Audio Programming in C++

The Beginner Level

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

Contents

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 expansive to build your own. Playing around with software like the free VCV Rack (from VCV Rack[1]) 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."

[1] 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 posses 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 x 100 (200 Hz), 3 x 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 w01 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 w02 and w03, 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[2]

[2]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 offer 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 filer 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[3]

[3]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![4]

[4] 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[5]

[5] 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 [6]

[6] 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-la.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![7]

[7] 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 `genereratephasedistortionwave` contains all the logic for the synthesis method.

Phase Distortion Synthesis with a Casio CZ-1000

Casio CZ-1000 | The Cosmo Synthesizer[8]

[8] 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++

`genereratescannedwave` 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[9]

[9] 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[10]

[10] 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[11]

[11] 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 wellsuited 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[12]

[12] https://www.youtube.com/watch?v=9bZ_VGFi5X0

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[13]

[13] 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.

```cpp
// Signal Generators
//
// Compile: clang++ -std=c++20 -lpng 00-signalgenerators.cpp -o 00-signalgenerators
//
// If #include <png++/png.hpp> is missing:
// > brew install png++
// png++ (header-only) is used to render audio as images to visualize the waveform.
// Install with homebrew on macOS or with other package manager.
// https://savannah.nongnu.org/projects/pngpp/
//
// Uses Adam Stark's AudioFile (header-only) to write AIFF-files.
// Check out the code from github link below.
// https://github.com/adamstark/AudioFile
//
// Uses p-ranav Indicators
// > git clone https://github.com/p-ranav/indicators
// > cd indicators
// > mkdir build && cd build
// > cmake -DINDICATORS_SAMPLES=ON -DINDICATORS_DEMO=ON ..
// > make
//
// python3 utils/amalgamate/amalgamate.py -c single_include.json -s .
//
// If clang++ is missing:
// clang++ from XCode:
// > xcode-select --install
// > xcode-select -p
//
// clang++ from homebrew:
// > brew install llvm
// If clang++ is needed, install it with homebrew for macOS or...
// https://clang.llvm.org/get_started.html

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
```

```
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
}

namespace SignalGenerators
{
        void generatesinewaveaiff();
        void rendersinewavepng();
        void generatesumoffirsteightpartialsaiff();
        void rendersumoffirsteightpartialspng();
        void generatesawtoothwaveaiff();
        void rendersawtoothwavepng();
        void generatesquarewaveaiff();
        void rendersquarewavepng();
        void generatetrianglewaveaiff();
        void rendertrianglewavepng();
        void generatetwentyfivepulsewaveaiff();
        void rendertwentyfivepulsewavepng();
        void generateconvexcurveaiff();
        void renderconvexcurvepng();
        void generateconcavecurveaiff();
        void renderconcavecurvepng();
        void generatefrequencymodulationaiff();
        void renderfrequencymodulationpng();
        void generatephasemodulationaiff();
        void renderphasemodulationpng();
        void generateamplitudedmodulationaiff();
        void renderamplitudedmodulationpng();
        void generateringmodulationaiff();
        void renderringmodulationpng();
        void generatenoiseaiff();
        void rendernoisepng();
        void generatepulsewaveaiff();
```

```
        void renderpulsewavepng();
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("signalgenerators");

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Generating: Sine Wave 1/28"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(13);

        SignalGenerators::generatesinewaveaiff();

        // Update progress
        bar.set_progress(19);
        bar.set_option(option::PostfixText{"Rendering: Sine Wave 2/28"});

        SignalGenerators::rendersinewavepng();

        // Update progress
        bar.set_progress(22);
        bar.set_option(option::PostfixText{"Generating: Sum of first eight partials 3/28"});

        SignalGenerators::generatesumoffirsteightpartialsaiff();

        // Update progress
```

```
        bar.set_progress(25);
        bar.set_option(option::PostfixText{"Rendering: Sum of first eight partials 4/28"});

        SignalGenerators::rendersumoffirsteightpartialspng();

        // Update progress
        bar.set_progress(28);
        bar.set_option(option::PostfixText{"Generating: Saw Tooth Wave 5/28"});

        SignalGenerators::generatesawtoothwaveaiff();

        // Update progress
        bar.set_progress(31);
        bar.set_option(option::PostfixText{"Rendering: Saw Tooth Wave 6/28"});

        SignalGenerators::rendersawtoothwavepng();

        // Update progress
        bar.set_progress(34);
        bar.set_option(option::PostfixText{"Generating: Square Wave 7/28"});

        SignalGenerators::generatesquarewaveaiff();

        // Update progress
        bar.set_progress(37);
        bar.set_option(option::PostfixText{"Rendering: Square Wave 8/28"});

        SignalGenerators::rendersquarewavepng();

        // Update progress
        bar.set_progress(40);
        bar.set_option(option::PostfixText{"Generating: Triangle Wave 9/28"});

        SignalGenerators::generatetrianglewaveaiff();

        // Update progress
        bar.set_progress(43);
        bar.set_option(option::PostfixText{"Rendering: Triangle Wave 10/28"});

        SignalGenerators::rendertrianglewavepng();

        // Update progress
        bar.set_progress(46);
```

```
        bar.set_option(option::PostfixText{"Generating: 25% Pulse Wave 11/28"});

        SignalGenerators::generatetwentyfivepulsewaveaiff();

        // Update progress
        bar.set_progress(49);
        bar.set_option(option::PostfixText{"Rendering: 25% Pulse Wave 12/28"});

        SignalGenerators::rendertwentyfivepulsewavepng();

        // Update progress
        bar.set_progress(52);
        bar.set_option(option::PostfixText{"Generating: Convex Curve 13/28"});

        SignalGenerators::generateconvexcurveaiff();

        // Update progress
        bar.set_progress(55);
        bar.set_option(option::PostfixText{"Rendering: Convex Curve 14/28"});

        SignalGenerators::renderconvexcurvepng();

        // Update progress
        bar.set_progress(58);
        bar.set_option(option::PostfixText{"Generating: Concave Curve 15/28"});

        SignalGenerators::generateconcavecurveaiff();

        // Update progress
        bar.set_progress(61);
        bar.set_option(option::PostfixText{"Rendering: Concave Curve 16/28"});

        SignalGenerators::renderconcavecurvepng();

        // Update progress
        bar.set_progress(64);
        bar.set_option(option::PostfixText{"Generating: Frequency Modulation 17/28"});


        SignalGenerators::generatefrequencymodulationaiff();

        // Update progress
        bar.set_progress(67);
```

```
        bar.set_option(option::PostfixText{"Rendering: Frequency Modulation 18/28"});

        SignalGenerators::renderfrequencymodulationpng();

        // Update progress
        bar.set_progress(70);
        bar.set_option(option::PostfixText{"Generating: Phase Modulation 19/28"});

        SignalGenerators::generatephasemodulationaiff();

        // Update progress
        bar.set_progress(73);
        bar.set_option(option::PostfixText{"Rendering: Phase Modulation 20/28"});

        SignalGenerators::renderphasemodulationpng();

        // Update progress
        bar.set_progress(76);
        bar.set_option(option::PostfixText{"Generating: Amplitude Modulation 21/28"});

        SignalGenerators::generateamplitudedmodulationaiff();

        // Update progress
        bar.set_progress(79);
        bar.set_option(option::PostfixText{"Rendering: Amplitude Modulation 22/28"});

        SignalGenerators::renderamplitudedmodulationpng();

        // Update progress
        bar.set_progress(82);
        bar.set_option(option::PostfixText{"Generating: Ring Modulation 23/28"});

        SignalGenerators::generateringmodulationaiff();

        // Update progress
        bar.set_progress(85);
        bar.set_option(option::PostfixText{"Rendering: Ring Modulation 24/28"});

        SignalGenerators::renderringmodulationpng();

        // Update progress
        bar.set_progress(88);
        bar.set_option(option::PostfixText{"Generating: White Noise 25/28"});
```

```

        SignalGenerators::generatenoiseaiff();

        // Update progress
        bar.set_progress(91);
        bar.set_option(option::PostfixText{"Rendering: White Noise 26/28"});

        SignalGenerators::rendernoisepng();

        // Update progress
        bar.set_progress(94);
        bar.set_option(option::PostfixText{"Generating: Pulse Wave 27/28"});

        SignalGenerators::generatepulsewaveaiff();

        // Update progress
        bar.set_progress(97);
        bar.set_option(option::PostfixText{"Rendering: Pulse Wave 28/28"});

        SignalGenerators::renderpulsewavepng();

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 28/28"});

        // Show cursor
        show_console_cursor(true);

        return 0;
}

namespace SignalGenerators
{
        void generatesinewaveaiff()
        {
                const std::string sineName="signalgenerators/01-sine-wave.aiff";
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;

                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
```

```
                a.setNumSamplesPerChannel(44100);

                // Genrate Sine wave
                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                a.samples[channel][i]=sin((static_cast<double> (i) / sampleRate) * 
 * 2.0 * M_PI);
                        }
                }

                a.save(sineName,AudioFileFormat::Aiff);
        }

        void rendersinewavepng()
        {
                const std::string sineName="signalgenerators/02-sine-wave.png";
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;
                std::vector<uint32_t> signalY;
                signalY.resize(44100);

                for (int i=0;i<sampleRate;++i)
                {
                        double value=sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2
_PI);
                        if (value >= 0.0) {
                                signalY[i]=300-(300*value);
                        } else if (value < 0.0)
                        {
                                signalY[i]=300+(300*fabs(value));
                        }
                }

                png::image<png::rgb_pixel> image(44100,600);
                Render::fillbackground(image);
                Render::drawwave(image,signalY);
                image.write(sineName);
        }

        void generatesumoffirsteightpartialsaiff()
        {
```

```
                const std::string sineName="signalgenerators/03-sine-wave-first-eight-partials.aif\
f";
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;

                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                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;

                // Genrate Sine wave
                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                s1=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*1) *
                                s2=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*2) *
                                s3=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*3) *
                                s4=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*4) *
                                s5=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*5) *
                                s6=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*6) *
                                s7=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*7) *
                                s8=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*8) *
                                ss=(s1+s2+s3+s4+s5+s6+s7+s8)/8;
                                a.samples[channel][i]=ss;
                        }
                }

                a.save(sineName,AudioFileFormat::Aiff);
        }

        void rendersumoffirsteightpartialspng()
        {
                const std::string sineName="signalgenerators/04-sine-wave-first-eight-partials.png\
";
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;
                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;
                std::vector<uint32_t> signalY;
                signalY.resize(44100);
```

```

        // Genrate Sine waves
        for (int i=0;i<sampleRate;++i)
        {
            s1=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*1) * 2.0 * M_
                s2=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*2) *
                s3=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*3) *
                s4=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*4) *
                s5=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*5) *
                s6=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*6) *
                s7=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*7) *
                s8=sin((static_cast<double> (i) / sampleRate) * (frequencyInHz*8) *
                ss=(s1+s2+s3+s4+s5+s6+s7+s8)/8;
            if (ss >= 0.0) {
                signalY[i]=300-(300*ss);
            } else if (ss < 0.0)
            {
                signalY[i]=300+(300*fabs(ss));
            }
        }

        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
    }

    void generatesawtoothwaveaiff()
    {
        const std::string sineName="signalgenerators/05-saw-tooth-wave.aiff";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double ss;

        // Setup the audio file
        AudioFile<float> a;
        a.setNumChannels(1);
        a.setBitDepth(24);
        a.setNumSamplesPerChannel(44100);

        for (int i=0;i<a.getNumSamplesPerChannel();++i)
        {
            for (int channel=0;channel<a.getNumChannels();++channel)
```

```
                {
                        ss=-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate));
                        a.samples[channel][i]=ss;
                }
        }

        a.save(sineName,AudioFileFormat::Aiff);
}

void rendersawtoothwavepng()
{
        const std::string sineName="signalgenerators/06-saw-tooth-wave.png";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double ss;
        std::vector<uint32_t> signalY;
        signalY.resize(44100);

        for (int i=0;i<44100;++i)
        {
                ss=-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate));
                if (ss >= 0.0) {
                        signalY[i]=300-(300*ss);
                } else if (ss < 0.0)
                {
                        signalY[i]=300+(300*fabs(ss));
                }
        }

        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
}

void generatesquarewaveaiff()
{
        const std::string sineName="signalgenerators/07-square-wave.aiff";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double period=sampleRate/frequencyInHz;
        double dutyCycle=period*0.5;
        double ss;
```

```

        // Setup the audio file
        AudioFile<float> a;
        a.setNumChannels(1);
        a.setBitDepth(24);
        a.setNumSamplesPerChannel(44100);

        for (int i=0;i<a.getNumSamplesPerChannel();++i)
        {
            for (int channel=0;channel<a.getNumChannels();++channel)
            {
                if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
                {
                    ss=0.7;
                } else {
                    ss=-0.7;
                }
                a.samples[channel][i]=ss;
            }
        }
        a.samples[0][0]=0.0;
        a.samples[0][44099]=0.0;
        a.save(sineName,AudioFileFormat::Aiff);
    }

    void rendersquarewavepng()
    {
        const std::string sineName="signalgenerators/08-square-wave.png";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double period=sampleRate/frequencyInHz;
        double dutyCycle=period*0.5;
        std::vector<uint32_t> signalY;
        signalY.resize(44100);

        for (int i=0;i<44100;++i)
        {
            if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
            {
                signalY[i]=0;
            } else {
                signalY[i]=600;
            }
```

```
        }

        signalY[0]=300;
        signalY[44099]=300;

        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
    }

    void generatetrianglewaveaiff()
    {
        const std::string sineName="signalgenerators/09-trangle-wave.aiff";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double ss;

        // Setup the audio file
        AudioFile<float> a;
        a.setNumChannels(1);
        a.setBitDepth(24);
        a.setNumSamplesPerChannel(44100);

        for (int i=0;i<a.getNumSamplesPerChannel();++i)
        {
            for (int channel=0;channel<a.getNumChannels();++channel)
            {
                ss=M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate));
                a.samples[channel][i]=ss;
            }
        }
        a.save(sineName,AudioFileFormat::Aiff);
    }

    void rendertrianglewavepng()
    {
        const std::string sineName="signalgenerators/10-trangle-wave.png";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double ss;
        std::vector<uint32_t> signalY;
        signalY.resize(44100);
```

```

			for (int i=0;i<44100;++i)
			{
				ss=M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate));
				if (ss >= 0.0) {
					signalY[i]=300-(300*ss);
				} else if (ss < 0.0)
				{
					signalY[i]=300+(300*fabs(ss));
				}
			}

			png::image<png::rgb_pixel> image(44100,600);
			Render::fillbackground(image);
			Render::drawwave(image,signalY);
			image.write(sineName);
		}

		void generatetwentyfivepulsewaveaiff()
		{
			const std::string sineName="signalgenerators/11-twentyfive-pulse-wave.aiff";
			const double sampleRate=44100.0;
			const double frequencyInHz=440.0;
			double period=sampleRate/frequencyInHz;
			double dutyCycle=period*0.25;
			double ss;

			// Setup the audio file
			AudioFile<float> a;
			a.setNumChannels(1);
			a.setBitDepth(24);
			a.setNumSamplesPerChannel(44100);

			for (int i=0;i<a.getNumSamplesPerChannel();++i)
			{
				for (int channel=0;channel<a.getNumChannels();++channel)
				{
					if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
					{
						ss=0.7;
					} else {
						ss=-0.7;
					}
```

```
                        a.samples[channel][i]=ss;
                }
        }
        a.save(sineName,AudioFileFormat::Aiff);
}

void rendertwentyfivepulsewavepng()
{
        const std::string sineName="signalgenerators/12-twentyfive-pulse-wave.png";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double period=sampleRate/frequencyInHz;
        double dutyCycle=period*0.25;
        std::vector<uint32_t> signalY;
        signalY.resize(44100);

        for (int i=0;i<44100;++i)
        {
                if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
                {
                        signalY[i]=0;
                } else {
                        signalY[i]=600;
                }
        }

        signalY[0]=300;
        signalY[44099]=300;

        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
}

void generateconvexcurveaiff()
{
        const std::string sineName="signalgenerators/13-convex-curve.aiff";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double period=sampleRate/frequencyInHz;
        uint32_t maxValue=0,temp=0;
        double ss,sss;
```

```

		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<44100;++i)
		{
			temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
			if (temp > maxValue) {maxValue=temp;}
		}

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
				ss=double(temp)/double(maxValue);
				sss=(ss-0.5)*1.5;
				a.samples[channel][i]=sss;
			}
		}
		a.save(sineName,AudioFileFormat::Aiff);
	}

	void renderconvexcurvepng()
	{
		const std::string sineName="signalgenerators/14-convex-curve.png";
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
		double period=sampleRate/frequencyInHz;
		double ss;
		uint32_t maxValue=0,temp=0;
		std::vector<uint32_t> signalY;
		signalY.resize(44100);

		for (int i=0;i<44100;++i)
		{
			temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
			if (temp > maxValue) {maxValue=temp;}
		}

