing. Different devices may be sought out for specific sonic character of the way they.*

- 1) *The point at which magnetic tape reaches full magnetization due to an excess of sound level. This creates some distortion that some audiophiles describe as “analog warmth” a desirable quality in certain instances.*
- 2) *The audio distortion that occurs by overdriving a signal through a tube amplifier or preamp—again producing color and warmth in the sound that engineers often find appealing.*
- 3) *A digital plugin that emulates tape or tube saturation.*

### Sawtooth Wave

*A waveform that jumps from a zero value to a peak value and then immediately drops to a zero value for each cycle. (Sometimes also called “Ramp Wave.”)*

### Sawtooth

*One of the most common waveforms produced in a synthesizer. This ramp-shaped wave contains both even and odd harmonics, strongest at the fundamental frequency (the note being played) and diminishing at the higher frequencies. The result is very bright, loud, “brassy” sound.*

### Schmitt Trigger

*This is a type of gate detector that looks at a varying input signal and outputs either a “high” (typically 0, 10, or even 15 volts) signal or a “low” signal (typically 0 volts). When the input goes above one reference threshold – say, 4 volts – the output goes high. When the input then goes back below a second, different threshold – say, 1 volt – then the output goes back low.*

### scope

*This is a piece of test equipment that displays voltage fluctuations as graphical waveforms. A ‘scope can run at a wide range of frequencies, displaying slowly changing voltages like LFOs or envelopes, or quickly changing voltages like oscillators and noise. Oscilloscopes used to be bulky pieces of external equipment, but now you can get USB scopes that offload the display portion of the job to your computer, or scopes as modules.*

### Scratch

- 1) *A descriptive term meaning “temporary”.*
- 2) *A scratch vocal is a vocal done during a basic recording session to help the musicians play their parts. At a later date the final vocal track is overdubbed.*

3) *The action of a musician or disc jockey quickly moving a record back and forth on a turntable reproducing the stylus motion to create a rhythm pattern of sound.*

#### Scrubbing

*The action or function of shuttling a piece of recorded audio back and forth while monitoring it, typically to locate a certain point in the recording. In earlier days, scrubbing was done with reel-to-reel analog tape by manually turning the reels to pull the tape across the playhead. Today, scrubbing is primarily done digitally on a DAW by dragging the cursor back and forth across the waveform.*

#### Second Engineer

*An assistant recording engineer.*

#### SEM

*The Oberheim SEM (Synthesizer Expander Module) was one of their earliest products. It was an entire synthesizer voice – two oscillators, two simple envelopes, VCA, and a very popular two-pole state variable filter design with a knob that crossfaded between low pass, notch, and high pass outputs plus a separate bandpass setting – in a cube-like case. Most often today, when a modular manufacturer uses the magic letters “SEM”, they’re referring to a filter meant to emulate that in the original Oberheim synth.*

#### Semi-modular

*The components of a semi-modular synth – such as the oscillator, filter and amplifier – are pre-wired behind the front panel in what the manufacturer considers to be a typical, logical way. However, they also provide patch points either to access some of its functions (such as the individual waveform outputs of the oscillator) to send to other modules, or to override that pre-wiring. Many who are new to modular synthesis dip their toe in the water by getting a semi-modular synth, and then expanding it with additional modules.*

#### Semitone

*Also known as a half step or half tone, this is the smallest pitch division in most Western music – such as the difference between a C and a C#. With equal temperament (the most common way of tuning a Western scale), this pitch division is 1/12 of an octave.*

#### Send Level

*A control determining the signal level sent to a send bus.*

#### Sensitivity

1) *In audio settings, describes the amount of output that a microphone can produce from a standard level of sound, as compared to the output of another microphone from the same sound level.*

2) *In music, describes the artistic persona in general.*

#### Sequence

1) *A pre-programmed set of musical events, such as pitches, sounding of samples, and rests, to be played in order by a device. Also refers to the action of programming the device to play this set of musical events.*

2) *Loosely referring to a segment of music in general.*

#### Sequencer

*The most common type of sequencer you're going to see in a modular synth contains a row of knobs (also known as steps or stages) that may each be set to output a different voltage. A sequencer then goes through steps one at a time. This is most often used to create repetitive musical lines where each note has the same duration, which is popular in trance-like forms of music as well as the classic Berlin School style (70s-era Tangerine Dream and Klaus Schulze; current Red Shift and Node).*

*A computerized device or software that can be programmed to play a stepped order of musical events, including playing of pitches, sounding of samples, and rests.*

#### Sequential Switch

*This module comes in a few different forms; in the most common, a few different inputs are routed to one output (although they are usually symmetrical – one input can be switched between several outputs). A pulse or gate input then steps through the inputs one at a time, switching which ones is routed to the output. Fancier sequential switches allow you to set the number of stages, to divide an input clock so it switches at a slower tempo than the master clock, or might directly route a series of inputs to corresponding outputs (with usually a summed output as well).*

#### Serial Data

*A digital data stream where individual bits are transmitted one after another over a single connection (as opposed to “parallel data,” in which multiple bits can be sent at once). Most data connections in the recording studio transmit serial data—for example, USB, Firewire and MIDI.*

#### Series Connection

*Connecting devices (especially circuit elements) so that the electrical signal flows from one thing to the next, to the next, etc.*

#### Set Up

*The positioning of microphones, instruments, connections and monitoring in the studio, as well as the controls and levels on consoles, DAWs, etc., in preparation for recording.*

#### Shelf Filter

*A name for the circuit in an equalizer used to obtain the shelf.*

#### Shelf

*A frequency response of an equalization circuit where the boost or cut of frequencies forms a shelf on a frequency response graph. A high-frequency shelf control affects signal levels at the set frequency and all frequencies above it; a low-frequency shelf does the same for signals at and below the set frequency.*

#### Shield

*The outer conductive wrapping around an inner wire or wires in a cable, for the purpose of shielding the cable from picking up external electromagnetic interference.*

#### Shielded Cable

*Cable that has a shield around an inner conductor or inner conductors.*

#### Shock Mount

*An elastic mount on microphone stand that reduces the impact of unwanted vibrations that may affect the stand (for example, floor vibrations from footsteps).*

### Short Circuit

*A direct connection between two points in a circuit that (usually) should not be connected.*

### Short Delay

*Delay times under 20 milliseconds.*

### Shortest Path

*A technique in recording that routes the signal through the least amount of active (amplified) devices during recording.*

### Shotgun Microphone

*A microphone with a long line filter, a tube that acoustically cancels sound arriving from the side, to make the microphone pick up much better in one direction than in any other direction. This gives the shotgun mic a tight, hypercardioid pickup pattern. Shotgun microphones are commonly used to record dialogue in filming situations, usually held on a boom stand with a shock mount.*

### Sibilance

*Energy from a voice centered around 7 kHz, caused by pronouncing “s”, “sh” or “ch” sounds.*

### Sidechain

*An auxiliary input to a signal processor that allows control of the processing to be triggered by an external source. A common use of sidechaining is in compressors, particularly in ducking effects where the presence of a particular audio signal triggers the compression of another audio signal. (See also “Ducking.”)*

### Signal Flow

*1) In the general sense, the path that an audio signal travels from the sound source to the system output. (For example, from the vocalist’s voice into the microphone, through the cables, into the preamp, out of the preamp into the console, through all inserts and buses, and output into the DAW for recording.)*

*2) Signal flow is often specifically meant to refer to the routing of an audio signal through the console, from input to output.*