```

```
        for (int i=0;i<44100;++i)
        {
            temp=pow(abs((i % int(period)) - (int(period)/2)), 2.0);
            ss=double(temp)/double(maxValue);
            signalY[i]=600-(600*ss);
        }

        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
    }

    void generateconcavecurveaiff()
    {
        const std::string sineName="signalgenerators/15-concave-curve.aiff";
        const double sampleRate=44100.0;
        const double frequencyInHz=440.0;
        double period=sampleRate/frequencyInHz;
        uint32_t maxValue=0;
        double ss,sss,temp;

        // Setup the audio file
        AudioFile<float> a;
        a.setNumChannels(1);
        a.setBitDepth(24);
        a.setNumSamplesPerChannel(44100);

        for (int i=0;i<44100;++i)
        {
            temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
            if (int(temp) > maxValue) {maxValue=int(temp);}
        }

        for (int i=0;i<a.getNumSamplesPerChannel();++i)
        {
            for (int channel=0;channel<a.getNumChannels();++channel)
            {
                temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
                ss=temp/double(maxValue);
                sss=(ss-0.5)*1.5;
                a.samples[channel][i]=sss;
            }
```

```
            }
            a.save(sineName,AudioFileFormat::Aiff);
    }

    void renderconcavecurvepng()
    {
            const std::string sineName="signalgenerators/16-concave-curve.png";
            const double sampleRate=44100.0;
            const double frequencyInHz=440.0;
            double period=sampleRate/frequencyInHz;
            double ss,temp;
            uint32_t maxValue=0;
            std::vector<uint32_t> signalY;
            signalY.resize(44100);

            for (int i=0;i<44100;++i)
            {
                    temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
                    if (int(temp) > maxValue) {maxValue=int(temp);}
            }

            for (int i=0;i<44100;++i)
            {
                    temp=pow(abs((i % int(period)) - (period/2.0)), 0.5);
                    ss=temp/double(maxValue);
                    signalY[i]=600-(600*ss);
            }

            png::image<png::rgb_pixel> image(44100,600);
            Render::fillbackground(image);
            Render::drawwave(image,signalY);
            image.write(sineName);
    }

    void generatefrequencymodulationaiff()
    {
            const std::string sineName="signalgenerators/17-frequency-modulation.aiff";
            const double twoPI=2*M_PI;
            const double sampleRate=44100.0;
            const double frequencyInHz=440.0;
            const double frequencyRadian = twoPI / sampleRate;
            double modifierFrequency = frequencyInHz * 3;
            double modifierIncrement = frequencyRadian * modifierFrequency;
```

```
		double modifierPhase = 0;
		double carrierIncrement = 0;
		double carrierPhase = 0;
		double modifieramplitude = 2 * modifierFrequency;
		double modifierValue = 0;

		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=sinf(carrierPhase);
				modifierValue = modifieramplitude * sinf(modifierPhase);
				carrierIncrement = frequencyRadian * (frequencyInHz + modifierValue)
				carrierPhase = carrierPhase + carrierIncrement;
				modifierPhase = modifierPhase + modifierIncrement;
				if (carrierPhase >= twoPI) {
					carrierPhase -= twoPI;
				}
				else if (carrierPhase < 0) {
					carrierPhase += twoPI;
				}
				if (modifierPhase >= twoPI) {
					modifierPhase -= twoPI;
				}
			}
		}
		a.save(sineName,AudioFileFormat::Aiff);
	}

	void renderfrequencymodulationpng()
	{
		const std::string sineName="signalgenerators/18-frequency-modulation.png";
		const double twoPI=2*M_PI;
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
		const double frequencyRadian = twoPI / sampleRate;
		double modifierFrequency = frequencyInHz * 3;
```

```
            double modifierIncrement = frequencyRadian * modifierFrequency;
            double modifierPhase = 0;
            double carrierIncrement = 0;
            double carrierPhase = 0;
            double modifieramplitude = 2 * modifierFrequency;
            double modifierValue = 0;
            double value=0;
            std::vector<uint32_t> signalY;
            signalY.resize(44100);

            for (uint32_t i=0;i<sampleRate;++i)
            {
                    value=sinf(carrierPhase);
                    if (value >= 0.0) {
                            signalY[i]=300-(300*value);
                    } else if (value < 0.0)
                    {
                            signalY[i]=300+(300*fabs(value));
                    }
                    modifierValue = modifieramplitude * sinf(modifierPhase);
                    carrierIncrement = frequencyRadian * (frequencyInHz + modifierValue);
                    carrierPhase = carrierPhase + carrierIncrement;
                    modifierPhase = modifierPhase + modifierIncrement;
                    if (carrierPhase >= twoPI) {
                            carrierPhase -= twoPI;
                    }
                    else if (carrierPhase < 0) {
                            carrierPhase += twoPI;
                    }
                    if (modifierPhase >= twoPI) {
                            modifierPhase -= twoPI;
                    }
            }
            png::image<png::rgb_pixel> image(44100,600);
            Render::fillbackground(image);
            Render::drawwave(image,signalY);
            image.write(sineName);
    }

    void generatephasemodulationaiff()
    {
            const std::string sineName="signalgenerators/19-phase-modulation.aiff";
            const double twoPI=2*M_PI;
```

```
            const double sampleRate=44100.0;
            const double frequencyInHz=440.0;
            const double frequencyRadian = twoPI / sampleRate;
            double modifierFrequency = frequencyInHz * 3;
            double carrierIncrement = frequencyRadian * frequencyInHz;
            double modifieramplitude = frequencyRadian * (2 * modifierFrequency);
            double modifierValue = 0;
            double carrierPhase = 0;
            double modifierPhase = 0;
            double modifierIncrement = frequencyRadian * modifierFrequency;

            // Setup the audio file
            AudioFile<float> a;
            a.setNumChannels(1);
            a.setBitDepth(24);
            a.setNumSamplesPerChannel(44100);

            for (int i=0;i<a.getNumSamplesPerChannel();++i)
            {
                for (int channel=0;channel<a.getNumChannels();++channel)
                {
                    a.samples[channel][i]=sinf(carrierPhase);
                    modifierValue = modifieramplitude * sinf(modifierPhase);
                    carrierPhase = carrierPhase + carrierIncrement + modifierValue;
                    modifierPhase = modifierPhase + modifierIncrement;
                    if (carrierPhase >= twoPI) {
                        carrierPhase -= twoPI;
                    }
                    else if (carrierPhase < 0) {
                        carrierPhase += twoPI;
                    }
                    if (modifierPhase >= twoPI) {
                        modifierPhase -= twoPI;
                    }
                }
            }
            a.save(sineName,AudioFileFormat::Aiff);
        }

        void renderphasemodulationpng()
        {
            const std::string sineName="signalgenerators/20-phase-modulation.png";
            const double twoPI=2*M_PI;
```

```
            const double sampleRate=44100.0;
            const double frequencyInHz=440.0;
            const double frequencyRadian = twoPI / sampleRate;
            double modifierFrequency = frequencyInHz * 3;
            double carrierIncrement = frequencyRadian * frequencyInHz;
            double modifieramplitude = frequencyRadian * (2 * modifierFrequency);
            double modifierValue = 0;
            double carrierPhase = 0;
            double modifierPhase = 0;
            double modifierIncrement = frequencyRadian * modifierFrequency;
            double value=0;
            std::vector<uint32_t> signalY;
            signalY.resize(44100);

            for (uint32_t i=0;i<sampleRate;++i)
            {
                    value=sinf(carrierPhase);
                    if (value >= 0.0) {
                            signalY[i]=300-(300*value);
                    } else if (value < 0.0)
                    {
                            signalY[i]=300+(300*fabs(value));
                    }
                    modifierValue = modifieramplitude * sinf(modifierPhase);
                    carrierPhase = carrierPhase + carrierIncrement + modifierValue;
                    modifierPhase = modifierPhase + modifierIncrement;
                    if (carrierPhase >= twoPI) {
                            carrierPhase -= twoPI;
                    }
                    else if (carrierPhase < 0) {
                            carrierPhase += twoPI;
                    }
                    if (modifierPhase >= twoPI){
                            modifierPhase -= twoPI;
                    }
            }
            png::image<png::rgb_pixel> image(44100,600);
            Render::fillbackground(image);
            Render::drawwave(image,signalY);
            image.write(sineName);
    }

    void generateamplitudedmodulationaiff()
```

```
        {
                const std::string sineName="signalgenerators/21-amplituded-modulation.aiff";
                const double twoPI=2*M_PI;
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;
                const double frequencyRadian = twoPI / sampleRate;
                double modifierFrequency = frequencyInHz * 2.5;
                double modifierIncrement = frequencyRadian * modifierFrequency;
                double modifieramplitude = 1.0;
                double carrierPhase = 0;
                double modifierPhase = 0;
                double modifierScale = 1 / (1 + modifieramplitude);
                double modifierValue = 0;
                double carrierIncrement = frequencyRadian * frequencyInHz;

                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                modifierValue = 1.0 + (modifieramplitude * sinf(modifierPhase));
                                a.samples[channel][i]=((sinf(carrierPhase) * modifierValue) * modif
                                carrierPhase += carrierIncrement;
                                if (carrierPhase >= twoPI) {
                                        carrierPhase -= twoPI;
                                }
                                modifierPhase += modifierIncrement;
                                if (modifierPhase >= twoPI) {
                                        modifierPhase -= twoPI;
                                }
                        }
                }
                a.save(sineName,AudioFileFormat::Aiff);
        }

        void renderamplitudedmodulationpng()
        {
                const std::string sineName="signalgenerators/22-amplituded-modulation.png";
```

```
		const double twoPI=2*M_PI;
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
		const double frequencyRadian = twoPI / sampleRate;
		double modifierFrequency = frequencyInHz * 2.5;
		double modifierIncrement = frequencyRadian * modifierFrequency;
		double modifieramplitude = 1.0;
		double carrierPhase = 0;
		double modifierPhase = 0;
		double modifierScale = 1 / (1 + modifieramplitude);
		double modifierValue = 0;
		double carrierIncrement = frequencyRadian * frequencyInHz;
		double value=0;
		std::vector<uint32_t> signalY;
		signalY.resize(44100);

		for (uint32_t i=0;i<sampleRate;++i)
		{
			modifierValue = 1.0 + (modifieramplitude * sinf(modifierPhase));
			value=((sinf(carrierPhase) * modifierValue) * modifierScale);
			if (value >= 0.0) {
				signalY[i]=300-(300*value);
			} else if (value < 0.0)
			{
				signalY[i]=300+(300*fabs(value));
			}
			carrierPhase += carrierIncrement;
			if (carrierPhase >= twoPI) {
				carrierPhase -= twoPI;
			}
			modifierPhase += modifierIncrement;
			if (modifierPhase >= twoPI) {
				modifierPhase -= twoPI;
			}
		}
		png::image<png::rgb_pixel> image(44100,600);
		Render::fillbackground(image);
		Render::drawwave(image,signalY);
		image.write(sineName);
	}

	void generateringmodulationaiff()
	{
```

```
		const std::string sineName="signalgenerators/23-ring-modulation.aiff";
		const double twoPI=2*M_PI;
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
		const double frequencyRadian = twoPI / sampleRate;
		double modifieramplitude = 1.0;
		double modifierFrequency = frequencyInHz * 2.5;
		double modifierPhase = 0;
		double modifierIncrement = frequencyRadian * modifierFrequency;
		double carrierPhase = 0;
		double carrierIncrement = frequencyRadian * frequencyInHz;

		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=(sinf(carrierPhase) * modifieramplitude * sin
se));
				carrierPhase += carrierIncrement;
				if (carrierPhase >= twoPI) {
					carrierPhase -= twoPI;
				}
				modifierPhase += modifierIncrement;
				if (modifierPhase >= twoPI) {
					modifierPhase -= twoPI;
				}
			}
		}
		a.save(sineName,AudioFileFormat::Aiff);
	}

	void renderringmodulationpng()
	{
		const std::string sineName="signalgenerators/24-ring-modulation.png";
		const double twoPI=2*M_PI;
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
```

```
        const double frequencyRadian = twoPI / sampleRate;
        double modifieramplitude = 1.0;
        double modifierFrequency = frequencyInHz * 2.5;
        double modifierPhase = 0;
        double modifierIncrement = frequencyRadian * modifierFrequency;
        double carrierPhase = 0;
        double carrierIncrement = frequencyRadian * frequencyInHz;
        double value=0;
        std::vector<uint32_t> signalY;
        signalY.resize(44100);

        for (uint32_t i=0;i<sampleRate;++i)
        {
                value=(sinf(carrierPhase) * modifieramplitude * sinf(modifierPhase));
                if (value >= 0.0) {
                        signalY[i]=300-(300*value);
                } else if (value < 0.0)
                {
                        signalY[i]=300+(300*fabs(value));
                }
                carrierPhase += carrierIncrement;
                if (carrierPhase >= twoPI) {
                        carrierPhase -= twoPI;
                }
                modifierPhase += modifierIncrement;
                if (modifierPhase >= twoPI) {
                        modifierPhase -= twoPI;
                }
        }
        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
}

void generatenoiseaiff()
{
        const std::string sineName="signalgenerators/25-white-noise.aiff";
        std::random_device rd;
        std::mt19937 gen(rd());
        std::uniform_real_distribution<> dis(-1.0, 1.0);

        // Setup the audio file
```

```
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=dis(gen);
			}
		}
		a.save(sineName,AudioFileFormat::Aiff);
	}

	void rendernoisepng()
	{
		const std::string sineName="signalgenerators/26-white-noise.png";
		const double sampleRate=44100.0;
		std::random_device rd;
		std::mt19937 gen(rd());
		std::uniform_real_distribution<> dis(-1.0, 1.0);
		double value=0;
		std::vector<uint32_t> signalY;
		signalY.resize(44100);

		for (uint32_t i=0;i<sampleRate;++i)
		{
			value=dis(gen);
			if (value >= 0.0) {
				signalY[i]=300-(300*value);
			} else if (value < 0.0)
			{
				signalY[i]=300+(300*fabs(value));
			}
		}
		png::image<png::rgb_pixel> image(44100,600);
		Render::fillbackground(image);
		Render::drawwave(image,signalY);
		image.write(sineName);
	}

	void generatepulsewaveaiff()
```

```
        {
                const std::string sineName="signalgenerators/27-pulse-wave.aiff";
                const double twoPI=2*M_PI;
                const double sampleRate=44100.0;
                const double frequencyInHz=440.0;
                const double frequencyRadian = twoPI / sampleRate;
                double maxNumber = floor(sampleRate / (2.0 * frequencyInHz) - 0.5) - 1;
                double phaseIncrementDenominator = (M_PI / sampleRate) * frequencyInHz;
                double phaseIncrementNumerator = phaseIncrementDenominator * ((2.0 * maxNumber) + \
1);
                double phaseDenominator = 0.0;
                double phaseNumerator = 0.0;
                double amplitudeScale = 1.0 / (2.0 * maxNumber);
                double value = 0;

                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                double Denominator = sin(phaseDenominator);
                                if (Denominator == 0.0) {
                                        value = 1.0;
                                } else {
                                        value = amplitudeScale * ((sin(phaseNumerator) / Denominato
                                }
                                a.samples[channel][i]=value;
                                phaseDenominator += phaseIncrementDenominator;
                                if (phaseDenominator >= twoPI) {
                                        phaseDenominator -= twoPI;
                                }
                                phaseNumerator += phaseIncrementNumerator;
                                if (phaseNumerator >= twoPI) {
                                        phaseNumerator -= twoPI;
                                }
                        }
                }
                a.save(sineName,AudioFileFormat::Aiff);
```

```
	}

	void renderpulsewavepng()
	{
		const std::string sineName="signalgenerators/28-pulse-wave.png";
		const double twoPI=2*M_PI;
		const double sampleRate=44100.0;
		const double frequencyInHz=440.0;
		const double frequencyRadian = twoPI / sampleRate;
		double maxNumber = floor(sampleRate / (2.0 * frequencyInHz) - 0.5) - 1;
		double phaseIncrementDenominator = (M_PI / sampleRate) * frequencyInHz;
		double phaseIncrementNumerator = phaseIncrementDenominator * ((2.0 * maxNumber) + \
1);
		double phaseDenominator = 0.0;
		double phaseNumerator = 0.0;
		double amplitudeScale = 1.0 / (2.0 * maxNumber);
		double value = 0;
		std::vector<uint32_t> signalY;
		signalY.resize(44100);

		for (uint32_t i=0;i<sampleRate;++i)
		{
			double Denominator = sin(phaseDenominator);
			if (Denominator == 0.0) {
				value = 1.0;
			} else {
				value = amplitudeScale * ((sin(phaseNumerator) / Denominator) - 1);
			}

			if (value >= 0.0) {
				signalY[i]=300-(300*value);
			} else if (value < 0.0)
			{
				signalY[i]=300+(300*fabs(value));
			}

			phaseDenominator += phaseIncrementDenominator;
			if (phaseDenominator >= twoPI) {
				phaseDenominator -= twoPI;
			}
			phaseNumerator += phaseIncrementNumerator;
			if (phaseNumerator >= twoPI) {
				phaseNumerator -= twoPI;
```

```
            }
        }
        png::image<png::rgb_pixel> image(44100,600);
        Render::fillbackground(image);
        Render::drawwave(image,signalY);
        image.write(sineName);
    }
}

namespace Render
{
    void fillbackground(png::image<png::rgb_pixel>& image)
    {
        png::rgb_pixel px(0x04,0x13,0x31);
        for (uint32_t y=0;y<image.get_height();y++) {
            for (uint32_t x=0;x<image.get_width();++x) {
                image.set_pixel(x,y,px);
            }
        }
    }

    void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
    {
        if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
        {
            png::rgb_pixel px(0x7a,0xb1,0xe3);
            image.set_pixel(x,y,px);
        }
    }

    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
        int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
        dx = x2 - x1; dy = y2 - y1;
        if (dx == 0)
        {
            if (y2 < y1) std::swap(y1, y2);
            for (y = y1; y <= y2; y++)
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
```

```
{
	if (x2 < x1) std::swap(x1, x2);
	for (x = x1; x <= x2; x++)
		drawpx(image, x, y1);
	return;
}
dx1 = abs(dx); dy1 = abs(dy);
px = 2 * dy1 - dx1;	py = 2 * dx1 - dy1;
if (dy1 <= dx1)
{
	if (dx >= 0)
	{
		x = x1; y = y1; xe = x2;
	}
	else
	{
		x = x2; y = y2; xe = x1;
	}
	drawpx(image, x, y);
	for (i = 0; x<xe; i++)
	{
		x = x + 1;
		if (px<0)
			px = px + 2 * dy1;
		else
		{
			if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
			px = px + 2 * (dy1 - dx1);
		}
		drawpx(image, x, y);
	}
}
else
{
	if (dy >= 0)
	{
		x = x1; y = y1; ye = y2;
	}
	else
	{
		x = x2; y = y2; ye = y1;
	}
	drawpx(image, x, y);
```

```
				for (i = 0; y<ye; i++)
				{
					y = y + 1;
					if (py <= 0)
						py = py + 2 * dx1;
					else
					{
						if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
						py = py + 2 * (dx1 - dy1);
					}
					drawpx(image, x, y);
				}
			}
		}

	void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
	{
		uint32_t y=0,ox=0,oy=0;
		for (uint32_t x=0;x<image.get_width();++x)
		{
			y=signalY[x];
			if (x == 0) {ox=x;oy=y;}
			drawline(image,x,y,ox,oy);
			ox=x;oy=y;
		}
	}
}
```

Envelope.hpp - 3356 bytes.

```
#ifndef ADSR_H
#define ADSR_H

#include <vector>
#include <cmath>

namespace ADSR
{
	class Envelope
	{
	public:
		Envelope(float sampleRate,uint32_t duration)
		{
```

```
        sampleLength=sampleRate*duration;
        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;
    }
    ~Envelope() {}
    void generateenvelope(std::vector<float>& env)
    {
        env.resize(sampleLength);
        uint32_t height=600;
        coords[0]=0;// x1
        coords[1]=height;// y1
        coords[2]=sampleLength*A_l;// x2
        coords[3]=height-(height*A_r);// y2
        coords[4]=sampleLength*(A_l+D_l);// x3
        coords[5]=height-(height*D_r);// y3
        coords[6]=sampleLength*(A_l+D_l+S_l);// x4
        coords[7]=height-(height*S_r);// y4
        coords[8]=sampleLength;// x5
        coords[9]=height;// y5
        bridge(env,coords[0],coords[1],coords[2],coords[3]);
        bridge(env,coords[2],coords[3],coords[4],coords[5]);
        bridge(env,coords[4],coords[5],coords[6],coords[7]);
        bridge(env,coords[6],coords[7],coords[8],coords[9]);
    }
    void generateenvelope2(std::vector<float>& env)
    {
        env.resize(sampleLength);
        uint32_t height=600;
        coords[0]=0;// x1
        coords[1]=height;// y1
        coords[2]=sampleLength;// x5
        coords[3]=0;// y5
        bridge(env,coords[0],coords[1],coords[2],coords[3]);
    }
    void generateenvelope3(std::vector<float>& env)
    {
        env.resize(sampleLength);
        uint32_t height=600;
        coords[0]=0;// x1
        coords[1]=0;// y1
        coords[2]=sampleLength;// x5
        coords[3]=height;// y5
        bridge(env,coords[0],coords[1],coords[2],coords[3]);
    }
```

```
        void applyenvelope(std::vector<float>& v,std::vector<float>& env)
        {
            for (uint32_t i=0;i<env.size();++i)
            {
                v[i]=v[i]*env[i];
            }
        }
    private:
        void setpt(std::vector<float>& env, uint32_t x, uint32_t y)
        {
            if (((x >= 0) && (x < sampleLength)) && ((y >= 0) && (y < 600)))
            {
                env[x]=((600.0-float(y))/600.0);
            }
        }

        void bridge(std::vector<float>& env, uint32_t x1, uint32_t y1, uint32_t x2, uint32\
_t y2)
        {
            int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
            dx = x2 - x1; dy = y2 - y1;
            if (dx == 0)
            {
                if (y2 < y1) std::swap(y1, y2);
                for (y = y1; y <= y2; y++)
                    setpt(env, x1, y);
                return;
            }
            if (dy == 0)
            {
                if (x2 < x1) std::swap(x1, x2);
                for (x = x1; x <= x2; x++)
                    setpt(env, x, y1);
                return;
            }
            dx1 = std::abs(dx); dy1 = std::abs(dy);
            px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
            if (dy1 <= dx1)
            {
                if (dx >= 0)
                {
                    x = x1; y = y1; xe = x2;
                }
```

```
                else
                {
                    x = x2; y = y2; xe = x1;
                }
                setpt(env, x, y);
                for (i = 0; x<xe; i++)
                {
                    x = x + 1;
                    if (px<0)
                        px = px + 2 * dy1;
                    else
                    {
                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; els
                        px = px + 2 * (dy1 - dx1);
                    }
                    setpt(env, x, y);
                }
            }
            else
            {
                if (dy >= 0)
                {
                    x = x1; y = y1; ye = y2;
                }
                else
                {
                    x = x2; y = y2; ye = y1;
                }
                setpt(env, x, y);
                for (i = 0; y<ye; i++)
                {
                    y = y + 1;
                    if (py <= 0)
                        py = py + 2 * dx1;
                    else
                    {
                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; els
                        py = py + 2 * (dx1 - dy1);
                    }
                    setpt(env, x, y);
                }
            }
        }
```

```

			float A_l;
			float A_r;
			float D_l;
			float D_r;
			float S_l;
			float S_r;
			float R_l;
			float R_r;
	protected:
			uint32_t sampleLength;
			uint32_t coords[10];
	};
}
#endif
```

MoogFilter.hpp - 1628 bytes.

```
#ifndef MOOG_H
#define MOOG_H

#define MOOG_PI		3.14159265358979323846264338327950288
#include <vector>

namespace Moog
{
	class MoogFilter
	{
	public:
		MoogFilter(float sampleRate) : sampleRate(sampleRate)
		{
			memset(stage, 0, sizeof(stage));
			memset(delay, 0, sizeof(delay));
			SetCutoff(1000.0f);
			SetResonance(0.10f);
		}
		~MoogFilter() {}

		void Process(std::vector<float>& samples, uint32_t n)
		{
			for (int s = 0; s < n; ++s)
			{
				float x = samples[s] - resonance * stage[3];
```

```

                // Four cascaded one-pole filters (bilinear transform)
                stage[0] = x * p + delay[0]  * p - k * stage[0];
                stage[1] = stage[0] * p + delay[1] * p - k * stage[1];
                stage[2] = stage[1] * p + delay[2] * p - k * stage[2];
                stage[3] = stage[2] * p + delay[3] * p - k * stage[3];

                // Clipping band-limited sigmoid
                stage[3] -= (stage[3] * stage[3] * stage[3]) / 6.0;

                delay[0] = x;
                delay[1] = stage[0];
                delay[2] = stage[1];
                delay[3] = stage[2];

                samples[s] = stage[3];
            }
        }

        void SetResonance(float r)
        {
            resonance = r * (t2 + 6.0 * t1) / (t2 - 6.0 * t1);
        }

        void SetCutoff(float c)
        {
            cutoff = 2.0 * c / sampleRate;

            p = cutoff * (1.8 - 0.8 * cutoff);
            k = 2.0 * sin(cutoff * MOOG_PI * 0.5) - 1.0;
            t1 = (1.0 - p) * 1.386249;
            t2 = 12.0 + t1 * t1;

            SetResonance(resonance);
        }

        float GetResonance() { return resonance; }
        float GetCutoff() { return cutoff; }
    private:
        double stage[4];
        double delay[4];

        double p;
```

```
			double k;
			double t1;
			double t2;
		protected:
			float cutoff;
			float resonance;
			float sampleRate;
	};
}
#endif
```