### Signal Processing

*The practice of altering the character or sound of an audio signal through a variety of devices or plug-ins, such as equalizers, compressors, reverb units, etc.*

### Signal to Noise Ratio (SNR)

*The comparison of the strength of a signal level to the amount of noise emitted by the device, expressed in dB.*

### Signal

*1) In audio, an alternating current (or voltage) matching the waveform of, or being originally obtained from, a sound pressure wave.*

*2) Also in audio, an alternating current (or voltage) between 20 Hz and 20,000 Hz.*

*3) A digital audio bit stream.*

### Sine Wave

*1) In the general sense, the path that an audio signal travels from the sound source to the system*

*output. (For example, from the vocalist's voice into the microphone, through the cables, into the preamp, out of the preamp into the console, through all inserts and buses, and output into the DAW for recording.)*

*2) Signal flow is often specifically meant to refer to the routing of an audio signal through the console, from input to output.*

### Sine

*This is the purest waveform: It contains only the fundamental harmonic, and no higher harmonics. As a result, it's a great wave to use to create a sub bass as well as a kick drum or other pure drum tone; it's also a great source wave to use when exploring techniques such as frequency modulation (FM), amplitude modulation (AM), or wavefolding which add or shift harmonic content.*

### Slap Echo (also called Slapback)

*A single, distinct echo of a sound, which can result naturally from higher frequencies reflecting off a non-absorbent wall, or artificially reproduced by a signal processing unit or plugin. Slap echo creates a "live" sounding effect similar to what you would hear in an arena.*

### Slate

*Slate (Slating) – 1) In video/film, the identification of a scene and take at the beginning of the clip for the purpose of video editing. This is done by presenting the scene/take in written form in front of the camera on a clapboard, calling the scene/take verbally, then marking it audibly with the clapper for the purpose of syncing audio to the video.*

*2) In audio recording, the similar practice of identifying a take of music by an audible cue at the beginning of the recorded track. While some engineers still practice this, it was more necessary in the days of analog tape recording because it helped editors keep track of the location of takes on the recorder. Today, DAWs make it easier to keep track by identifying each take visually on the screen.*

### Slave

*1) In audio, any device which syncs to another device by reading the clock information emitted by the master device.*

*2) In MIDI, any device or instrument that is being operated remotely by MIDI information sent from another device.*

### Slew Limiter

*This function smoothes out an incoming signal so that the change in voltage level cannot exceed a certain number of volts per second. As a result, it is sometimes called a lag generator or processor, or more technically as an integrator.*

### Sliding Rails

*This is a common system for mounting modules into a case where the rails that the modules attach to contain channels rather than holes. A number of nuts are inserted into these channels, which can then be slid to any position to accommodate the mounting hole spacing of your modules. In a Eurorack case, these nuts tend to have a 2.5mm or 3mm hole and corresponding thread.*

### Slope Generator

*A slope generator creates ramps: rising or falling voltages. It is essentially a gate generator and a slew limiter (see above) wired together in the same module. A common example of a slope generator*

*is an attack/decay (AD) or attack/release (AR) envelope generator. However, since it can be used for generalized control voltage functions – even creating a sawtooth or triangle wave oscillator – some companies such as Buchla and Serge referred to by its elemental function of generating sloping voltage changes.*

#### Slope

*Most filters typically have a cutoff or corner frequency they are tuned to. It then reduces (filters) the frequency spectrum of a signal going through it so that its harmonics get progressively quieter the further away they are from this cutoff. The strength of this effect is referred to as its slope. Most filters have slopes that are defined multiples of 6 decibels (dB) weaker for each octave further away you get from the cutoff frequency. For example, a low-pass filter (LPF) with a slope of 24 dB/octave would attenuate harmonics one octave above its cutoff frequency by 24 decibels.*

#### Smart FSK (Frequency-Shift Key)

*Smart FSK – An updated form of Frequency-Shift Key (FSK) sync that enables MIDI devices to sync to analog tape recorders and/or other recording devices. A digital signal with MIDI Song Position Pointer (SPP) data is encoded onto a spare track, which identifies the exact bar, measure and beat for MIDI sequencers/devices at any point in the recording. This enables the device to start playing at exactly the right place and tempo no matter where you start the tape. (See also “Frequency-Shift Key.”)*

#### SMPTE Time Code

*(Abbreviated “SMPTE”) A standardized timing and sync signal protocol created by the Society of Motion Picture and Television Engineers for the purpose of syncing audio to video/film, which can also be used for syncing purposes in audio recording environments. Many audio professionals simply refer to this time code as “SMPTE.”*

#### SMPTE

- 1) Abbreviation for Society of Motion Picture and Television Engineers.
- 2) See “SMPTE Time Code.”

#### Snare

- 1) Abbreviation for “snare drum.”
- 2) The metal strands stretched across the bottom head of a snare drum, which help produce the piercing “cracking” sound when the snare drum is struck.

#### Sock Cymbal

*A rarely used alternate term for “hi-hat,” left over from the days when hi-hat cymbals were placed at “sock level.” (See also “Hi-Hat.”)*

#### Soft Knee

*In compression, refers to the gradual introduction of compression of the signal once the sound level crosses the threshold. (See also “Knee.”)*

#### Software Instrument (Virtual Instrument)

*One of a number of software-based synthesizers, samplers or sound samples that are stored and accessed via computer and performed by an external MIDI controller, rather than in a standalone synthesizer or module. Because of the wide versatility available from these instruments, a growing*



*number of composers and electronic musicians are working with virtual instruments that can be stored in hard drives, rather than purchasing stacks of keyboards and modules.*

### Soldering

*The action of making connections with solder, a soft metal alloy that is used to bond two metal surfaces by melting. In audio settings, soldering is used for a variety of purposes in building, modifying or repairing gear—perhaps most often to repair or build audio cables as a cost-saving effort, as opposed to buying new ones or sending them off for repair.*

### Solid State

*In electronics, refers to the use of transistors and semiconductors (solid materials) in the building of electronic devices, as opposed to tubes. In the recording studio, solid state amplifiers have different properties than tube amps, and each has its own advantages and disadvantages. A more recent application of solid state construction is in computer devices, particularly solid state hard drives (SSD), which transfer data more quickly than conventional spinning disc drives, and are less prone to breakage.*

### Solo Switch

*A switch that activates the solo function on a console or DAW.*

### Solo

- 1) A circuit in a console or DAW that allows one or more selected channels to be heard or to reach the output, while other channels are automatically muted.*
- 2) In music, a segment of a song in which a vocalist or instrument is featured above other instruments.*

### Song Position Pointer (SPP)

*A MIDI message that enables connected MIDI devices to locate a given point in the song. Used in conjunction with MIDI clock as a way of synchronizing devices or telling a connected device when to begin playing.*

### Sound Blanket

*A thick blanket that can be put on floors or hung to add sound absorption to the room, and help prevent sound reflections.*

### Sound Effects (SFX)

*Sounds other than dialogue, narration or music that are added to audio, usually in the context of film/video.*

### Sound File

*A digital audio recording that can be stored in a computer or on a digital storage medium (such as a hard disk).*

### Sound Modeling

*A technique that recreates a sound without directly modeling the physical device. An example is additive synthesis, which uses a combination of sine waves and noise to recreate sounds.*

### Sound Module

*An electronic instrument (tone generator, synth or sampler playback unit) that has no playable interface, but instead responds to incoming MIDI message. Often sound modules were created as*