01-additive.cpp - 21478 bytes.

```
// compile: clang++ -std=c++20 -lpng 01-additive.cpp -o 01-additive

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image);
	void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
	void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
	void saveimagefile(std::vector<uint32_t>& v, std::string filename);
	void saveimagefile(std::vector<float>& v2, std::string filename);
	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
	void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
```

```
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz);
        void generatenoise(std::vector<float>& v, int duration);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("additive");

        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> osc1;
        std::vector<float> osc2;
        std::vector<float> osc3;
        std::vector<float> osc4;
        std::vector<float> osc5;
        std::vector<float> osc6;
        std::vector<float> osc7;
        std::vector<float> osc8;
        std::vector<float> osc9;
        std::vector<float> osc10;
        std::vector<float> osc11;
        std::vector<float> osc12;
        std::vector<float> osc13;
        std::vector<float> osc14;
        std::vector<float> osc15;
        std::vector<float> osc16;
        std::vector<float> osc17;
        std::vector<float> osc18;
        std::vector<float> osc19;
        std::vector<float> osc20;
        std::vector<float> osc21;
        std::vector<float> osc22;
        std::vector<float> osc23;
        std::vector<float> osc24;
        std::vector<float> osc25;
        std::vector<float> osc26;
        std::vector<float> osc27;
```

```
        std::vector<float> osc28;
        std::vector<float> osc29;
        std::vector<float> osc30;
        std::vector<float> osc31;
        std::vector<float> osc32;
        std::vector<float> osc33;
        std::vector<float> mixer;
        std::vector<float> envelope;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Setting: Sine Oscillator1 @ 55 Hz 1/35"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::generatesinewave(osc1,duration,55);
        SignalGenerators::gain(osc1,0.5);
        SignalGenerators::saveaudiofile(osc1,"01-osc1.aiff");
        Render::saveimagefile(osc1,"02-osc1.png");

        // Update progress
        bar.set_progress(3);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator2 @ 110 Hz 2/35"});

        SignalGenerators::generatesinewave(osc2,duration,110);
        SignalGenerators::gain(osc2,0.8);
        SignalGenerators::saveaudiofile(osc2,"03-osc2.aiff");
        Render::saveimagefile(osc2,"04-osc2.png");
```

```
        // Update progress
        bar.set_progress(6);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator3 @ 165 Hz 3/35"});

        SignalGenerators::generatesinewave(osc3,duration,165);
        SignalGenerators::gain(osc3,0.7);
        SignalGenerators::saveaudiofile(osc3,"05-osc3.aiff");
        Render::saveimagefile(osc3,"06-osc3.png");

        // Update progress
        bar.set_progress(8);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator4 @ 220 Hz 4/35"});

        SignalGenerators::generatesinewave(osc4,duration,220);
        SignalGenerators::gain(osc4,0.8);
        SignalGenerators::saveaudiofile(osc4,"07-osc4.aiff");
        Render::saveimagefile(osc4,"08-osc4.png");

        // Update progress
        bar.set_progress(11);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator5 @ 275 Hz 5/35"});

        SignalGenerators::generatesinewave(osc5,duration,275);
        SignalGenerators::gain(osc5,0.4);
        SignalGenerators::saveaudiofile(osc5,"09-osc5.aiff");
        Render::saveimagefile(osc5,"10-osc5.png");

        // Update progress
        bar.set_progress(14);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator6 @ 330 Hz 6/35"});

        SignalGenerators::generatesinewave(osc6,duration,330);
        SignalGenerators::gain(osc6,0.7);
        SignalGenerators::saveaudiofile(osc6,"11-osc6.aiff");
        Render::saveimagefile(osc6,"12-osc6.png");

        // Update progress
        bar.set_progress(17);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator7 @ 385 Hz 7/35"});

        SignalGenerators::generatesinewave(osc7,duration,385);
        SignalGenerators::gain(osc7,0.3);
        SignalGenerators::saveaudiofile(osc7,"13-osc7.aiff");
```

```
        Render::saveimagefile(osc7,"14-osc7.png");

        // Update progress
        bar.set_progress(20);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator8 @ 440 Hz 8/35"});

        SignalGenerators::generatesinewave(osc8,duration,440);
        SignalGenerators::gain(osc8,0.9);
        SignalGenerators::saveaudiofile(osc8,"15-osc8.aiff");
        Render::saveimagefile(osc8,"16-osc8.png");

        // Update progress
        bar.set_progress(23);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator9 @ 495 Hz 9/35"});

        SignalGenerators::generatesinewave(osc9,duration,495);
        SignalGenerators::gain(osc9,0.5);
        SignalGenerators::saveaudiofile(osc9,"17-osc8.aiff");
        Render::saveimagefile(osc9,"18-osc8.png");

        // Update progress
        bar.set_progress(26);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator10 @ 550 Hz 10/35"});

        SignalGenerators::generatesinewave(osc10,duration,550);
        SignalGenerators::gain(osc10,0.5);
        SignalGenerators::saveaudiofile(osc10,"19-osc8.aiff");
        Render::saveimagefile(osc10,"20-osc8.png");

        // Update progress
        bar.set_progress(29);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator11 @ 605 Hz 11/35"});

        SignalGenerators::generatesinewave(osc11,duration,605);
        SignalGenerators::gain(osc11,0.5);
        SignalGenerators::saveaudiofile(osc11,"21-osc8.aiff");
        Render::saveimagefile(osc11,"22-osc8.png");

        // Update progress
        bar.set_progress(31);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator12 @ 660 Hz 12/35"});

        SignalGenerators::generatesinewave(osc12,duration,660);
```

```
        SignalGenerators::gain(osc12,0.5);
        SignalGenerators::saveaudiofile(osc12,"23-osc8.aiff");
        Render::saveimagefile(osc12,"24-osc8.png");

        // Update progress
        bar.set_progress(34);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator13 @ 715 Hz 13/35"});

        SignalGenerators::generatesinewave(osc13,duration,715);
        SignalGenerators::gain(osc13,0.7);
        SignalGenerators::saveaudiofile(osc13,"25-osc8.aiff");
        Render::saveimagefile(osc13,"26-osc8.png");

        // Update progress
        bar.set_progress(37);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator14 @ 770 Hz 14/35"});

        SignalGenerators::generatesinewave(osc14,duration,770);
        SignalGenerators::gain(osc14,0.7);
        SignalGenerators::saveaudiofile(osc14,"27-osc8.aiff");
        Render::saveimagefile(osc14,"28-osc8.png");

        // Update progress
        bar.set_progress(40);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator15 @ 825 Hz 15/35"});

        SignalGenerators::generatesinewave(osc15,duration,825);
        SignalGenerators::gain(osc15,0.9);
        SignalGenerators::saveaudiofile(osc15,"29-osc8.aiff");
        Render::saveimagefile(osc15,"30-osc8.png");

        // Update progress
        bar.set_progress(43);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator16 @ 880 Hz 16/35"});

        SignalGenerators::generatesinewave(osc16,duration,880);
        SignalGenerators::gain(osc16,0.9);
        SignalGenerators::saveaudiofile(osc16,"31-osc8.aiff");
        Render::saveimagefile(osc16,"32-osc8.png");

        // Update progress
        bar.set_progress(46);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator17 @ 935 Hz 17/35"});
```

```
        SignalGenerators::generatesinewave(osc17,duration,935);
        SignalGenerators::gain(osc17,0.8);
        SignalGenerators::saveaudiofile(osc17,"33-osc8.aiff");
        Render::saveimagefile(osc17,"34-osc8.png");

        // Update progress
        bar.set_progress(49);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator18 @ 990 Hz 18/35"});

        SignalGenerators::generatesinewave(osc18,duration,990);
        SignalGenerators::gain(osc18,0.8);
        SignalGenerators::saveaudiofile(osc18,"35-osc8.aiff");
        Render::saveimagefile(osc18,"36-osc8.png");

        // Update progress
        bar.set_progress(51);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator19 @ 1045 Hz 19/35"});

        SignalGenerators::generatesinewave(osc19,duration,1045);
        SignalGenerators::gain(osc19,0.7);
        SignalGenerators::saveaudiofile(osc19,"37-osc8.aiff");
        Render::saveimagefile(osc19,"38-osc8.png");

        // Update progress
        bar.set_progress(54);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator20 @ 1100 Hz 20/35"});

        SignalGenerators::generatesinewave(osc20,duration,1100);
        SignalGenerators::gain(osc20,0.7);
        SignalGenerators::saveaudiofile(osc20,"39-osc8.aiff");
        Render::saveimagefile(osc20,"40-osc8.png");

        // Update progress
        bar.set_progress(57);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator21 @ 1155 Hz 21/35"});

        SignalGenerators::generatesinewave(osc21,duration,1155);
        SignalGenerators::gain(osc21,0.6);
        SignalGenerators::saveaudiofile(osc21,"41-osc8.aiff");
        Render::saveimagefile(osc21,"42-osc8.png");

        // Update progress
```

```
        bar.set_progress(60);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator22 @ 1210 Hz 22/35"});

        SignalGenerators::generatesinewave(osc22,duration,1210);
        SignalGenerators::gain(osc22,0.6);
        SignalGenerators::saveaudiofile(osc22,"43-osc8.aiff");
        Render::saveimagefile(osc22,"44-osc8.png");

        // Update progress
        bar.set_progress(63);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator23 @ 1265 Hz 23/35"});

        SignalGenerators::generatesinewave(osc23,duration,1265);
        SignalGenerators::gain(osc23,0.5);
        SignalGenerators::saveaudiofile(osc23,"45-osc8.aiff");
        Render::saveimagefile(osc23,"46-osc8.png");

        // Update progress
        bar.set_progress(66);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator24 @ 1320 Hz 24/35"});

        SignalGenerators::generatesinewave(osc24,duration,1320);
        SignalGenerators::gain(osc24,0.5);
        SignalGenerators::saveaudiofile(osc24,"47-osc8.aiff");
        Render::saveimagefile(osc24,"48-osc8.png");

        // Update progress
        bar.set_progress(69);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator25 @ 1375 Hz 25/35"});

        SignalGenerators::generatesinewave(osc25,duration,1375);
        SignalGenerators::gain(osc25,0.4);
        SignalGenerators::saveaudiofile(osc25,"49-osc8.aiff");
        Render::saveimagefile(osc25,"50-osc8.png");

        // Update progress
        bar.set_progress(71);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator26 @ 1430 Hz 26/11"});

        SignalGenerators::generatesinewave(osc26,duration,1430);
        SignalGenerators::gain(osc26,0.4);
        SignalGenerators::saveaudiofile(osc26,"51-osc8.aiff");
        Render::saveimagefile(osc26,"52-osc8.png");
```

```

        // Update progress
        bar.set_progress(74);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator27 @ 1485 Hz 27/35"});

        SignalGenerators::generatesinewave(osc27,duration,1485);
        SignalGenerators::gain(osc27,0.3);
        SignalGenerators::saveaudiofile(osc27,"53-osc8.aiff");
        Render::saveimagefile(osc27,"54-osc8.png");

        // Update progress
        bar.set_progress(77);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator28 @ 1540 Hz 28/35"});

        SignalGenerators::generatesinewave(osc28,duration,1540);
        SignalGenerators::gain(osc28,0.3);
        SignalGenerators::saveaudiofile(osc28,"55-osc8.aiff");
        Render::saveimagefile(osc28,"56-osc8.png");

        // Update progress
        bar.set_progress(80);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator29 @ 1595 Hz 29/35"});

        SignalGenerators::generatesinewave(osc29,duration,1595);
        SignalGenerators::gain(osc29,0.3);
        SignalGenerators::saveaudiofile(osc29,"57-osc8.aiff");
        Render::saveimagefile(osc29,"58-osc8.png");

        // Update progress
        bar.set_progress(83);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator30 @ 1650 Hz 30/35"});

        SignalGenerators::generatesinewave(osc30,duration,1650);
        SignalGenerators::gain(osc30,0.3);
        SignalGenerators::saveaudiofile(osc30,"59-osc8.aiff");
        Render::saveimagefile(osc30,"60-osc8.png");

        // Update progress
        bar.set_progress(86);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator31 @ 1705 Hz 31/35"});

        SignalGenerators::generatesinewave(osc31,duration,1705);
        SignalGenerators::gain(osc31,0.3);
```

```
        SignalGenerators::saveaudiofile(osc31,"61-osc8.aiff");
        Render::saveimagefile(osc31,"62-osc8.png");

        // Update progress
        bar.set_progress(89);
        bar.set_option(option::PostfixText{"Setting: Sine Oscillator32 @ 1760 Hz 32/35"});

        SignalGenerators::generatesinewave(osc32,duration,1760);
        SignalGenerators::gain(osc32,0.3);
        SignalGenerators::saveaudiofile(osc32,"63-osc8.aiff");
        Render::saveimagefile(osc32,"64-osc8.png");

        // Update progress
        bar.set_progress(91);
        bar.set_option(option::PostfixText{"Setting: Noise Oscillator33 33/35"});

        SignalGenerators::generatenoise(osc33,duration);
        SignalGenerators::gain(osc33,0.1);
        SignalGenerators::saveaudiofile(osc33,"65-osc9.aiff");
        Render::saveimagefile(osc33,"66-osc9.png");

        // Update progress
        bar.set_progress(94);
        bar.set_option(option::PostfixText{"Mixing all 33 oscillators 34/35"});

        mixer.resize(duration*sampleRate);
        SignalGenerators::addwaves(osc1,osc2,mixer);
        SignalGenerators::addwaves(mixer,osc3,mixer);
        SignalGenerators::addwaves(mixer,osc4,mixer);
        SignalGenerators::addwaves(mixer,osc5,mixer);
        SignalGenerators::addwaves(mixer,osc6,mixer);
        SignalGenerators::addwaves(mixer,osc7,mixer);
        SignalGenerators::addwaves(mixer,osc8,mixer);
        SignalGenerators::addwaves(mixer,osc9,mixer);
        SignalGenerators::addwaves(mixer,osc10,mixer);
        SignalGenerators::addwaves(mixer,osc11,mixer);
        SignalGenerators::addwaves(mixer,osc12,mixer);
        SignalGenerators::addwaves(mixer,osc13,mixer);
        SignalGenerators::addwaves(mixer,osc14,mixer);
        SignalGenerators::addwaves(mixer,osc15,mixer);
        SignalGenerators::addwaves(mixer,osc16,mixer);
        SignalGenerators::addwaves(mixer,osc17,mixer);
        SignalGenerators::addwaves(mixer,osc18,mixer);
```

```
        SignalGenerators::addwaves(mixer,osc19,mixer);
        SignalGenerators::addwaves(mixer,osc20,mixer);
        SignalGenerators::addwaves(mixer,osc21,mixer);
        SignalGenerators::addwaves(mixer,osc22,mixer);
        SignalGenerators::addwaves(mixer,osc23,mixer);
        SignalGenerators::addwaves(mixer,osc24,mixer);
        SignalGenerators::addwaves(mixer,osc25,mixer);
        SignalGenerators::addwaves(mixer,osc26,mixer);
        SignalGenerators::addwaves(mixer,osc27,mixer);
        SignalGenerators::addwaves(mixer,osc28,mixer);
        SignalGenerators::addwaves(mixer,osc29,mixer);
        SignalGenerators::addwaves(mixer,osc30,mixer);
        SignalGenerators::addwaves(mixer,osc31,mixer);
        SignalGenerators::addwaves(mixer,osc32,mixer);
        SignalGenerators::addwaves(mixer,osc33,mixer);
        SignalGenerators::normalize(mixer);
        SignalGenerators::saveaudiofile(mixer,"67-mixed.aiff");
        Render::saveimagefile(mixer,"68-mixed.png");

        // Update progress
        bar.set_progress(97);
        bar.set_option(option::PostfixText{"Adding ADSR Envelope 35/35"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        env.applyenvelope(mixer,envelope);
        SignalGenerators::normalize(mixer);
        Render::saveenvelopeimage(envelope,"69-envelope.png");
        SignalGenerators::saveaudiofile(mixer,"70-mixednenveloped.aiff");
        Render::saveimagefile(mixer,"71-mixednenveloped.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 35/35"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
```

```
    {
        for (uint32_t i=0;i<v.size();++i) {
            v[i]=v[i]*gain;
        }
    }

    void normalize(std::vector<float>& v)
    {
        float max=0.0,value=0.0;
        for (uint32_t i=0;i<v.size();++i) {
            value=v[i];
            if (value > max) {max=value;}
        }
        // max=std::ceil(max);
        for (uint32_t i=0;i<v.size();++i) {
            // v[i]=v[i]/max;
            v[i]=(v[i]/max)*0.707;
        }
    }

    void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
    {
        for (uint32_t i=0;i<v1.size();++i)
        {
            v3[i]=v1[i]+v2[i];
        }
    }

    void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz)
    {
        const double sampleRate=44100.0;
        for (uint32_t i=0;i<sampleRate*duration;++i) {
            v.push_back(sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2.0
PI));
        }
    }

    void generatenoise(std::vector<float>& v, int duration)
    {
        const double sampleRate=44100.0;
        std::random_device rd;
        std::mt19937 gen(rd());
        std::uniform_real_distribution<> dis(-1.0, 1.0);
```

```

		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(dis(gen));
		}
	}

	void saveaudiofile(std::vector<float>& v, std::string filename)
	{
		const std::string path="additive/";
		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=v[i];
			}
		}
		a.save(path+filename,AudioFileFormat::Aiff);
	}
}

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image)
	{
		png::rgb_pixel px(0x04,0x13,0x31);
		for (uint32_t y=0;y<image.get_height();y++) {
			for (uint32_t x=0;x<image.get_width();++x) {
				image.set_pixel(x,y,px);
			}
		}
	}

	void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
	{
		if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
		{
```

```
                png::rgb_pixel px(0x7a,0xb1,0xe3);
                image.set_pixel(x,y,px);
            }
    }

    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
        int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
        dx = x2 - x1; dy = y2 - y1;
        if (dx == 0)
        {
            if (y2 < y1) std::swap(y1, y2);
            for (y = y1; y <= y2; y++)
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
        {
            if (x2 < x1) std::swap(x1, x2);
            for (x = x1; x <= x2; x++)
                drawpx(image, x, y1);
            return;
        }
        dx1 = abs(dx); dy1 = abs(dy);
        px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
        if (dy1 <= dx1)
        {
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
```

```
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
            ox=x;oy=y;
        }
    }
```

```

	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
	{
		uint32_t halfHeight=image.get_height()/2;
		double value=0.0;
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}

		for (uint32_t i=0;i<v1.size();++i)
		{
			value=v1[i];
			if (value >= 0.0) {
				v2[i]=halfHeight-(halfHeight*value);
			} else if (value < 0.0)
			{
				v2[i]=halfHeight+(halfHeight*fabs(value));
			}
		}
	}

	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
	{
		fillbackground(image);
		drawwave(image,v);
	}

	void saveimagefile(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="additive/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="additive/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
```

```
			image.write(path+filename);
		}

		void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
		{
			uint32_t height=image.get_height();
			if (v2.size() == 0 || v2.size() > v1.size()) {
				v2.resize(v1.size());
			}
			for (uint32_t i=0;i<v1.size();++i)
			{
				v2[i]=height-(height*v1[i]);
			}
		}

		void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
		{
			const std::string path="additive/";
			png::image<png::rgb_pixel> image(44100,600);
			renderimage(image,v);
			image.write(path+filename);
		}

		void saveenvelopeimage(std::vector<float>& v2, std::string filename)
		{
			const std::string path="additive/";
			png::image<png::rgb_pixel> image(44100,600);
			std::vector<uint32_t> v;
			normalizedenvelopetoimg(image,v2,v);
			renderimage(image,v);
			image.write(path+filename);
		}

}
```

02-subtractive.cpp - 14546 bytes.

```
// compile: clang++ -std=c++20 -lpng 02-subtractive.cpp -o 02-subtractive

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "MoogFilter.hpp"
#include "Envelope.hpp"


#include <iostream>

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatetrianglewave(std::vector<float>& v, int duration, double frequencyInHz\
);
        void generateinversesawtoothwave(std::vector<float>& v, int duration, double freque\
ncyInHz);
```

```
        void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz\
);
        void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz);
        void generatethirtyfivesquarewave(std::vector<float>& v, int duration, double frequ\
encyInHz);
        void generatetwentyfivesquarewave(std::vector<float>& v, int duration, double frequ\
encyInHz);
        void generatenoise(std::vector<float>& v, int duration);
        void generatelfo(std::vector<float>& v, int osc, int duration, double frequencyInHz\
);
        void generateoscillator(std::vector<float>& v, int osc, int duration, double freque\
ncyInHz);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("subtractive");

        enum osc: int { triangle, inversesaw, saw, square, thirty, twenty, noise };
        const int duration=1;
        const double sampleRate=44100.0;
        const double noiseGain=0.5;
        std::vector<float> osc1;
        std::vector<float> osc2;
        std::vector<float> osc3;
        std::vector<float> osc4;
        std::vector<float> mixer;
        std::vector<float> envelope;
        Moog::MoogFilter mf(44100.0);

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
```

```
                option::End{" ]"},
                option::PostfixText{"Setting: Oscillator 1 to Triangle Wave @ 440 Hz 1/8"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::generateoscillator(osc1,triangle,duration,440);

        // Update progress
        bar.set_progress(12);
        bar.set_option(option::PostfixText{"Setting: Oscillator 2 to Triangle Wave @ 880 Hz\
 2/8"});

        SignalGenerators::generateoscillator(osc2,triangle,duration,440*2);

        // Update progress
        bar.set_progress(25);
        bar.set_option(option::PostfixText{"Setting: Oscillator 3 to Triangle Wave @ 1320 H\
z 3/8"});

        SignalGenerators::generateoscillator(osc3,triangle,duration,440*3);

        // Update progress
        bar.set_progress(37);
        bar.set_option(option::PostfixText{"Setting: Oscillator 4 (The White Noise Oscillat\
or) 4/8"});

        SignalGenerators::generateoscillator(osc4,noise,duration,440);

        // SignalGenerators::generatelfo(lfo,inversesaw,duration,15);
        // SignalGenerators::addwaves(osc3,lfo,osc3);

        SignalGenerators::gain(osc4,noiseGain);

        // Update progress
        bar.set_progress(50);
        bar.set_option(option::PostfixText{"Setting Oscillator 4 Gain to: 50% 5/8"});

        mixer.resize(duration*sampleRate);
        SignalGenerators::addwaves(osc1,osc2,mixer);
```

```
        SignalGenerators::normalize(mixer);
        SignalGenerators::addwaves(mixer,osc3,mixer);
        SignalGenerators::normalize(mixer);
        Render::saveimagefile(mixer,"01-mixerpreosc4.png");
        SignalGenerators::saveaudiofile(mixer,"02-mixerpreosc4.aiff");
        SignalGenerators::addwaves(mixer,osc4,mixer);
        SignalGenerators::normalize(mixer);
        Render::saveimagefile(mixer,"03-mixerpostosc4.png");
        SignalGenerators::saveaudiofile(mixer,"04-mixerpostosc4.aiff");

        // Update progress
        bar.set_progress(62);
        bar.set_option(option::PostfixText{"Mixed Oscillator 1 to 4 together 6/8"});

        Render::saveimagefile(mixer,"05-prefilter.png");
        SignalGenerators::saveaudiofile(mixer,"06-prefilter.aiff");

        mf.Process(mixer,mixer.size());

        SignalGenerators::normalize(mixer);
        Render::saveimagefile(mixer,"07-postfilter.png");
        SignalGenerators::saveaudiofile(mixer,"08-postfilter.aiff");

        // Update progress
        bar.set_progress(75);
        bar.set_option(option::PostfixText{"Adding MoogFilter to the mixed signal 7/8"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        Render::saveenvelopeimage(envelope,"09-envelope.png");

        // Update progress
        bar.set_progress(87);
        bar.set_option(option::PostfixText{"Adding ADSR Envelope to the mixed signal 8/8"});

        env.applyenvelope(mixer,envelope);
        Render::saveimagefile(mixer,"10-envelopedmix.png");
        SignalGenerators::saveaudiofile(mixer,"11-envelopedmix.aiff");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 8/8"});

```

```
        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        void generatetrianglewave(std::vector<float>& v, int duration, double frequencyInHz)
        {
                const double sampleRate=44100.0;

                for (uint32_t i=0;i<sampleRate*duration;++i) {
                                v.push_back(M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate)));
                }
```

```
	}

	void generateinversesawtoothwave(std::vector<float>& v, int duration, double freque\
ncyInHz)
	{
		const double sampleRate=44100.0;
		double period=sampleRate/frequencyInHz;
		std::vector<float> temp;

		for (uint32_t i=0;i<period;++i) {
			temp.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
		}
		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(temp[temp.size()-(i%temp.size())]);
		}
	}

	void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz)
	{
		const double sampleRate=44100.0;

		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
		}
	}

	void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz)
	{
		const double sampleRate=44100.0;
		double period=sampleRate/frequencyInHz;
		double dutyCycle=period*0.5;
		double ss=0.0;

		for (uint32_t i=0;i<sampleRate*duration;++i) {
			if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
			{
				ss=0.7;
			} else {
				ss=-0.7;
			}
			v.push_back(ss);
		}
		v[0]=0.0;
```

```
			v[v.size()-1]=0.0;
		}

		void generatethirtyfivesquarewave(std::vector<float>& v, int duration, double frequ\
encyInHz)
		{
			const double sampleRate=44100.0;
			double period=sampleRate/frequencyInHz;
			double dutyCycle=period*0.35;
			double ss;

			for (uint32_t i=0;i<sampleRate*duration;++i) {
				if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
				{
					ss=0.7;
				} else {
					ss=-0.7;
				}
				v.push_back(ss);
			}
		}

		void generatetwentyfivesquarewave(std::vector<float>& v, int duration, double frequ\
encyInHz)
		{
			const double sampleRate=44100.0;
			double period=sampleRate/frequencyInHz;
			double dutyCycle=period*0.25;
			double ss;

			for (uint32_t i=0;i<sampleRate*duration;++i) {
				if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
				{
					ss=0.7;
				} else {
					ss=-0.7;
				}
				v.push_back(ss);
			}
		}

		void generatenoise(std::vector<float>& v, int duration)
		{
```

```
        const double sampleRate=44100.0;
        std::random_device rd;
        std::mt19937 gen(rd());
        std::uniform_real_distribution<> dis(-1.0, 1.0);

        for (uint32_t i=0;i<sampleRate*duration;++i) {
            v.push_back(dis(gen));
        }
    }

    void generatelfo(std::vector<float>& v, int osc, int duration, double frequencyInHz)
    {
        if (frequencyInHz >= 0.0 && frequencyInHz <= 20.0) {
            switch(osc) {
                case 0: generatetrianglewave(v,duration,frequencyInHz);
                        break;
                case 1: generateinversesawtoothwave(v,duration,frequencyInHz);
                        break;
                case 2: generatesawtoothwave(v,duration,frequencyInHz);
                        break;
                case 3: generatesquarewave(v,duration,frequencyInHz);
                        break;
                case 4: generatethirtyfivesquarewave(v,duration,frequencyInHz);
                        break;
                case 5: generatetwentyfivesquarewave(v,duration,frequencyInHz);
                        break;
            }
        }
    }

    void generateoscillator(std::vector<float>& v, int osc, int duration, double freque\
ncyInHz)
    {
        if (frequencyInHz >= 20.0 && frequencyInHz <= 20000.0) {
            switch(osc) {
                case 0: generatetrianglewave(v,duration,frequencyInHz);
                        break;
                case 1: generateinversesawtoothwave(v,duration,frequencyInHz);
                        break;
                case 2: generatesawtoothwave(v,duration,frequencyInHz);
                        break;
                case 3: generatesquarewave(v,duration,frequencyInHz);
                        break;
```

```
                                case 4: generatethirtyfivesquarewave(v,duration,frequencyInHz);
                                                break;
                                case 5: generatetwentyfivesquarewave(v,duration,frequencyInHz);
                                                break;
                                case 6: generatenoise(v,duration);
                                                break;
                        }
                }
        }

        void saveaudiofile(std::vector<float>& v, std::string filename)
        {
                const std::string path="subtractive/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                a.samples[channel][i]=v[i];
                        }
                }
                a.save(path+filename,AudioFileFormat::Aiff);
        }
}

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image)
        {
                png::rgb_pixel px(0x04,0x13,0x31);
                for (uint32_t y=0;y<image.get_height();y++) {
                        for (uint32_t x=0;x<image.get_width();++x) {
                                image.set_pixel(x,y,px);
                        }
                }
        }

        void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
```

```
        {
                if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
                {
                        png::rgb_pixel px(0x7a,0xb1,0xe3);
                        image.set_pixel(x,y,px);
                }
        }

        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
        {
                int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
                dx = x2 - x1; dy = y2 - y1;
                if (dx == 0)
                {
                        if (y2 < y1) std::swap(y1, y2);
                        for (y = y1; y <= y2; y++)
                                drawpx(image, x1, y);
                        return;
                }
                if (dy == 0)
                {
                        if (x2 < x1) std::swap(x1, x2);
                        for (x = x1; x <= x2; x++)
                                drawpx(image, x, y1);
                        return;
                }
                dx1 = abs(dx); dy1 = abs(dy);
                px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
                if (dy1 <= dx1)
                {
                        if (dx >= 0)
                        {
                                x = x1; y = y1; xe = x2;
                        }
                        else
                        {
                                x = x2; y = y2; xe = x1;
                        }
                        drawpx(image, x, y);
                        for (i = 0; x<xe; i++)
                        {
                                x = x + 1;
```

```
                    if (px<0)
                        px = px + 2 * dy1;
                    else
                    {
                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                        px = px + 2 * (dy1 - dx1);
                    }
                    drawpx(image, x, y);
                }
            }
            else
            {
                if (dy >= 0)
                {
                    x = x1; y = y1; ye = y2;
                }
                else
                {
                    x = x2; y = y2; ye = y1;
                }
                drawpx(image, x, y);
                for (i = 0; y<ye; i++)
                {
                    y = y + 1;
                    if (py <= 0)
                        py = py + 2 * dx1;
                    else
                    {
                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                        py = py + 2 * (dx1 - dy1);
                    }
                    drawpx(image, x, y);
                }
            }
        }

        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
        {
            uint32_t y=0,ox=0,oy=0;
            for (uint32_t x=0;x<image.get_width();++x)
            {
                y=signalY[x];
                if (x == 0) {ox=x;oy=y;}
```

```
                                drawline(image,x,y,ox,oy);
                                ox=x;oy=y;
                        }
        }

        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
        {
                uint32_t halfHeight=image.get_height()/2;
                double value=0.0;
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }

                for (uint32_t i=0;i<v1.size();++i)
                {
                        value=v1[i];
                        if (value >= 0.0) {
                                v2[i]=halfHeight-(halfHeight*value);
                        } else if (value < 0.0)
                        {
                                v2[i]=halfHeight+(halfHeight*fabs(value));
                        }
                }
        }

        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
        {
                fillbackground(image);
                drawwave(image,v);
        }

        void saveimagefile(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="subtractive/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveimagefile(std::vector<float>& v2, std::string filename)
        {
                const std::string path="subtractive/";
```

```
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedtoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
        {
                uint32_t height=image.get_height();
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v2[i]=height-(height*v1[i]);
                }
        }

        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="subtractive/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="subtractive/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

03-formant.cpp - 16411 bytes.

```cpp
// compile: clang++ -std=c++20 -lpng 03-formant.cpp -o 03-formant

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz);
        void generatenoise(std::vector<float>& v, int duration);
        void generateformant(std::vector<float>& v, int formantnumber, int duration);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
```

```
        namespace fs = std::filesystem;
        fs::create_directory("formant");

        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> formant0;
        std::vector<float> formant1;
        std::vector<float> formant2;
        std::vector<float> formant3;
        std::vector<float> formant4;
        std::vector<float> formant5;
        std::vector<float> formant6;
        std::vector<float> formant7;
        std::vector<float> formant8;
        std::vector<float> formant9;
        std::vector<float> formant10;
        std::vector<float> envelope;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Generate Formant0 1/23"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        // Vowel [i]
        SignalGenerators::generateformant(formant0,0,duration);
        SignalGenerators::saveaudiofile(formant0,"01-formant0-vowel-i.aiff");
        Render::saveimagefile(formant0,"02-formant0-vowel-i.png");

```

```
        // Update progress
        bar.set_progress(4);
        bar.set_option(option::PostfixText{"Generate Formant1 2/23"});

        // Vowel [□]
        SignalGenerators::generateformant(formant1,1,duration);
        SignalGenerators::saveaudiofile(formant1,"03-formant1-vowel-□.aiff");
        Render::saveimagefile(formant0,"04-formant1-vowel-□.png");

        // Update progress
        bar.set_progress(9);
        bar.set_option(option::PostfixText{"Generate Formant2 3/23"});

        // Vowel [e]
        SignalGenerators::generateformant(formant2,2,duration);
        SignalGenerators::saveaudiofile(formant2,"05-formant2-vowel-e.aiff");
        Render::saveimagefile(formant0,"06-formant2-vowel-e.png");

        // Update progress
        bar.set_progress(13);
        bar.set_option(option::PostfixText{"Generate Formant3 4/23"});

        // Vowel [□]
        SignalGenerators::generateformant(formant3,3,duration);
        SignalGenerators::saveaudiofile(formant3,"07-formant3-vowel-□.aiff");
        Render::saveimagefile(formant0,"08-formant3-vowel-□.png");

        // Update progress
        bar.set_progress(17);
        bar.set_option(option::PostfixText{"Generate Formant4 5/23"});

        // Vowel [æ]
        SignalGenerators::generateformant(formant4,4,duration);
        SignalGenerators::saveaudiofile(formant4,"09-formant4-vowel-æ.aiff");
        Render::saveimagefile(formant0,"10-formant4-vowel-æ.png");

        // Update progress
        bar.set_progress(22);
        bar.set_option(option::PostfixText{"Generate Formant5 6/23"});

        // Vowel [□]
        SignalGenerators::generateformant(formant5,5,duration);
        SignalGenerators::saveaudiofile(formant5,"11-formant5-vowel-□.aiff");
```

```
        Render::saveimagefile(formant0,"12-formant5-vowel-□.png");

        // Update progress
        bar.set_progress(26);
        bar.set_option(option::PostfixText{"Generate Formant6 7/23"});

        // Vowel [□]
        SignalGenerators::generateformant(formant6,6,duration);
        SignalGenerators::saveaudiofile(formant6,"13-formant6-vowel-□.aiff");
        Render::saveimagefile(formant0,"14-formant6-vowel-□.png");

        // Update progress
        bar.set_progress(30);
        bar.set_option(option::PostfixText{"Generate Formant7 8/23"});

        // Vowel [o]
        SignalGenerators::generateformant(formant7,7,duration);
        SignalGenerators::saveaudiofile(formant7,"15-formant7-vowel-o.aiff");
        Render::saveimagefile(formant0,"16-formant7-vowel-o.png");

        // Update progress
        bar.set_progress(35);
        bar.set_option(option::PostfixText{"Generate Formant8 9/23"});

        // Vowel [□]
        SignalGenerators::generateformant(formant8,8,duration);
        SignalGenerators::saveaudiofile(formant8,"17-formant8-vowel-□.aiff");
        Render::saveimagefile(formant0,"18-formant8-vowel-□.png");

        // Update progress
        bar.set_progress(39);
        bar.set_option(option::PostfixText{"Generate Formant9 10/23"});

        // Vowel [u]
        SignalGenerators::generateformant(formant9,9,duration);
        SignalGenerators::saveaudiofile(formant9,"19-formant9-vowel-u.aiff");
        Render::saveimagefile(formant0,"20-formant9-vowel-u.png");

        // Update progress
        bar.set_progress(43);
        bar.set_option(option::PostfixText{"Generate Formant10 11/23"});

        // Vowel [□]
```

```
        SignalGenerators::generateformant(formant10,10,duration);
        SignalGenerators::saveaudiofile(formant10,"21-formant10-vowel-□.aiff");
        Render::saveimagefile(formant0,"22-formant10-vowel-□.png");

        // Update progress
        bar.set_progress(48);
        bar.set_option(option::PostfixText{"Generate Envelope 12/23"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        Render::saveenvelopeimage(envelope,"23-envelope.png");

        // Update progress
        bar.set_progress(52);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant0 13/23"});

        env.applyenvelope(formant0,envelope);
        SignalGenerators::normalize(formant0);
        SignalGenerators::saveaudiofile(formant0,"24-env-formant0-vowel-i.aiff");
        Render::saveimagefile(formant0,"25-env-formant0-vowel-i.png");