*the “brains” of popular synthesizers, cheaper versions of the product that could be added to an existing MIDI configuration. Today, sound modules can also occur as software versions or plugins to be accessed on a computer.*

#### Sound Pressure Level (SPL)

*In scientific/technical terms, the measure of the change in air pressure caused by a sound wave, measured in dB. We hear and perceive SPL in terms of amplitude, volume or loudness of the sound.*

#### Sound Pressure Level

*In scientific/technical terms, the measure of the change in air pressure caused by a sound wave, measured in dB. We hear and perceive SPL in terms of amplitude, volume or loudness of the sound.*

#### Sound Source

*The origin of a sound, whose vibrations create sound waves.*

#### Sound Wave

*(Also called “Sound Pressure Wave”) A wave caused by a vibration that results in slight variations in air pressure, which we hear as sound.*

#### Soundtrack

*1) Broadly speaking, refers to any/all audio that accompanies an instance of visual media, whether music, dialogue or SFX.*

*2) In more common terms, refers to the musical score and/or licensed music synced to a film, video, TV program or video game.*

#### Source of Uncertainty

*This was the name for the Buchla 265 and 266 modules that create random control voltages. Its name is often used for random source modules that follow or are inspired by the original Buchla template.*

#### Spaced Pair

*(Also called “A/B Technique”) A stereo microphone placement technique in which two cardioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart from each other (depending on the size of the sound source) to create a left/right stereo image.*

#### Speaker

*A device that converts electrical signals to sound; more technically, a transducer that changes an electrical audio signal into sound pressure waves.*

#### Speed of Sound

*Generally speaking, the time it takes for a sound wave to travel through a medium. Sound travels at different speeds through solids, liquids and gases, and though we usually think of sound as traveling through the air, differences in temperature, air pressure and humidity can also affect how fast sound travels. For a starting frame of reference, the speed of sound is generally defined by aerospace engineers as “Mach 1.0,” translating to 340.29 meters per second (approx. 761.1 mph, or 1116 feet per second), which is how fast sound travels through the air at sea level at a temperature of 15 degrees Celsius (59 degrees Fahrenheit). By contrast, at 70 degrees Fahrenheit under standard atmospheric conditions, the speed of sound is about 344 m/s, or 770 mph.*

### Splicing

*Historically, the act of attaching previously cut pieces of audio tape or film in precise locations by applying a special kind of adhesive tape on the back. This is/was done for the purpose of shortening sections of audio or editing film. Today, splicing has become a very simple process by editing sections of audio or video digitally with a DAW or film editing software.*

### Splitter

*The short definition is something that can divide a signal into two or more copies, such as a splitter cable where two outputs are wired to one input. For a deeper discussion, see the entry on multiple, as there are ways of going about this beyond simple wiring.*

### Spread

*A few oscillator modules can produce more than one tone at the same time. Slightly detuning or “spreading” these tones from each other creates an often pleasing chorusing sound. Depending on the module, you might even be able to spread these tones to form intervals, triads, and chords.*

### Spring Reverb

*A device that simulates reverberation by creating vibrations within a metal spring by attaching it to a transducer and sending the audio signal through it. A pickup at the other end converts those vibrations into an electrical signal which is mixed with the original audio signal. While the physical spring reverbs still exist, most studios emulate spring reverb with the use of plug-ins or hardware reverb units.*

### Square wave

*This is a common waveform produced by a synthesizer’s oscillator. It alternates between a high and low voltage (typically +/-5 or 8 volts for an audio oscillator; sometimes low frequency oscillators go between 0v and a positive voltage). Aside from being a really easy waveshape to generate with analog circuitry, it has an interesting harmonic series: it has a strong fundamental, then gradually weaker odd harmonics: a component at three times the fundamental frequency, one at five times the fundamental, and so forth. The result is a more open, hollow sound, especially when compared to a sawtooth (ramp) wave that has both odd and even harmonics present.*

*A wave shape in which the voltage rises instantly to one level, stays at that level for a time, instantly falls to another level and stays at that level, and finally instantly rises to its original level to complete the wave cycle.*

### Stackable Cable

*Many banana style cables are constructed that each plug has a jack built into its back, allowing you to plug another cable directly in top of the original plug. These are used by Buchla and Serge-compatible systems. TipTop makes a similar cable using 3.5mm plugs and jacks for Eurorack format users called Stackables.*

### Stage Monitor

*A speaker on the stage that enables performers to hear themselves and to hear what the other musicians are playing on stage.*

### Stage

*1) The partially enclosed or raised area where live musicians perform.*

2) *In reverberation effects devices, an echo added before the reverberation to simulate echoes that would come from a concert stage.*

*In the most general terms, a stage is the next change in voltage among a series of changes. In an 8-step sequencer, for example, each new note that it produces in order is a stage. In an envelope generator such as an ADSR (Attack/Decay/Sustain/Release), each phase – such as attack, where the envelope generally rises from 0 volts to the highest voltage it can output – is a stage. You might also hear it used to describe the number of sample stages in a BBD (Bucket Brigade Delay), described elsewhere.*

#### Standard Operating Level

*A reference voltage level or maximum average level that should not be exceeded in normal operation.*

#### Standing Wave

*An unwanted sound wave pattern that often occurs when the sound wave bounces between two reflective parallel surfaces in a room, and the reflected waves interfere with the initial wave coming from the sound source, in which the combined wavelength of the affected frequency is effectively the length of the room. This creates the audible illusion that the wave is standing still, so the frequency is amplified to an unwanted level in certain parts of the room while nearly absent in others. Standing waves are most common in square or rectangular rooms with parallel surfaces, so acoustic designers try to prevent these waves by installing absorptive materials or introducing other items to offset the parallel surfaces.*

#### Step Mode

*A setting in a sequencer or DAW in which notes are input manually, one note or step at a time.*

#### Step Sequencer

*This usually refers to a type of sequencer where you step to and pause on a stage, enter the note (and possibly the duration) for that stage, move on to the next step, and so forth.*

#### Step

*Step is often used interchangeably with stage (see above), especially when talking about sequencers.*

#### Stereo Image

*The audible perception of stereo, in which different sound sources appear to be coming from far left, far right or any place in between.*

#### Stereo Micing

*Placement of two or more mics so that their outputs combine to create a stereo image.*

#### Stereo

*A recording or reproduction of at least two channels where positioning of instrument sounds left to right can be perceived.*

#### Strike

*This term appears on several Make Noise modules, although it has been creeping into the general lingo. Some filters, amplifiers, and low pass gates (LPGs) that use or simulate vactrols (a light sensitive resistor placed next to a light source such as an LED, allowing a voltage to be turned into a resistance to control a parameter) may have a strike input. When you flash an LED at a light sensitive resistor,*

*it does not change the resistance instantaneously and stay there – instead, there is some delay as it glides to the desired resistance. When you turn the LED off, the resistance may not go instantaneously to full; instead it might take a brief moment to decay. These characteristics are useful for creating percussive sounds and attacks. The purpose of a strike input is either to pass just a short pulse, or to allow you to re-attack while the LED is otherwise still on.*

*To put away equipment and clean up after a recording session.*

#### Subcode

*Additional information bits that are recorded alongside digital audio, used for control and playback purposes.*

#### Subframe

*A unit smaller than one frame in SMPTE time code.*

#### Subgroup

*A number of input channels on a console that can be controlled and adjusted as a single set before sending the combined signal to the master output. Sometimes also called “Submix,” “Bus” or just “Group.”*

#### Subharmonic

*A circuit that divides the fundamental harmonic of the incoming sound to produce lower frequencies, and therefore subharmonics. The most common is an octave divider or sub bass circuit that divides creates a subharmonic by dividing the fundamental by 2 (some can also create a subharmonic two octaves below the fundamental by dividing it by 4).*

#### Submaster / Sub-Master

*The fader which controls the combined level of sound from several channels during mixdown or recording.*

#### Submix

*Submix – See “Subgroup.”*

#### Suboctave

*A module that creates a new tone one or two octaves below the fundamental harmonic – the “pitch” – of the sound coming into it, to emphasize the bass. (Subharmonics are discussed in detail elsewhere in this glossary.) This tone is usually a square wave, although some clever modules may create something more sine-like, or that more closely resembles the original waveform.*

#### Subtractive Synthesis

*The most common synthesis technique: You start with one or more oscillators outputting waveforms with a large number of harmonics, and then pass this mix through a filter that removes some of the harmonics to create the desired sound or timbre. This modified tone is then sent to an amplifier that adds articulation to the note by varying its loudness.*

*An old-school method of sound synthesis in which sounds are designed and created by generating harmonically rich waveforms, then filtering out unwanted harmonics to arrive at the desired sound.*

#### Sum

*To sum is a fancy way of saying you added two (or more) things together; the sum is the result. It*

*usually is used in the context of adding together control voltages, although it can also be used for audio or even mixes of harmonics. The opposite is difference, which subtracts one input from another. A signal that is the mix of the two stereo channels at equal level and in phase.*

#### Summing

*The process of blending two or more signals into one mixed signal. In summing audio, each successive channel adds volume to the overall signal, so channels must be mixed in order to prevent peaking the combined signal.*

#### Super-Cardioid Pattern

*A very tight cardioid microphone pattern with maximum sensitivity on axis and the least amount of sensitivity approximately 150 degrees off-axis.*

#### Surround Sound

*A technique of recording and playback in which the listener hears various aspects of the sound from front to back as well as side-to-side—a 360-degree audio image, as opposed to the standard stereo left-right image. Surround sound can occur in various formats with different numbers of speakers arrayed through the room. Surround sound today is most commonly used in film and TV production.*

#### Sustain

*This is a common stage of an envelope generator where a voltage – usually being sent to a filter's cutoff frequency or an amplifier's level – is being held a steady level while a note is still being held down. The knowledge that a note is being held is usually provided by a gate signal, that stays high as long as a note is held down, although some envelope generators may have a dedicated time control for how long the sustain stage should last. Envelopes that contain sustain stages include the ADSR (Attack/Decay/Sustain/Release) and AR (Attack/Release, which usually assumes a sustain stage).*

#### SVF

*A state variable filter (SVF) is a common design for synth filters. This design lends itself to allowing low pass, high pass, and bandpass all being available simultaneously. Another side effect is that they are not prone to oscillating at high feedback (resonance) settings, although some have certainly figured out how to make this happen. The Oberheim SEM (Synthesizer Expander Module) filter is perhaps the most famous state variable design.*

#### Sweetening

*A vague term referring to the fine-tuning of audio in the post-production stage of recording. Effectively, any small “tweaks” to to make the audio sound better is considered sweetening.*

#### Switch Trigger

*Some systems – such as the original Moog modular – use an s-trigger (switch or shorting trigger) instead of a normal gate, which was a wire that was shorted to 0 volts ground, like the closing of a switch wired to ground. You cannot interconnect these two systems without some form of conversion between the two, which can be as simple as a special cable.*

#### Switch

*A device that makes and/or breaks electrical connections.*

#### Switchable Pattern Microphone

*A microphone having the capability of two or more pickup patterns, which can be toggled by use of a switch on the microphone.*

### Switching Power Supply

*A switching power supply starts by directly converting the incoming high-voltage AC signal into a high-voltage DC signal. They then rapidly switch that output on and off to average a lower output voltage. This switched voltage is then smoothed out to create a constant DC supply at the desired voltage. Switching power supplies tend to be lighter, cooler, and less expensive, at the cost of often higher noise – both in the output voltage, and in radio frequencies (this is why they are often surrounded by a shielding cage). Many are moving to a hybrid power supply that combines a switcher with a small linear supply or regulator to get the best of both worlds.*

### Sync Pop

*A short tone (usually a sine wave at 1 kHz, and the length of a frame of film) that is placed exactly two seconds before the start of a piece of film or music. The sync pop is used to make sure that all related audio and video tracks stay in sync with each other through all stages of post-production.*

### Sync24

*Sync24 is an alternate name used for the Roland-created standard DIN Sync, which sends a clock signal at the rate of 24 pulses per quarter note at the current tempo. Korg equipment used a variation of this running at 48 pulses per quarter note, also known as Sync48.*

### Sync

*Sync can have two different meanings, depending on whether we're talking about oscillators or about clock signals. Some oscillators support a mode where they reset their waveshapes to the beginning when they receive a signal from another oscillator. If there is not a precise octave relationship between the two oscillators, the result is a modified waveform that has been reset prematurely, following the frequency of the second oscillator. You can create some very cool "ripping" sounds by modulating the frequency of the slave oscillator; a simple AD envelope works well. In the context of timing, when you are synchronizing sequencers or drum patterns, it is common to send a master timing or sync signal around the modular for all the relevant modules to follow. This is typically a gate or trigger signal.*

*Short for "Synchronization." In audio/studio settings, sync refers to the correlating of two or more pieces of audio or video in relation to each other. This can include syncing two recording/playback devices timed to a sync signal like SMPTE Time Code, synchronizing audio with video in film or TV, and many other examples. Licensing a song or piece of music for placement in film, TV or video is also referred to as "syncing."*

### Synthesizer Expander Module

*The Oberheim SEM (Synthesizer Expander Module) was one of their earliest products. It was an entire synthesizer voice – two oscillators, two simple envelopes, VCA, and a very popular two-pole state variable filter design with a knob that crossfaded between low pass, notch, and high pass outputs plus a separate bandpass setting – in a cube-like case. Most often today, when a modular manufacturer uses the magic letters "SEM", they're referring to a filter meant to emulate that in the original Oberheim synth.*

### Synthesizer

*A musical instrument that uses electrical oscillators to generate tones artificially, either to simulate the sounds of other instruments or to create other sounds not possible with other instruments.*

### System Exclusive

*System Exclusive (SysEx) – A MIDI message that will only be recognized by a unit of a particular manufacturer.*

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### Tach/Tachometer

*In analog tape recording, a device on the recorder that measures and regulates tape speed by emitting pulses as the tape moves across the head.*

### Tails Out

*A method of winding audio tape so that the end of the last recorded selection is at the outside of the reel.*

### Take Notation

*Writing down the takes of the tune being recorded on a take sheet or on the track log with comments. Take notation was/is recommended for analog tape recording, but in most studios, this function is now accomplished on the DAW.*

### Take

*The recording that is done between one start and stop of a tape recorder or DAW.*

### Talk Box

*An effects unit that enables a musician to modulate the sound of his/her instrument via a tube placed into the mouth. Historically, talk boxes have been used as an effect for guitars, but they can be used to modify other instruments, as well.*

### Talkback

*A microphone in the control room carried on a separate circuit from the recorded channels, allowing the engineer to communicate with the musicians in the live room or sound booths through the monitoring system.*