        // Update progress
        bar.set_progress(57);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant1 14/23"});

        env.applyenvelope(formant1,envelope);
        SignalGenerators::normalize(formant1);
        SignalGenerators::saveaudiofile(formant1,"26-env-formant1-vowel-□.aiff");
        Render::saveimagefile(formant1,"27-env-formant1-vowel-□.png");

        // Update progress
        bar.set_progress(61);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant2 15/23"});

        env.applyenvelope(formant2,envelope);
        SignalGenerators::normalize(formant2);
        SignalGenerators::saveaudiofile(formant2,"28-env-formant2-vowel-e.aiff");
        Render::saveimagefile(formant2,"29-env-formant2-vowel-e.png");

        // Update progress
        bar.set_progress(65);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant3 16/23"});

```

```
        env.applyenvelope(formant3,envelope);
        SignalGenerators::normalize(formant3);
        SignalGenerators::saveaudiofile(formant3,"30-env-formant3-vowel-□.aiff");
        Render::saveimagefile(formant3,"31-env-formant3-vowel-□.png");

        // Update progress
        bar.set_progress(70);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant4 17/23"});

        env.applyenvelope(formant4,envelope);
        SignalGenerators::normalize(formant4);
        SignalGenerators::saveaudiofile(formant4,"32-env-formant4-vowel-æ.aiff");
        Render::saveimagefile(formant4,"33-env-formant4-vowel-æ.png");

        // Update progress
        bar.set_progress(74);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant5 18/23"});

        env.applyenvelope(formant5,envelope);
        SignalGenerators::normalize(formant5);
        SignalGenerators::saveaudiofile(formant5,"34-env-formant5-vowel-□.aiff");
        Render::saveimagefile(formant5,"35-env-formant5-vowel-□.png");

        // Update progress
        bar.set_progress(78);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant6 19/23"});

        env.applyenvelope(formant6,envelope);
        SignalGenerators::normalize(formant6);
        SignalGenerators::saveaudiofile(formant6,"36-env-formant6-vowel-□.aiff");
        Render::saveimagefile(formant6,"37-env-formant6-vowel-□.png");

        // Update progress
        bar.set_progress(83);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant7 20/23"});

        env.applyenvelope(formant7,envelope);
        SignalGenerators::normalize(formant7);
        SignalGenerators::saveaudiofile(formant7,"38-env-formant7-vowel-o.aiff");
        Render::saveimagefile(formant7,"39-env-formant7-vowel-o.png");

        // Update progress
        bar.set_progress(87);
```

```
        bar.set_option(option::PostfixText{"Apply Envelope to Formant8 21/23"});

        env.applyenvelope(formant8,envelope);
        SignalGenerators::normalize(formant8);
        SignalGenerators::saveaudiofile(formant8,"40-env-formant8-vowel-□.aiff");
        Render::saveimagefile(formant8,"41-env-formant8-vowel-□.png");

        // Update progress
        bar.set_progress(91);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant9 22/23"});

        env.applyenvelope(formant9,envelope);
        SignalGenerators::normalize(formant9);
        SignalGenerators::saveaudiofile(formant9,"42-env-formant9-vowel-u.aiff");
        Render::saveimagefile(formant9,"43-env-formant9-vowel-u.png");

        // Update progress
        bar.set_progress(96);
        bar.set_option(option::PostfixText{"Apply Envelope to Formant10 23/23"});

        env.applyenvelope(formant10,envelope);
        SignalGenerators::normalize(formant10);
        SignalGenerators::saveaudiofile(formant10,"44-env-formant10-vowel-□.aiff");
        Render::saveimagefile(formant10,"45-env-formant10-vowel-□.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 23/23"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

```

```
	void normalize(std::vector<float>& v)
	{
		float max=0.0,value=0.0;
		for (uint32_t i=0;i<v.size();++i) {
			value=v[i];
			if (value > max) {max=value;}
		}
		// max=std::ceil(max);
		for (uint32_t i=0;i<v.size();++i) {
			// v[i]=v[i]/max;
			v[i]=(v[i]/max)*0.707;
		}
	}

	void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
	{
		for (uint32_t i=0;i<v1.size();++i)
		{
			v3[i]=v1[i]+v2[i];
		}
	}

	void generatesinewave(std::vector<float>& v, int duration, float frequencyInHz)
	{
		const double sampleRate=44100.0;
		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(sin((static_cast<double> (i) / sampleRate) * frequencyInHz * 2.
PI));
		}
	}

	void generatenoise(std::vector<float>& v, int duration)
	{
		const double sampleRate=44100.0;
		std::random_device rd;
		std::mt19937 gen(rd());
		std::uniform_real_distribution<> dis(-1.0, 1.0);

		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(dis(gen));
		}
	}
```

```
    void generateformant(std::vector<float>& mixer, int formantnumber, int duration)
    {
        // Vowel        [i]         [□]         [e]         [□]         [æ]         [□]
        // F1            280         370         405         600         860         830
        // F2            2230         2090         2080         1930         1550         1
        const double sampleRate=44100.0;
        float f1[]={280.0,370.0,405.0,600.0,860.0,830.0,560.0,430.0,400.0,330.0,680.0};
        float f2[]={2230.0,2090.0,2080.0,1930.0,1550.0,1170.0,820.0,980.0,1100.0,1260.0,13\
10};
        mixer.resize(duration*sampleRate);
        std::vector<float> osc1;
        std::vector<float> osc2;
        std::vector<float> osc3;
        generatesinewave(osc1,duration,f1[formantnumber]);
        generatesinewave(osc2,duration,f2[formantnumber]);
        generatenoise(osc3,duration);
        gain(osc3,0.2);
        addwaves(osc1,osc2,mixer);
        addwaves(mixer,osc3,mixer);
        normalize(mixer);
    }

    void saveaudiofile(std::vector<float>& v, std::string filename)
    {
        const std::string path="formant/";
        // Setup the audio file
        AudioFile<float> a;
        a.setNumChannels(1);
        a.setBitDepth(24);
        a.setNumSamplesPerChannel(44100);

        for (int i=0;i<a.getNumSamplesPerChannel();++i)
        {
            for (int channel=0;channel<a.getNumChannels();++channel)
            {
                a.samples[channel][i]=v[i];
            }
        }
        a.save(path+filename,AudioFileFormat::Aiff);
    }
}

namespace Render
```

```
{
	void fillbackground(png::image<png::rgb_pixel>& image)
	{
		png::rgb_pixel px(0x04,0x13,0x31);
		for (uint32_t y=0;y<image.get_height();y++) {
			for (uint32_t x=0;x<image.get_width();++x) {
				image.set_pixel(x,y,px);
			}
		}
	}

	void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
	{
		if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
		{
			png::rgb_pixel px(0x7a,0xb1,0xe3);
			image.set_pixel(x,y,px);
		}
	}

	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
	{
		int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
		dx = x2 - x1; dy = y2 - y1;
		if (dx == 0)
		{
			if (y2 < y1) std::swap(y1, y2);
			for (y = y1; y <= y2; y++)
				drawpx(image, x1, y);
			return;
		}
		if (dy == 0)
		{
			if (x2 < x1) std::swap(x1, x2);
			for (x = x1; x <= x2; x++)
				drawpx(image, x, y1);
			return;
		}
		dx1 = abs(dx); dy1 = abs(dy);
		px = 2 * dy1 - dx1;	py = 2 * dx1 - dy1;
		if (dy1 <= dx1)
		{
```

```
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
```

```
                                drawpx(image, x, y);
                        }
                }
        }

        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
        {
                uint32_t y=0,ox=0,oy=0;
                for (uint32_t x=0;x<image.get_width();++x)
                {
                        y=signalY[x];
                        if (x == 0) {ox=x;oy=y;}
                        drawline(image,x,y,ox,oy);
                        ox=x;oy=y;
                }
        }

        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
        {
                uint32_t halfHeight=image.get_height()/2;
                double value=0.0;
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }

                for (uint32_t i=0;i<v1.size();++i)
                {
                        value=v1[i];
                        if (value >= 0.0) {
                                v2[i]=halfHeight-(halfHeight*value);
                        } else if (value < 0.0)
                        {
                                v2[i]=halfHeight+(halfHeight*fabs(value));
                        }
                }
        }

        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
        {
                fillbackground(image);
                drawwave(image,v);
        }
```

```

	void saveimagefile(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="formant/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="formant/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}

	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
v1,std::vector<uint32_t>& v2)
	{
		uint32_t height=image.get_height();
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}
		for (uint32_t i=0;i<v1.size();++i)
		{
			v2[i]=height-(height*v1[i]);
		}
	}

	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="formant/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveenvelopeimage(std::vector<float>& v2, std::string filename)
	{
		const std::string path="formant/";
```

```
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

04-granular.cpp - 9996 bytes.

```
// compile: clang++ -std=c++20 -lpng 04-granular.cpp -o 04-granular
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
```

```cpp
	void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
	void generatesample(std::vector<float>& v, int duration);
	void extendsample(std::vector<float>& v1, std::vector<float>& v2, int n);
	void saveaudiofile(std::vector<float>& v, std::string filename, int duration);
	void loadaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
	namespace fs = std::filesystem;
	fs::create_directory("granular");

	using namespace indicators;
	// Hide cursor
	show_console_cursor(false);

	// Setup ProgressBar
	ProgressBar bar{
		option::BarWidth{50},
		option::Start{"["},
		option::Fill{"□"},
		option::Lead{"□"},
		option::Remainder{"-"},
		option::End{" ]"},
		option::PostfixText{"Generate Sample 1/5"},
		option::ForegroundColor{Color::cyan},
		option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
	};

	// Update progress
	bar.set_progress(0);

	const int duration=1;
	std::vector<float> sample;
	std::vector<float> sample_extended;
	SignalGenerators::generatesample(sample,duration);
	SignalGenerators::saveaudiofile(sample,"01-generatedsample.aiff",1);
	Render::saveimagefile(sample,"02-generatedsample.png");

	// Update progress
	bar.set_progress(20);
	bar.set_option(option::PostfixText{"Extend Sample 2/5"});

```

```
        SignalGenerators::extendsample(sample,sample_extended,4);
        SignalGenerators::saveaudiofile(sample_extended,"03-extendedsample.aiff",4);

        // Update progress
        bar.set_progress(40);
        bar.set_option(option::PostfixText{"Get grains from sample 3/5"});

        // Get grains from sample.
        // Add grains to sample_extended.
        float grains[100][44];
        uint32_t ptr1=0,ptr2=0;
        std::random_device rd;
        std::mt19937 gen(rd());
        std::uniform_int_distribution<> distria(1, 990);
        std::uniform_int_distribution<> distrib(1, 4000);
        std::uniform_int_distribution<> distric(0, 99);

        for (uint32_t i=0;i<100;++i) {
                ptr1=distria(gen)*44;
                for (uint32_t j=0;j<44;++j) {
                        grains[i][j]=sample[ptr1++];
                }
        }

        // Update progress
        bar.set_progress(60);
        bar.set_option(option::PostfixText{"Add grains to sample_extended 4/5"});

        for (uint32_t i=0;i<800;++i) {
                ptr1=distric(gen);
                ptr2=distrib(gen)*44;
                for (uint32_t j=0;j<44;++j) {
                        sample_extended[ptr2++]=grains[ptr1][j];
                }
        }

        // Update progress
        bar.set_progress(80);
        bar.set_option(option::PostfixText{"Final Granular 5/5"});

        SignalGenerators::normalize(sample_extended);
        SignalGenerators::saveaudiofile(sample_extended,"04-finalgranular.aiff",4);

```

```
        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 5/5"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        void generatesample(std::vector<float>& v, int duration)
        {
                const double sampleRate=44100.0;
```

```
                std::vector<float> envelope1;
                std::vector<float> envelope2;
                std::vector<float> sample1;
                std::vector<float> sample2;

                ADSR::Envelope env(sampleRate,duration);
                env.generateenvelope2(envelope1);
                env.generateenvelope3(envelope2);

                loadaudiofile(sample1,"additive/67-mixed.aiff");
                loadaudiofile(sample2,"subtractive/08-postfilter.aiff");

                env.applyenvelope(sample1,envelope2);
                env.applyenvelope(sample2,envelope1);

                v.resize(duration*sampleRate);
                addwaves(sample1,sample2,v);
                normalize(v);
        }

        void extendsample(std::vector<float>& v1, std::vector<float>& v2, int n)
        {
                for (uint32_t i=0;i<v1.size();++i) {
                        for (uint32_t j=0;j<n;++j) {
                                v2.push_back(v1[i]);
                        }
                }
        }

        void saveaudiofile(std::vector<float>& v, std::string filename, int duration)
        {
                const std::string path="granular/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100*duration);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                a.samples[channel][i]=v[i];
```

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

        void loadaudiofile(std::vector<float>& v, std::string filename)
        {
            AudioFile<float> a;
        bool loadedOK = a.load(filename);
        if (loadedOK) {
            for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                    for (int channel=0;channel<1;++channel)
                    {
                        v.push_back(a.samples[channel][i]);
                    }
                }
        }
        }
}

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image)
        {
            png::rgb_pixel px(0x04,0x13,0x31);
            for (uint32_t y=0;y<image.get_height();y++) {
                for (uint32_t x=0;x<image.get_width();++x) {
                    image.set_pixel(x,y,px);
                }
            }
        }

        void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
        {
            if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
            {
                png::rgb_pixel px(0x7a,0xb1,0xe3);
                image.set_pixel(x,y,px);
            }
        }

```

```cpp
    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
        int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
        dx = x2 - x1; dy = y2 - y1;
        if (dx == 0)
        {
            if (y2 < y1) std::swap(y1, y2);
            for (y = y1; y <= y2; y++)
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
        {
            if (x2 < x1) std::swap(x1, x2);
            for (x = x1; x <= x2; x++)
                drawpx(image, x, y1);
            return;
        }
        dx1 = abs(dx); dy1 = abs(dy);
        px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
        if (dy1 <= dx1)
        {
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
```

```
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
            ox=x;oy=y;
        }
    }

    void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
    {
        uint32_t halfHeight=image.get_height()/2;
```

```
        double value=0.0;
        if (v2.size() == 0 || v2.size() > v1.size()) {
            v2.resize(v1.size());
        }

        for (uint32_t i=0;i<v1.size();++i)
        {
            value=v1[i];
            if (value >= 0.0) {
                v2[i]=halfHeight-(halfHeight*value);
            } else if (value < 0.0)
            {
                v2[i]=halfHeight+(halfHeight*fabs(value));
            }
        }
    }

    void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
    {
        fillbackground(image);
        drawwave(image,v);
    }

    void saveimagefile(std::vector<uint32_t>& v, std::string filename)
    {
        const std::string path="granular/";
        png::image<png::rgb_pixel> image(44100,600);
        renderimage(image,v);
        image.write(path+filename);
    }

    void saveimagefile(std::vector<float>& v2, std::string filename)
    {
        const std::string path="granular/";
        png::image<png::rgb_pixel> image(44100,600);
        std::vector<uint32_t> v;
        normalizedtoimg(image,v2,v);
        renderimage(image,v);
        image.write(path+filename);
    }

    void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
v1,std::vector<uint32_t>& v2)
```

```
        {
                uint32_t height=image.get_height();
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v2[i]=height-(height*v1[i]);
                }
        }

        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="granular/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="granular/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