### Tape Delay

*A signal processing technique for creating artificial delay or echoes by manipulating time delays with analog tape machines. This technique began by routing the signal to a separate tape recorder and mixing the delayed response back in with the signal; it then evolved to the use of dedicated machines that could adjust the length of the delay by adjusting the distance between the record and playback heads. Today, most tape delay effects in the studio are simulated digitally through plug-ins in a DAW.*

### Tape Guide

*Any stationary or rotating device which directs the tape past the heads on a tape machine, or from one reel to the other.*



### Tape Hiss

*The natural high-frequency noise that occurs on analog tape due to the magnetic particles from which the tape is made. Tape hiss constitutes most of the noise floor that occurs in analog recording, and can be reduced by using tape constructed of finer magnetic particles. (See also “Noise Floor.”)*

### Tape Loop

*A length of tape with the ends spliced together so that the recording will play continuously.*

### Tape Recording Equalization

*The increase in amplitude of signals, in a tape machine’s electronics, at the high frequencies as a tape is recorded to keep high-frequency signals recorded above the tape hiss.*

### Telephone Filter

*A filter used to simulate the audio heard through a telephone receiver by removing signals at frequencies below 300 Hz and above 3500 Hz.*

### Tempo Mapping

*The act of programming a sequencer or DAW to follow the tempo variations of a recorded performance. Unlike beat mapping or beatmatching, both of which effectively adjust the recording to fit a set tempo, tempo mapping adjusts the tempo of the project (especially the MIDI instruments) to match the natural tempo nuances of the recorded material. (See also “Beat Mapping,” “Beatmatching.”)*

### Tempo

*The rate at which the music moves, measured in Beats Per Minute (BPM).*

### Terminal

- 1) A point of connection between two wires, including the plug on the end of a cable, and the jack on a piece of equipment.*
- 2) Refers to the keyboard and monitor of a computer that enable the user to enter information and to access data.*

### Test Oscillator

*A device that generates audio waveforms at various frequencies for testing purposes.*

### Test Pressing

*One of a few initial vinyl record copies pressed from the first stamper made, which is listened to and visually inspected to approve the quality before more copies are pressed.*

### Test Tones

- 1) A recording of several single-frequency tones at the beginning of a tape reel at the magnetic reference level that will be used to record the program.*
- 2) Artificially generated tones that are used to calibrate an audio system.*

### Thin Sound

*A vague term describing an audio signal that is lacking in certain frequencies, especially on the low end. Over-filtering a signal with an EQ can produce a thin sound, for example.*

### Threaded Inserts

*A common system for mounting modules into a case is called sliding rails or nuts. A number of nuts*

*are inserted into these channels, which can then be slid to any position to accommodate the mounting whole spacing of your modules. Some don't like this system, so they replace the nuts with strip of metal inserted into the channel that have been pre-drilled for the standard Eurorack mounting hole spacing. They may be drilled for 2.5 or 3 mm screws; pay attention when buying the rails or a case that has them pre-installed.*

#### Three-To-One Rule

*A principle of microphone placement that says when multiple mics are used at once, the distance between microphones should be at least three times the distance between each microphone and its respective sound source. The three-to-one rule is used to prevent phasing issues between the audio signals.*

#### Three-Way Speaker

*A speaker system that has separate speakers to reproduce the bass, mid-range and treble frequencies.*

#### Threshold of Hearing

*Described as the sound pressure level at which people can hear only 50 percent of the time.*

#### Threshold

*A threshold is generally a voltage level a signal needs to cross before a module takes an action. For example, when the output of an envelope follower (a module that creates a voltage that corresponds to the current level of an audio signal) rises above a threshold level, then its gate signal will go high indicating a note has started. When the output of the envelope follower falls before a threshold (which may be the same or different than the note-on threshold), then the gate goes low, indicating the note should be finishing.*

*The level at which a dynamics processing unit will begin to change the gain of the incoming signal.*

#### Throat

*In a speaker, the small opening in a horn or in a driver through which the sound wave passes from the driver to the horn.*

#### Through-Zero Frequency Modulation

*TZFM is the abbreviation for Through-Zero Frequency Modulation. Think of a patch where you feed the output of one oscillator (the modulator) into the frequency control voltage input of a second oscillator (the carrier). As the waveform output of the modulator rises above zero volts, it is added to the normal pitch control voltage for the carrier, and the pitch of the carrier goes up. As the waveform output of the modulator goes below zero, it is subtracted from the normal pitch control voltage, and the pitch goes down. But what happens if the result of subtracting the modulator from the pitch control goes below zero volts? In an oscillator that explicitly says it implements through-zero frequency modulation, the carrier will start playing backwards – in essence, a negative frequency. This generally produces a more pleasing result, and is a desirable characteristic for an oscillator.*

#### Throw

*1) In speakers and in microphones, describes the amount of unrestricted movement that the diaphragm can make. In microphone, this affects the mic's sensitivity; in speakers, it affects the distance of sound projection. (A speaker designed for smaller spaces has a "short throw," while one designed for a farther projection has a "long throw.")*

2) In speakers, “throw” may also be used to describe the speaker’s directional output, often based on the frequencies it emits. A horn, for example, emits high frequencies in a limited angle of direction, so it has a “long throw,” while a subwoofer emits low frequencies in all directions and has a “short throw.”

3) Something a producer, engineer or musician might do with whatever is in his/her hand during a moment of intense frustration.

#### Tie Lines

*Tie Lines – Cables with connectors at both ends, which are usually run through walls or floors in the studio, for the purpose of sending signals between rooms. Tie lines provide a great semi-permanent way to route and configure signal paths quickly through various parts of the studio and help the engineer keep track of signal flow.*

#### Timbre

*This word is often used to describe the unique tonal characteristic of a sound you are creating, separate from its pitch or loudness. Different sounds, by definition, have different timbres. When you change a parameter of a sound that changes its tonal characteristic – such as changing the filter cutoff, pulse width, amount of wavefolding, etc. – you are changing its timbre. The timbre often changes during life of a note.*

*The sound quality that makes one instrument sound different from other instruments, even while playing the same pitch. The timbre of a trumpet, for example, is what makes it sound like a trumpet and not like a flute. Timbre is largely shaped through the presence, absence and complexity of harmonics when the instrument is played.*

#### Time Code

*A standardized timing signal used to help devices sync with one another, or to sync audio to video. Common time codes used in the studio are MIDI Time Code (MTC) and SMPTE time code.*

#### Time Compression / Expansion

*(Also called “Time Stretching” or “Time Shifting”) The process of speeding up or slowing down an audio recording without changing the pitch of the sounds.*

#### Time Constant

*A complex mathematical idea that basically describes the time delay between when an electrical voltage is applied to a circuit and when the circuit responds to it.*

#### Tini-Jax

*This is a special design of jack made by Switchcraft that is used by Buchla (and many of their clones) to carry audio signals. They are 3.5mm in diameter, but differ slightly physically from a common 3.5 mm jack. 1/8” plugs would be loose in when plugged into a Tini-Jax jack; a Tini-Jax plug might not fit into or might even damage a 1/8” jack.*

#### Toms

*The small drums (as little as 10 inch diameter) that mount on racks above the kick drum and the large drums in a drum set.*

#### Tone Generator

1) A device that puts out test tones at various frequencies to align a tape machine or for other testing

*purposes.*

2) *The circuits in a synthesizer that create the audio signals put out by the unit, usually to emulate the sound of another instrument.*