05-fm.cpp - 8869 bytes.

```
// compile: clang++ -std=c++20 -lpng 05-fm.cpp -o 05-fm
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"
```

```

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image);
	void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
	void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
	void saveimagefile(std::vector<uint32_t>& v, std::string filename);
	void saveimagefile(std::vector<float>& v2, std::string filename);
	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
	void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
	void gain(std::vector<float>& v, double gain);
	void normalize(std::vector<float>& v);
	void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
	void generatefrequencymodulation(std::vector<float>& v, int duration, float frequen\
cyInHz);
	void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
	namespace fs = std::filesystem;
	fs::create_directory("fm");

	const int duration=1;
	const double sampleRate=44100.0;
	std::vector<float> fm;
	std::vector<float> envelope;

	using namespace indicators;
	// Hide cursor
	show_console_cursor(false);

	// Setup ProgressBar
```

```
    ProgressBar bar{
        option::BarWidth{50},
        option::Start{"["},
        option::Fill{"□"},
        option::Lead{"□"},
        option::Remainder{"-"},
        option::End{" ]"},
        option::PostfixText{"Generate Frequency Modulation (FM) 1/2"},
        option::ForegroundColor{Color::cyan},
        option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
    };

    // Update progress
    bar.set_progress(0);

    SignalGenerators::generatefrequencymodulation(fm,duration,440);
    SignalGenerators::normalize(fm);
    SignalGenerators::saveaudiofile(fm,"01-fm.aiff");
    Render::saveimagefile(fm,"02-fm.png");

    // Update progress
    bar.set_progress(50);
    bar.set_option(option::PostfixText{"Apply Envelope to FM 2/2"});

    ADSR::Envelope env(sampleRate,duration);
    env.generateenvelope(envelope);
    Render::saveenvelopeimage(envelope,"03-envelope.png");

    env.applyenvelope(fm,envelope);
    SignalGenerators::normalize(fm);
    SignalGenerators::saveaudiofile(fm,"04-final_fm.aiff");
    Render::saveimagefile(fm,"05-final_fm.png");

    // Update progress
    bar.set_progress(100);
    bar.set_option(option::PostfixText{"Done 1/2"});

    // Show cursor
    show_console_cursor(true);
    return 0;
}

namespace SignalGenerators
```

```
{
    void gain(std::vector<float>& v, double gain)
    {
        for (uint32_t i=0;i<v.size();++i) {
            v[i]=v[i]*gain;
        }
    }

    void normalize(std::vector<float>& v)
    {
        float max=0.0,value=0.0;
        for (uint32_t i=0;i<v.size();++i) {
            value=v[i];
            if (value > max) {max=value;}
        }
        // max=std::ceil(max);
        for (uint32_t i=0;i<v.size();++i) {
            // v[i]=v[i]/max;
            v[i]=(v[i]/max)*0.707;
        }
    }

    void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
    {
        for (uint32_t i=0;i<v1.size();++i)
        {
            v3[i]=v1[i]+v2[i];
        }
    }

    void generatefrequencymodulation(std::vector<float>& v, int duration, float frequen\
cyInHz)
    {
        const double twoPI=2*M_PI;
        const double sampleRate=44100.0;
        const double frequencyRadian = twoPI / sampleRate;
        double modulatorFrequency = frequencyInHz * 3;
        double modulatorIncrement = frequencyRadian * modulatorFrequency;
        double modulatorPhase = 0;
        double carrierIncrement = 0;
        double carrierPhase = 0;
        double modulatoramplitude = 2 * modulatorFrequency;
        double modulatorValue = 0;
```

```
		for (uint32_t i=0;i<sampleRate*duration;++i) {
			v.push_back(sinf(carrierPhase));
			modulatorValue = modulatoramplitude * sinf(modulatorPhase);
			carrierIncrement = frequencyRadian * (frequencyInHz + modulatorValue);
			carrierPhase = carrierPhase + carrierIncrement;
			modulatorPhase = modulatorPhase + modulatorIncrement;
			if (carrierPhase >= twoPI) {
				carrierPhase -= twoPI;
			}
			else if (carrierPhase < 0) {
				carrierPhase += twoPI;
			}
			if (modulatorPhase >= twoPI) {
				modulatorPhase -= twoPI;
			}
		}
	}

	void saveaudiofile(std::vector<float>& v, std::string filename)
	{
		const std::string path="fm/";
		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=v[i];
			}
		}
		a.save(path+filename,AudioFileFormat::Aiff);
	}
}

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image)
	{
```

```
        png::rgb_pixel px(0x04,0x13,0x31);
        for (uint32_t y=0;y<image.get_height();y++) {
            for (uint32_t x=0;x<image.get_width();++x) {
                image.set_pixel(x,y,px);
            }
        }
    }

    void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
    {
        if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
        {
            png::rgb_pixel px(0x7a,0xb1,0xe3);
            image.set_pixel(x,y,px);
        }
    }

    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
        int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
        dx = x2 - x1; dy = y2 - y1;
        if (dx == 0)
        {
            if (y2 < y1) std::swap(y1, y2);
            for (y = y1; y <= y2; y++)
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
        {
            if (x2 < x1) std::swap(x1, x2);
            for (x = x1; x <= x2; x++)
                drawpx(image, x, y1);
            return;
        }
        dx1 = abs(dx); dy1 = abs(dy);
        px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
        if (dy1 <= dx1)
        {
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
```

```
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
```

```
        }

        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
        {
                uint32_t y=0,ox=0,oy=0;
                for (uint32_t x=0;x<image.get_width();++x)
                {
                        y=signalY[x];
                        if (x == 0) {ox=x;oy=y;}
                        drawline(image,x,y,ox,oy);
                        ox=x;oy=y;
                }
        }

        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
        {
                uint32_t halfHeight=image.get_height()/2;
                double value=0.0;
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }

                for (uint32_t i=0;i<v1.size();++i)
                {
                        value=v1[i];
                        if (value >= 0.0) {
                                v2[i]=halfHeight-(halfHeight*value);
                        } else if (value < 0.0)
                        {
                                v2[i]=halfHeight+(halfHeight*fabs(value));
                        }
                }
        }

        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
        {
                fillbackground(image);
                drawwave(image,v);
        }

        void saveimagefile(std::vector<uint32_t>& v, std::string filename)
        {
```

```
		const std::string path="fm/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="fm/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}

	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
	{
		uint32_t height=image.get_height();
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}
		for (uint32_t i=0;i<v1.size();++i)
		{
			v2[i]=height-(height*v1[i]);
		}
	}

	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="fm/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveenvelopeimage(std::vector<float>& v2, std::string filename)
	{
		const std::string path="fm/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedenvelopetoimg(image,v2,v);
```

```
			renderimage(image,v);
			image.write(path+filename);
		}

}
```

06-la.cpp - 9077 bytes.

```
// compile: clang++ -std=c++20 -lpng 06-la.cpp -o 06-la
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image);
	void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
	void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
	void saveimagefile(std::vector<uint32_t>& v, std::string filename);
	void saveimagefile(std::vector<float>& v2, std::string filename);
	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
	void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
	void gain(std::vector<float>& v, double gain);
	void normalize(std::vector<float>& v);
	void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
	void generatesubtractive(std::vector<float>& v);
	void generatesample(std::vector<float>& v);
```

```
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("la");

        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> sample;
        std::vector<float> subtractive;
        std::vector<float> envelope1;
        std::vector<float> envelope2;
        std::vector<float> envelope;
        std::vector<float> lineararithmetic;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Setup Linear Arithmetic 1/2"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::generatesample(sample);
        SignalGenerators::normalize(sample);

        SignalGenerators::generatesubtractive(subtractive);
        SignalGenerators::normalize(subtractive);

```

```
        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        env.generateenvelope3(envelope1);
        env.generateenvelope2(envelope2);

        env.applyenvelope(sample,envelope1);
        SignalGenerators::normalize(sample);
        env.applyenvelope(subtractive,envelope2);
        SignalGenerators::normalize(sample);
        lineararithmetic.resize(sample.size());
        SignalGenerators::addwaves(sample,subtractive,lineararithmetic);
        SignalGenerators::normalize(lineararithmetic);
        SignalGenerators::saveaudiofile(lineararithmetic,"01-lineararithmetic.aiff");
        Render::saveimagefile(lineararithmetic,"02-lineararithmetic.png");

        // Update progress
        bar.set_progress(50);
        bar.set_option(option::PostfixText{"Generating: Linear Arithmetic 2/2"});

        env.applyenvelope(lineararithmetic,envelope);
        SignalGenerators::normalize(lineararithmetic);
        SignalGenerators::saveaudiofile(lineararithmetic,"03-final_lineararithmetic.aiff");
        Render::saveimagefile(lineararithmetic,"04-final_lineararithmetic.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 2/2"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
```

```
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        void generatesubtractive(std::vector<float>& v)
        {
                AudioFile<float> a;
                a.load("subtractive/08-postfilter.aiff");

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                v.push_back(a.samples[channel][i]);
                        }
                }
        }

        void generatesample(std::vector<float>& v)
        {
                AudioFile<float> a;
                a.load("fm/01-fm.aiff");

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
```

```
			{
				v.push_back(a.samples[channel][i]);
			}
		}
	}

	void saveaudiofile(std::vector<float>& v, std::string filename)
	{
		const std::string path="la/";
		// Setup the audio file
		AudioFile<float> a;
		a.setNumChannels(1);
		a.setBitDepth(24);
		a.setNumSamplesPerChannel(44100);

		for (int i=0;i<a.getNumSamplesPerChannel();++i)
		{
			for (int channel=0;channel<a.getNumChannels();++channel)
			{
				a.samples[channel][i]=v[i];
			}
		}
		a.save(path+filename,AudioFileFormat::Aiff);
	}
}

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image)
	{
		png::rgb_pixel px(0x04,0x13,0x31);
		for (uint32_t y=0;y<image.get_height();y++) {
			for (uint32_t x=0;x<image.get_width();++x) {
				image.set_pixel(x,y,px);
			}
		}
	}

	void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
	{
		if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
		{
```

```
                png::rgb_pixel px(0x7a,0xb1,0xe3);
                image.set_pixel(x,y,px);
            }
    }

    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
        int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
        dx = x2 - x1; dy = y2 - y1;
        if (dx == 0)
        {
            if (y2 < y1) std::swap(y1, y2);
            for (y = y1; y <= y2; y++)
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
        {
            if (x2 < x1) std::swap(x1, x2);
            for (x = x1; x <= x2; x++)
                drawpx(image, x, y1);
            return;
        }
        dx1 = abs(dx); dy1 = abs(dy);
        px = 2 * dy1 - dx1;     py = 2 * dx1 - dy1;
        if (dy1 <= dx1)
        {
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
```

```
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
            ox=x;oy=y;
        }
    }
```

```

        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
        {
                uint32_t halfHeight=image.get_height()/2;
                double value=0.0;
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }

                for (uint32_t i=0;i<v1.size();++i)
                {
                        value=v1[i];
                        if (value >= 0.0) {
                                v2[i]=halfHeight-(halfHeight*value);
                        } else if (value < 0.0)
                        {
                                v2[i]=halfHeight+(halfHeight*fabs(value));
                        }
                }
        }

        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
        {
                fillbackground(image);
                drawwave(image,v);
        }

        void saveimagefile(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="la/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveimagefile(std::vector<float>& v2, std::string filename)
        {
                const std::string path="la/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedtoimg(image,v2,v);
                renderimage(image,v);
```

```
                image.write(path+filename);
        }

        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
        {
                uint32_t height=image.get_height();
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v2[i]=height-(height*v1[i]);
                }
        }

        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="la/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="la/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

07-pd.cpp - 8581 bytes.

```
// compile: clang++ -std=c++20 -lpng 07-pd.cpp -o 07-pd

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void genereratephasedistortionwave(std::vector<float>& v, int duration, float frequ\
encyInHz, float x1, float y1, float x2, bool isCosine);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
```

```
        fs::create_directory("pd");

        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> pd;
        std::vector<float> envelope;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Generate Phase Distortion 1/2"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::genereratephasedistortionwave(pd,duration,440,0.2,0.5,0.7,true);
        SignalGenerators::saveaudiofile(pd,"01-pd.aiff");
        Render::saveimagefile(pd,"02-pd.png");

        // Update progress
        bar.set_progress(50);
        bar.set_option(option::PostfixText{"Final Phase Distortion 2/2"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        Render::saveenvelopeimage(envelope,"03-envelope.png");

        env.applyenvelope(pd,envelope);
        SignalGenerators::normalize(pd);
        SignalGenerators::saveaudiofile(pd,"04-final-pd.aiff");
        Render::saveimagefile(pd,"05-final-pd.png");
```

```

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 2/2"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        void genereratephasedistortionwave(std::vector<float>& v, int duration, float frequ\
encyInHz, float x1, float y1, float x2, bool isCosine)
```

```
        {
                const double sampleRate=44100.0;
                float delta=frequencyInHz/(sampleRate*duration);
                float pos=0;
                float xDelta=(x2-x1)/(sampleRate*duration);
                float warpedPos;
                float m;
                float b;
                float x=x1;

                for (uint32_t i=0;i<sampleRate*duration;++i)
                {
                        if (pos < x)
                        {
                                m=y1/x;
                                warpedPos=m*pos;
                        } else {
                                m=(1.0-y1)/(1.0-x);
                                b=1.0-m;
                                warpedPos=m*pos+b;
                        }
                        if (isCosine) {
                                v.push_back(cos(2.0*M_PI*warpedPos));
                        } else {
                                v.push_back(sin(2.0*M_PI*warpedPos));
                        }
                        pos += delta;
                        while (pos >= 1.0) {pos-=1.0;}
                        x+=xDelta;
                }
        }

        void saveaudiofile(std::vector<float>& v, std::string filename)
        {
                const std::string path="pd/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
```

```
				for (int channel=0;channel<a.getNumChannels();++channel)
				{
					a.samples[channel][i]=v[i];
				}
			}
			a.save(path+filename,AudioFileFormat::Aiff);
		}
}

namespace Render
{
	void fillbackground(png::image<png::rgb_pixel>& image)
	{
		png::rgb_pixel px(0x04,0x13,0x31);
		for (uint32_t y=0;y<image.get_height();y++) {
			for (uint32_t x=0;x<image.get_width();++x) {
				image.set_pixel(x,y,px);
			}
		}
	}

	void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
	{
		if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
		{
			png::rgb_pixel px(0x7a,0xb1,0xe3);
			image.set_pixel(x,y,px);
		}
	}

	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
	{
		int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
		dx = x2 - x1; dy = y2 - y1;
		if (dx == 0)
		{
			if (y2 < y1) std::swap(y1, y2);
			for (y = y1; y <= y2; y++)
				drawpx(image, x1, y);
			return;
		}
		if (dy == 0)
```

```
                {
                        if (x2 < x1) std::swap(x1, x2);
                        for (x = x1; x <= x2; x++)
                                drawpx(image, x, y1);
                        return;
                }
                dx1 = abs(dx); dy1 = abs(dy);
                px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
                if (dy1 <= dx1)
                {
                        if (dx >= 0)
                        {
                                x = x1; y = y1; xe = x2;
                        }
                        else
                        {
                                x = x2; y = y2; xe = x1;
                        }
                        drawpx(image, x, y);
                        for (i = 0; x<xe; i++)
                        {
                                x = x + 1;
                                if (px<0)
                                        px = px + 2 * dy1;
                                else
                                {
                                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                                        px = px + 2 * (dy1 - dx1);
                                }
                                drawpx(image, x, y);
                        }
                }
                else
                {
                        if (dy >= 0)
                        {
                                x = x1; y = y1; ye = y2;
                        }
                        else
                        {
                                x = x2; y = y2; ye = y1;
                        }
                        drawpx(image, x, y);
```

```
                for (i = 0; y<ye; i++)
                {
                    y = y + 1;
                    if (py <= 0)
                        py = py + 2 * dx1;
                    else
                    {
                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                        py = py + 2 * (dx1 - dy1);
                    }
                    drawpx(image, x, y);
                }
            }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
            ox=x;oy=y;
        }
    }

    void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
    {
        uint32_t halfHeight=image.get_height()/2;
        double value=0.0;
        if (v2.size() == 0 || v2.size() > v1.size()) {
            v2.resize(v1.size());
        }

        for (uint32_t i=0;i<v1.size();++i)
        {
            value=v1[i];
            if (value >= 0.0) {
                v2[i]=halfHeight-(halfHeight*value);
            } else if (value < 0.0)
            {
```

```
				v2[i]=halfHeight+(halfHeight*fabs(value));
			}
		}
	}

	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
	{
		fillbackground(image);
		drawwave(image,v);
	}

	void saveimagefile(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="pd/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="pd/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}