Tone

1) *Any single-frequency signal or sound.*

2) *The sound quality of an instrument's sound relative to the amount of energy present at different frequencies.*

Tonguing

*The technique of controlling the start of a note in a brass or woodwind instrument with the tongue.*

Total Harmonic Distortion (THD)

*The measure of the difference between the level of harmonic frequencies at the output stage of an amplifier as compared with the input stage, a ratio expressed as a percentage. It's a fine-tuning specification barely perceptible to many ears, but the lower the THD, the more accurately the amplifier/speaker is reproducing the sound.*

Touch Sensitive

*Touch Sensitive – See “Velocity Sensitive.”*

Track & Hold

*This is a variation of a Sample & Hold. Both have two inputs – a gate signal, and a voltage reference signal – and a voltage output. When a Sample & Hold receives a gate high signal, it freezes and outputs the voltage reference coming into the reference input. This voltage is maintained until a new gate high signal; gate low signals are ignored. With a Track & Hold, when the gate is high, the reference input it passed along to the voltage output (this is the “tracking” phase); when the gate goes low, the input voltage at that instant is frozen and maintained at the voltage output until a new gate high signal is received.*

Track Log / Track Assignment Sheet

*Track Log/Track Assignment Sheet – A sheet of paper kept with a multitrack tape which tells which instrument was recorded on each track.*

Track

1) *One audio recording made on a portion of the width of a multitrack tape, or created as a digital representation using a DAW.*

2) *One set of control commands in a sequencer or DAW that is used to control one instrument over one MIDI channel.* 3) See “Band Track.”

Tracking

*Tracking usually refers to how well an oscillator follows the pitch control voltage (CV) sent to it. As the voltage rises, the oscillator “tracks” it and produces a higher pitch. Most (but not all!) synths follow a 1 volt per octave system where a rise of 1.00 volts on the pitch input should produce exactly a doubling (one octave rise) in the oscillator's pitch. If this is indeed what happens, the oscillator has good tracking. If the oscillator goes slightly out of tune, it is considered a tracking error, or to have poor tracking. Sometimes you will find voltage-controlled filters have a “tracking” switch for a CV input where the pitch of the filter's corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding*

*change of the pitch input. This can prevent high notes from sounding too bright without the bass notes sounding too dull. Sometimes you will find voltage-controlled filters have a “tracking” switch for a CV input where the pitch of the filter’s corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding change of the pitch input.*

*The act of recording the individual tracks of a multitrack recording.*

#### Transducer

*A device that converts energy from one medium to another. Transducers are prevalent throughout the equipment in a recording studio.*

#### Transient

*The initial high-energy peak at the beginning of a waveform, such as one caused by the percussive action of a pick or hammer hitting a string, or the strike of a drum.*

#### Transistor Ladder Filter

*This term is often used to describe the design of the much-loved Moog low-pass filter, which is still held up by many as being the gold standard in low pass filter sound. Moog actually received a patent for this design (it has since expired); many of their competitors either outright copied it or did their best to emulate it.*

#### Transport

- 1) The portion of a tape machine that moves the tape from the supply reel, past the heads, to the take-up reel.*
- 2) The set of controls found on a DAW or sequencer for starting, stopping pausing, fast-forward and rewind, emulating the functions of a tape machine transport.*

#### Transpose

*In the simplest terms, to transpose the pitch of a musical line is to shift it up or down by a fixed number of semitones or octaves. This is sometimes referred to as “chromatic” transposition. A more sophisticated variation is “scalar” transposition where each note is shifted by a set number of scale steps; this differs from chromatic transposition because some scales may have differing numbers of semitones between steps than other scales.*

*To shift a set of musical notes by a fixed interval. This can happen in a number of ways—for example: 1) by rewriting an entire piece of music in a new key; 2) by shifting the tuning of an instrument so that it plays at a lower or higher interval than the note played (either artificially, as with an electronic keyboard, or by the natural tuning of a transposed instrument, like a trumpet); or 3) Transposing on-the-fly, playing at a set interval above or below what is written (also known as transposing by sight).*

#### Trap

- 1) A filter designed to reject audio signals at certain frequencies.*
- 2) An object designed with acoustically absorptive material, placed into walls to reduce low frequency reflections in the room (also called “bass trap”).*
- 3) Another word for a drum set (as in “trap set”).*

#### Tremolo

*This is the effect of varying the amplitude (loudness) of a note. A way to create this effect on a*

*modular synth is to patch a low frequency oscillator (LFO) to one of the control voltage inputs on an amplifier. Tremolo is different than vibrato; the latter is a warbling in pitch rather than loudness. A wavering or “shaking” musical effect, created either by quick reiterations of the notes (as in a violin tremolo) or by rapid shifts in amplitude.*

### Triangle

*The triangle is a common synthesizer waveform. When selected for the output of an oscillator, it was a more mellow sound than the standard square or sawtooth waves, with fewer and weaker higher harmonics. It is also a popular output for low frequency oscillators (LFOs), as it produces a relatively smooth up and down variation in whatever it controls, while being easier to create than the even smoother sine wave.*

### Triangular Wave

*A harmonically rich waveform that appears triangular in shape when depicted graphically, due to a combination of the presence of odd harmonics and rapid rolloff.*

### Trigger

*A trigger is a very short electrical pulse signal, rising from 0 volts to a standard level such as 5 or 10 volts for a few milliseconds before falling back to 0 volts. It is often used to start or “trigger” the playback of a percussion sound, including starting an envelope generator. They can also be used to pass clock signals around a synth so connected modules all know when a note (or finer subdivision of a note) starts. A trigger usually has a fixed duration, compared to a gate signal which also rises from 0 volts to a higher voltage and falls back to zero again, but which stays “high” a variable length of time depending on the length of a note.*

*The signal or the action of sending a signal to control the start of an event.*

### Trim / Trim Control

*A device that reduces or increases the signal strength in an amplifier, often over a restricted range. Often used interchangeably with gain, but usually referring to fine-tuning signal strength, rather than merely amplifying it.*

### Truncation

- 1) *The shortening of an audio signal, sample or song, typically by cutting off the end.*
- 2) *The dropping of bits of data when the bit resolution is reduced (for example, from 24-bit to 16-bit), causing digital distortion unless dithering is applied.*

### Tune

*The act of adjusting the pitch of a synthesizer’s oscillator (the main pitch-generating element) to match another oscillator, instrument, or reference is known as tuning it.*

### Tuning Fork

*A metal fork with two prongs that vibrate with a fairly pure tone of one frequency when the fork is struck.*

### Turntable

*A device to support and rotate a phonograph record during playback.*

### Tweeter

*A speaker designed to reproduce only the higher frequencies of the sound.*

#### Two Quadrant Multiplier

\*A two-quadrant multiplier performs a simple version of amplitude modulation (AM), where that varies the amplitude or loudness of one signal known as the carrier (typically an audio signal, swinging both above and below 0 volts) with a second signal called the modulator. In the typical amplitude modulation (AM) scenario, a low frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into the control input of a voltage controlled amplifier to add vibrato to an audio signal passing through it. Any negative swings in the modulation signal are ignored; when patching tremolo, you may need to make sure an offset voltage is being added to your LFO to make sure the sound doesn't cut out on the lower excursions of the LFO's waveform. (The case where the modulator's negative as well as positive excursions are used is referred to as a four quadrant multiplier.) \*

#### Two-Way Speaker

*A speaker system with separate speakers to reproduce the lower frequencies (woofer) and the higher frequencies (tweeter).*

#### TZFM

*TZFM is the abbreviation for Through-Zero Frequency Modulation. Think of a patch where you feed the output of one oscillator (the modulator) into the frequency control voltage input of a second oscillator (the carrier). As the waveform output of the modulator rises above zero volts, it is added to the normal pitch control voltage for the carrier, and the pitch of the carrier goes up. As the waveform output of the modulator goes below zero, it is subtracted from the normal pitch control voltage, and the pitch goes down. But what happens if the result of subtracting the modulator from the pitch control goes below zero volts? In an oscillator that explicitly says it implements through-zero frequency modulation, the carrier will start playing backwards – in essence, a negative frequency. This generally produces a more pleasing result, and is a desirable characteristic for an oscillator.*

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#### U

*Rack-mounted equipment usually follows a standard set of dimensions, including 19" (48.3 cm) for width, and a "rack unit" (or U for short) for height equaling 1.75" (4.4 cm) per U. Many common modular synthesizer formats follow the rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25" or 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometimes referred to as MU for Moog Unit).*

#### Unbalanced Audio

*Most audio signals are passed around on cables with two wires: one for the voltage that represents the audio vibrations, and one for ground. This arrangement is often referred to as unbalanced audio.*

#### Unbalanced Cable

*A cable with two conductors (a signal wire and a ground wire) and connectors on each end. Unbalanced cables are often susceptible to electromagnetic interference and noise. Examples of unbalanced cables are guitar/instrument cables (also called tip-sleeve or TS cables) and RCA cables.*

### Unidirectional Pattern

*A microphone pick-up pattern which is more sensitive to sound arriving from one direction than from any other.*

### Unipolar

*Many voltages in a modular synth – including the output of an audio oscillator, and most low frequency oscillators – fluctuates between positive and negative voltages. This is known as a bipolar voltage. Some voltages – such as the output of an envelope generator – only vary between 0 volts and some maximum positive voltage; this is referred to as unipolar.*

### Unison

*Several performers, instruments or sound sources that are sounding at the same time and with the same pitch.*

### Unity Gain

*The scenario in which there is no increase or decrease in signal strength at the output of an amplifier or device compared to the signal strength at the input (typically described as 0 dB).*

### Unity

*Usually used in the phrase “unity gain” this mean a signal keeps the exact same level from input to output.*

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### Vacuum Tube

*A diode, a glass tube with the gases removed, through which electrical current can flow. In audio, vacuum tubes are used in amplifiers, oscillators, and other analog devices.*

### Vamp and Fade

*A method of ending the recording of a song where the music has a repeating part and the engineer reduces volume until the music fades out.*

### Vamp

*A part of a song or chord progression that is repeated, usually at the end of the song, and usually the chorus or part of the chorus.*

### Vari-Speed

*A control on a tape machine that changes the play speed.*

### Variable-D

*A trademarked, patented technology of ElectroVoice in its microphone designs to vary the proximity effect in its microphones. Variable-D places several ports along the microphone body, each of which has a reduced level of sensitivity to higher frequencies the further they are placed from the microphone’s diaphragm.*

### VCA Automation

*A system of mix automation in some mixing consoles in which sound levels or other functions are altered through the use of voltage controlled amplifiers.*



### VCA Group

*Several VCA faders that are fed control voltages from a group master slide. A feature in higher-end mixing boards that enables the engineer to control groupings of independent signals by a single fader that uses VCA to adjust the voltage sent to each channel.*

### Velocity Message

*In synthesizers and keyboard controllers, a MIDI message that transmits data on how hard the key was struck. Velocity messages can be used to transmit volume information, as well as triggering different samples on a multi-sampled instrument patch.*

### Velocity Microphone / Pressure-Gradient Microphone

*Velocity Microphone – See “Pressure-Gradient Microphone.”*

### Velocity Sensitive

*(Also called “Touch Sensitive”) A feature on a MIDI instrument such as a keyboard that transmits a MIDI velocity message depending on how hard the key is struck.*

### Vibrato

*A smooth and repeated changing of the pitch up and down from the regular musical pitch, often done by singers or performed by string and wind players.*

### Virtual Instrument

*(Also called Software Instrument) One of a number of software-based synthesizers, samplers or sound samples that are stored and accessed via computer and performed by an external MIDI controller, rather than in a standalone synthesizer or module. Because of the wide versatility available from these instruments, a growing number of composers and electronic musicians are working with virtual instruments that can be stored in hard drives, rather than purchasing stacks of keyboards and modules.*

### Vocal Booth

*A room in the recording studio that is used for recording vocals in isolation. This practice prevents bleed-through of the sounds of other instruments into the vocal microphone, and also reduces natural ambience and reverberation in the vocal recording.*

### Vocoder

*An audio processing device effects device or plug-in that analyzes the characteristics of an audio signal and uses them to affect another synthesized signal. Primarily developed for the purpose of producing synthesized voice effects from human speech, a vocoder creates the characteristic robotic vocal sound or the “human synthesizer” effect that makes it sound like the synth is speaking or singing words.*

### Voice Over

*The recording of vocal announcements or narration over a bed of music in video, film or commercials.*

### Voice

*1) Besides the obvious definition of the sound humans make from their mouths...in synthesizers, a voice refers to one of a number of sounds/pitches that may be played at the same time. “Monophonic” means only one voice plays at a time, while “polyphonic” means multiple voices can sound at once.*

*(See also “Polyphonic”, “Monophonic.”)*

*2) In some synthesizers, like Yamaha, “voice” may also refer to a specific sound patch available on the synth.*

#### **Volatile Memory**

*Computer memory whose data will be lost when the computer is turned off. RAM (Random Access Memory) is the most common form of volatile memory.*

#### **Voltage Controlled Amplifier (VCA)**

*An amplifier whose gain level is affected by an external voltage being sent to it. VCAs are commonly used in synthesizers, signal processors, and as a means of automation for some mixing consoles.*

#### **Voltage Controlled Filter**

*A filter (especially a low-pass filter) that will change its cutoff frequency according to a control voltage fed to its control input.*

#### **Voltage Controlled Oscillator (VCO)**

*An oscillator whose frequencies are modified by voltage input. Most commonly found in synthesizers.*

#### **Voltage**

*The difference in electrical force or pressure (“potential”) between two objects, causing a flow of electric current between them.*

#### **Volume Unit (VU)**

*A unit to measure perceived loudness changes in audio. The unit is basically the decibel change of the average level as read by a VU Meter. (See also “VU Meter.”)*

#### **Volume**

*A common, non-technical term that either refers to sound pressure level (which we hear as loudness), or to audio voltage level.*

#### **Vox**

*A Latin word meaning “voice,” often used as an abbreviation for track logs in the studio.*

#### **VU Meter**

*A meter that reads audio voltage levels in or out of a piece of equipment and is designed to match the ear’s response to sudden changes in level.*

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#### **Watt**

*Unit of electrical power.*

#### **Wave**

*This is the pattern of vibrations – up and down fluctuations in voltage – output by an oscillator. Different patterns generate different sounds.*

#### **Wavefolder**

*A wavefolder is a very specific design of waveshaper that uses a comparator and some other circuitry.*

*What they do is look to see if the wave goes above (or below) a specific threshold. When it does, instead of clipping off the top and bottom of the wave, they create a mirror image of it and reflect that portion of the wave back upon itself, creating more high harmonics and interesting spectra in the process.*

#### Waveform

*This is the pattern of vibrations – up and down fluctuations in voltage – output by an oscillator. Different patterns generate different sounds.*

*A visual representation or graphic of a sound wave, audio signal or other type of wave, showing the wave's oscillations above and below the zero line.*

#### Wavelength

*The physical length of one cycle of a wave, measured in feet, inches, etc. The longer the wavelength of a sound wave, the lower its frequency; the shorter the wavelength, the higher the frequency.*

#### Waveshaper

*It would be a bit obvious to say “a circuit that changes the shape of the waveform going through it”, but that is the point. Waveshapers often have specific goals in mind, such as converting an incoming triangle wave into an outgoing sine wave, or to add tube-like soft clipping to the peaks and transients of waves. Many waveshapers are simply intended to mangle (er, add higher harmonics to) waveforms in interesting ways, creating noisier (er, more complex and bright) harmonic spectra to create new sounds.*

#### Wavetable

*This term can have two related but slightly different meanings. A digital oscillator often produces sound by reading a table of numbers in order, jumping from the level described by one number to the next. This table of numbers describes one cycle of a wave, and therefore is often called a wavetable. Many digital oscillators have multiple wave tables lined up, and can move between these tables – either by jumping suddenly (which the original PPG Wave synths did), or by crossfading between them (what most digital wavetable oscillators today do). Some people refer to each table as a “wave” and a set of individual waves as a wavetable.*

#### Weighting

*An equalization curve used in audio tests that compensates for the Fletcher Munson Curve at various levels. (See also “Fletcher-Munson Curves.”)*

#### West Coast Synthesis

*The so-called “West Coast” approach to synthesis – traditionally associated with companies such as Buchla and Serge – is often based around adding harmonics to simple waveforms, rather than removing (filtering) them from complex waveforms. This is often accomplished by using a pair of oscillators (sometimes combined into what's called a “complex oscillator”) where one modulates the frequency (FM) or amplitude (AM) of the other; another common West Coast module is a waveshaper or a wavefolder. You may also find two-stage envelope generators such as an AD or AR (often called slope generators) rather than four-stage ADSRs, as well as more of an emphasis on control voltage manipulation. A common feature is also voltage controlled amplifiers that have low-pass filters built into them, creating what's known as a Low Pass Gate (LPG). The West Coast approach also embraces non-traditional controllers, such as touch plates and the such. Today it's common to mix both East*

*Coast and West Coast approaches in the same system.*

wet sound

*Sometimes people will say a filter has a “wet” sound. This usually refers to a fewer-than-4-pole filter sound – often low or bandpass – with resonance turned up a bit, but not to the point of self-oscillation. It’s a sound that is popular in acid house and other similar techno styles.*

Wet

*A sound with effects (such as reverb) mixed is referred to as “wet”; a sound with no effects is referred to as “dry.” Effects units or mixers often have wet/dry mix amounts that set the ratio between the original, unprocessed sound and the fully-effected sound.*

*Refers to a signal that has the full amount of an effect (like reverb) applied to it, as opposed to “dry,” which refers to the un-effected sound. Many times, the preferred sound in mixing will be a blend of wet and dry signals. (See also “Dry.”)*

White Noise

*Noise is a random signal that does not have a distinct pitch, such as hissing, breath noise, or the sound of wind or the surf. Noise is often described by different “colors” such as white, pink, red, or blue which have different frequency distributions. White noise has equal power per unit of frequency (such as every 1000 hertz), resulting in a brighter, hissier sound.*

*A noise signal containing an equal spread of energy across all audible frequencies. Like pink noise, engineers often send a white noise signal through audio equipment for tuning and calibration purposes, or in EQ-ing a live audio space. (See also “Pink Noise.”)*

Whole Step

*A change in pitch equivalent to two half steps, or the difference in pitch between two piano keys.*

Wild Sound

*In film and video, audio that is recorded separately from the visual that may be added to the audio track later, and does not need to be synchronized with the picture.*

Wind Controller

*A device that is played like a wind instrument to control a synthesizer, module or virtual instrument via MIDI signals, as opposed to a keyboard controller.*

Windscreen

*A covering that fits over a microphone to reduce the excessive noise resulting from wind blowing into the mic. Typically used for recording in outdoor locations.*

Wireless Microphone

*A microphone that transmits its signal over an FM frequency to a receiver offstage, rather than traveling over an audio cable.*

Woofers

*A speaker that is designed to reproduce bass frequencies only.*

Write Mode

*A mode of operation in an automated console where the engineer is in control of channel gain and the computer is recording the gain changes over time.*

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**XLR Cable**

*A balanced microphone cable utilizing XLR connectors. (See also “XLR Connector.”)*

**XLR Connector**

*A balanced cable connector consisting of 3 or 7 pins, most commonly used in microphone cables.*

**XY Miking**

*A coincident stereo microphone placement technique in which two cardioid microphones are placed with their heads toward each other at a 90-degree angle, and as close together as possible. (See also “Coincident Miking.”)*

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**Y-Cord**

*A cable with three connectors so that one output may be sent to two inputs. Basically, a signal splitter done with spliced wires rather than components.*

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**Zenith**

*In analog tape recording, refers to the tilt of the tape head in the direction perpendicular to the tape travel.*

**Zero-Order Hold (ZOH)**

*Refers to the mathematical expression of the signal processing done by a conventional digital-to-analog converter (DAC).*

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# Appendix C: URL for Source Code

00-signalgenerators.cpp

<https://gist.github.com/chbtoys/978de513b61ae51d5a3f6e8aec67861>

01-additive.cpp

<https://gist.github.com/chbtoys/d00429015a2025924d59afc82a86fc14>

02-subtractive.cpp

<https://gist.github.com/chbtoys/8c63f4c47e69adca5a10c1527fd21997>

03-formant.cpp

<https://gist.github.com/chbtoys/287deba92291d0dbcb4a71f46260de13>

04-granular.cpp

<https://gist.github.com/chbtoys/715f3c29737aa40a8edd640352aa6446>

05-fm.cpp

<https://gist.github.com/chbtoys/9d6deae69c525ff2ef46817f63ea1b67>

06-la.cpp

<https://gist.github.com/chbtoys/e0e85676130ebc447d6f77b771774947>

07-pd.cpp

<https://gist.github.com/chbtoys/c12da6644a0294f30a838d3cf9da9ff4>

08-scanned.cpp

<https://gist.github.com/chbtoys/02da87aff752f0c42110a6ce5049f149>

09-vectorsynth.cpp

<https://gist.github.com/chbtoys/ed489aa83edbb0c71d2a6d7efd245c41>

10-virtualanalog.cpp

<https://gist.github.com/chbtoys/79bcbd5e42aed04410c0a6bf1ac4b3a4>

11-wavetable.cpp

<https://gist.github.com/chbtoys/e7a25df378b680bbd2fe0f04d4da89a6>

12-physicalmodelling\_karplus\_strong.cpp

<https://gist.github.com/chbtoys/45bf343a9b339d44be389e644c75cf7e>

Envelope.hpp

<https://gist.github.com/chbtoys/83ac32fe2517985efdd056b96ca1d280>

MoogFilter.hpp

<https://gist.github.com/chbtoys/926ef26ace2eb9069a3a3d7d083efe48>

quiz.cpp

<https://gist.github.com/chbtoys/6ba96b42e76b10fd60af4ec8256b6c13>

quiz2.cpp

<https://gist.github.com/chbtoys/4ada0db42e7e0ba8b9e3c49117ea94ff>