	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
	{
		uint32_t height=image.get_height();
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}
		for (uint32_t i=0;i<v1.size();++i)
		{
			v2[i]=height-(height*v1[i]);
		}
	}

	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
```

```
        {
                const std::string path="pd/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="pd/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

08-scanned.cpp - 9965 bytes.

```
// compile: clang++ -std=c++20 -lpng 08-scanned.cpp -o 08-scanned

#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

#include <iostream>

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
```

```
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void genereratescannedwave(std::vector<float>& v, int duration, float frequencyInHz\
);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("scanned");

        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> scanned;
        std::vector<float> envelope;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Generate Scanned 1/2"},
                option::ForegroundColor{Color::cyan},
```

```
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::genereratescannedwave(scanned,duration,440);
        SignalGenerators::saveaudiofile(scanned,"01-scanned.aiff");
        Render::saveimagefile(scanned,"02-scanned.png");

        // Update progress
        bar.set_progress(50);
        bar.set_option(option::PostfixText{"Final Scanned 2/2"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);

        env.applyenvelope(scanned,envelope);
        SignalGenerators::normalize(scanned);
        SignalGenerators::saveaudiofile(scanned,"03-final-scanned.aiff");
        Render::saveimagefile(scanned,"04-final-scanned.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 2/2"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
```

```
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        typedef struct {
                float k, b;
                float updateRate, updateSize;
        } SystemDesc;

        typedef struct {
                int size;
                float *position;
                float *velocity;
        } ScanState;

        void scan(std::vector<float>& v,ScanState *state, SystemDesc *system, float frequen\
cyInHz, float duration)
        {
                const double sampleRate=44100.0;
                float *previousPosition=new float[128];
                for (int i=0;i<sampleRate*duration;++i) {
                        auto scanPos = state->size * frequencyInHz * (float(i) / sampleRate);
                        int index = ((int) scanPos) % state->size;
                        v.push_back(state->position[index]);

                        float elapsed = 0;
                        while(elapsed < system->updateRate) {
                    previousPosition[state->size - 1] = state->position[state->size - 1];
```

```
                for (int j = 0; j < state->size; j++) {
                        auto prevIndex = j > 0 ? j - 1 : state->size - 1;
                        auto nextIndex = j < state->size - 1 ? j + 1 : 0;
                        previousPosition[j] = state->position[j];
                        previousPosition[nextIndex] = state->position[nextIndex];

                        float prev = previousPosition[prevIndex];
                        float next = previousPosition[nextIndex];
                        float deltaX = (prev + next + 0) / 3 - state->position[j];
                        float force = system->k * deltaX - system->b * state->veloc

                        state->velocity[j] += force * system->updateSize;
                        state->position[j] += state->velocity[j] * system->updateSiz
                    }
                    elapsed += system->updateSize;
                }
            }

            delete [] previousPosition;
        }

        void genereratescannedwave(std::vector<float>& v, int duration, float frequencyInHz)
        {
            const double sampleRate=44100.0;
            SystemDesc system = {.8, .01, .016, .01};
            ScanState state;
            state.size = 128;
            state.position = new float[state.size];
            state.velocity = new float[state.size];
            std::random_device rd;
            std::mt19937 gen(rd());
            std::uniform_real_distribution<> dis(0.0, 1.0);

            for (int i=0;i<state.size;++i) {
                float randOffset=dis(gen);
                float amplitude=0.1 + 0.5 * ((1+1) %2);
                state.position[i]=sin(2*M_PI*i/state.size/4) * amplitude + randOffset;
                state.velocity[i]=dis(gen);
            }

            scan(v,&state,&system,frequencyInHz,duration);

            delete [] state.position;
```

```
                delete [] state.velocity;
        }

        void saveaudiofile(std::vector<float>& v, std::string filename)
        {
                const std::string path="scanned/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                a.samples[channel][i]=v[i];
                        }
                }
                a.save(path+filename,AudioFileFormat::Aiff);
        }
}

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image)
        {
                png::rgb_pixel px(0x04,0x13,0x31);
                for (uint32_t y=0;y<image.get_height();y++) {
                        for (uint32_t x=0;x<image.get_width();++x) {
                                image.set_pixel(x,y,px);
                        }
                }
        }

        void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
        {
                if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
                {
                        png::rgb_pixel px(0x7a,0xb1,0xe3);
                        image.set_pixel(x,y,px);
                }
```

```
        }

        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
        {
                int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
                dx = x2 - x1; dy = y2 - y1;
                if (dx == 0)
                {
                        if (y2 < y1) std::swap(y1, y2);
                        for (y = y1; y <= y2; y++)
                                drawpx(image, x1, y);
                        return;
                }
                if (dy == 0)
                {
                        if (x2 < x1) std::swap(x1, x2);
                        for (x = x1; x <= x2; x++)
                                drawpx(image, x, y1);
                        return;
                }
                dx1 = abs(dx); dy1 = abs(dy);
                px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
                if (dy1 <= dx1)
                {
                        if (dx >= 0)
                        {
                                x = x1; y = y1; xe = x2;
                        }
                        else
                        {
                                x = x2; y = y2; xe = x1;
                        }
                        drawpx(image, x, y);
                        for (i = 0; x<xe; i++)
                        {
                                x = x + 1;
                                if (px<0)
                                        px = px + 2 * dy1;
                                else
                                {
                                        if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                                        px = px + 2 * (dy1 - dx1);
                                }
```

```
				drawpx(image, x, y);
			}
		}
		else
		{
			if (dy >= 0)
			{
				x = x1; y = y1; ye = y2;
			}
			else
			{
				x = x2; y = y2; ye = y1;
			}
			drawpx(image, x, y);
			for (i = 0; y<ye; i++)
			{
				y = y + 1;
				if (py <= 0)
					py = py + 2 * dx1;
				else
				{
					if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
					py = py + 2 * (dx1 - dy1);
				}
				drawpx(image, x, y);
			}
		}
	}

	void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
	{
		uint32_t y=0,ox=0,oy=0;
		for (uint32_t x=0;x<image.get_width();++x)
		{
			y=signalY[x];
			if (x == 0) {ox=x;oy=y;}
			drawline(image,x,y,ox,oy);
			ox=x;oy=y;
		}
	}

	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
```

```
	{
		uint32_t halfHeight=image.get_height()/2;
		double value=0.0;
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}

		for (uint32_t i=0;i<v1.size();++i)
		{
			value=v1[i];
			if (value >= 0.0) {
				v2[i]=halfHeight-(halfHeight*value);
			} else if (value < 0.0)
			{
				v2[i]=halfHeight+(halfHeight*fabs(value));
			}
		}
	}

	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
	{
		fillbackground(image);
		drawwave(image,v);
	}

	void saveimagefile(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="scanned/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="scanned/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}
```

```
	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
	{
		uint32_t height=image.get_height();
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}
		for (uint32_t i=0;i<v1.size();++i)
		{
			v2[i]=height-(height*v1[i]);
		}
	}

	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="scanned/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveenvelopeimage(std::vector<float>& v2, std::string filename)
	{
		const std::string path="scanned/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedenvelopetoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}

}
```

09-vectorsynth.cpp - 10237 bytes.

```cpp
// compile: clang++ -std=c++20 -lpng 09-vectorsynth.cpp -o 09-vectorsynth
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatevectorsynth(std::vector<float>& v, int duration, double frequencyInHz);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("vectorsynth");

```

```
        const int duration=1;
        const double sampleRate=44100.0;
        std::vector<float> vectorsynth;
        std::vector<float> envelope;

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Setup Vector Synth 1/2"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::generatevectorsynth(vectorsynth,duration,440);
        SignalGenerators::saveaudiofile(vectorsynth,"01-vectorsynth.aiff");
        Render::saveimagefile(vectorsynth,"02-vectorsynth.png");

        // Update progress
        bar.set_progress(50);
        bar.set_option(option::PostfixText{"Generating: Vector Synth 2/2"});

        ADSR::Envelope env(sampleRate,duration);
        env.generateenvelope(envelope);
        env.applyenvelope(vectorsynth,envelope);
        SignalGenerators::saveaudiofile(vectorsynth,"03-final_vectorsynth.aiff");
        Render::saveimagefile(vectorsynth,"04-final_vectorsynth.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 2/2"});

```

```
        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }

        void generatetrianglewave(std::vector<float>& v, int duration, double frequencyInHz)
        {
                const double sampleRate=44100.0;

                for (uint32_t i=0;i<sampleRate*duration;++i) {
                                v.push_back(M_2_PI*asin(sin(frequencyInHz*2*M_PI*i/sampleRate)));
                }
```

```
    }

    void generateinversesawtoothwave(std::vector<float>& v, int duration, double freque\
ncyInHz)
    {
        const double sampleRate=44100.0;
        double period=sampleRate/frequencyInHz;
        std::vector<float> temp;

        for (uint32_t i=0;i<period;++i) {
            temp.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
        }
        for (uint32_t i=0;i<sampleRate*duration;++i) {
            v.push_back(temp[temp.size()-(i%temp.size())]);
        }
    }

    void generatesawtoothwave(std::vector<float>& v, int duration, double frequencyInHz)
    {
        const double sampleRate=44100.0;

        for (uint32_t i=0;i<sampleRate*duration;++i) {
            v.push_back(-2/M_PI*atan(1/tan(frequencyInHz*M_PI*i/sampleRate)));
        }
    }

    void generatesquarewave(std::vector<float>& v, int duration, double frequencyInHz)
    {
        const double sampleRate=44100.0;
        double period=sampleRate/frequencyInHz;
        double dutyCycle=period*0.5;
        double ss=0.0;

        for (uint32_t i=0;i<sampleRate*duration;++i) {
            if ((i%int(period)) >= 0 && (i%int(period)) < int(dutyCycle))
            {
                ss=0.7;
            } else {
                ss=-0.7;
            }
            v.push_back(ss);
        }
        v[0]=0.0;
```

```
                v[v.size()-1]=0.0;
        }

        void generatevectorsynth(std::vector<float>& v, int duration, double frequencyInHz)
        {
                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,\
0.5},{0.0,0.25,0.5,1.0}};
                const double sampleRate=44100.0;
                std::vector<float> sample1;
                std::vector<float> sample2;
                std::vector<float> sample3;
                std::vector<float> sample4;

                generatetrianglewave(sample1,duration,frequencyInHz);
                generateinversesawtoothwave(sample2,duration,frequencyInHz);
                generatesawtoothwave(sample3,duration,frequencyInHz);
                generatesquarewave(sample4,duration,frequencyInHz);

                int ptr=0;
                for (int i=0;i<4;++i) {
                        for (int j=0;j<11025;++j) {
                                sample1[ptr]=sample1[ptr]*vectortransition[i][0];
                                sample2[ptr]=sample2[ptr]*vectortransition[i][1];
                                sample3[ptr]=sample3[ptr]*vectortransition[i][2];
                                sample4[ptr]=sample4[ptr]*vectortransition[i][3];
                                ptr++;
                        }
                }

                v.resize(sample1.size());
                addwaves(sample1,sample2,v);
                addwaves(v,sample3,v);
                addwaves(v,sample4,v);
                normalize(v);
        }

        void saveaudiofile(std::vector<float>& v, std::string filename)
        {
                const std::string path="vectorsynth/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
```

```
			a.setNumSamplesPerChannel(44100);

			for (int i=0;i<a.getNumSamplesPerChannel();++i)
			{
				for (int channel=0;channel<a.getNumChannels();++channel)
				{
					a.samples[channel][i]=v[i];
				}
			}
			a.save(path+filename,AudioFileFormat::Aiff);
		}
}

namespace Render
{
		void fillbackground(png::image<png::rgb_pixel>& image)
		{
			png::rgb_pixel px(0x04,0x13,0x31);
			for (uint32_t y=0;y<image.get_height();y++) {
				for (uint32_t x=0;x<image.get_width();++x) {
					image.set_pixel(x,y,px);
				}
			}
		}

		void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
		{
			if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
			{
				png::rgb_pixel px(0x7a,0xb1,0xe3);
				image.set_pixel(x,y,px);
			}
		}

		void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
		{
			int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
			dx = x2 - x1; dy = y2 - y1;
			if (dx == 0)
			{
				if (y2 < y1) std::swap(y1, y2);
				for (y = y1; y <= y2; y++)
```

```
                drawpx(image, x1, y);
            return;
        }
        if (dy == 0)
        {
            if (x2 < x1) std::swap(x1, x2);
            for (x = x1; x <= x2; x++)
                drawpx(image, x, y1);
            return;
        }
        dx1 = abs(dx); dy1 = abs(dy);
        px = 2 * dy1 - dx1;        py = 2 * dx1 - dy1;
        if (dy1 <= dx1)
        {
            if (dx >= 0)
            {
                x = x1; y = y1; xe = x2;
            }
            else
            {
                x = x2; y = y2; xe = x1;
            }
            drawpx(image, x, y);
            for (i = 0; x<xe; i++)
            {
                x = x + 1;
                if (px<0)
                    px = px + 2 * dy1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
```

```
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
            ox=x;oy=y;
        }
    }

    void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
    {
        uint32_t halfHeight=image.get_height()/2;
        double value=0.0;
        if (v2.size() == 0 || v2.size() > v1.size()) {
            v2.resize(v1.size());
        }

        for (uint32_t i=0;i<v1.size();++i)
        {
            value=v1[i];
```

```
				if (value >= 0.0) {
					v2[i]=halfHeight-(halfHeight*value);
				} else if (value < 0.0)
				{
					v2[i]=halfHeight+(halfHeight*fabs(value));
				}
			}
		}

		void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
		{
			fillbackground(image);
			drawwave(image,v);
		}

		void saveimagefile(std::vector<uint32_t>& v, std::string filename)
		{
			const std::string path="vectorsynth/";
			png::image<png::rgb_pixel> image(44100,600);
			renderimage(image,v);
			image.write(path+filename);
		}

		void saveimagefile(std::vector<float>& v2, std::string filename)
		{
			const std::string path="vectorsynth/";
			png::image<png::rgb_pixel> image(44100,600);
			std::vector<uint32_t> v;
			normalizedtoimg(image,v2,v);
			renderimage(image,v);
			image.write(path+filename);
		}

		void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
		{
			uint32_t height=image.get_height();
			if (v2.size() == 0 || v2.size() > v1.size()) {
				v2.resize(v1.size());
			}
			for (uint32_t i=0;i<v1.size();++i)
			{
				v2[i]=height-(height*v1[i]);
```

```
                }
        }

        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="vectorsynth/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="vectorsynth/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

10-virtualanalog.cpp - 1351 bytes.

```
// compile: clang++ -std=c++20 10-virtualanalog.cpp -o 10-virtualanalog
#include <iostream>

int main()
{
        std::cout << "An analog modeling synthesizer is a synthesizer that generates the so\
unds of traditional analog synthesizers using DSP components and software algorithms\
. Analog modeling synthesizers simulate the behavior of the original electric and el\
ectronic circuitry in order to digitally replicate their tone.\n\n";
        std::cout << "This method of synthesis is also referred to as Virtual Analog or VA.\
 Analog modeling synthesizers can be more reliable than their true analog counterpar\
ts since the oscillator pitch is ultimately maintained by a digital clock, and the d\
igital hardware is typically less susceptible to temperature changes.\n\n";
        std::cout << "While analog synthesizers need an oscillator circuit for each voice o\
f polyphony, analog modeling synthesizers don't face this problem. This means that m\
any of them, especially the more modern models, can produce as many polyphonic voice\
s as the CPU on which they run can handle.\n\n";
        std::cout << "Modeling synths also provide patch storage capabilities and MIDI supp\
```

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

11-wavetable.cpp - 10677 bytes.

```
// compile: clang++ -std=c++20 -lpng 11-wavetable.cpp -o 11-wavetable
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatewavetable(std::vector<float>& v, int duration);
        void saveaudiofile(std::vector<float>& v, std::string filename);
```

```
}

int main()
{
    namespace fs = std::filesystem;
    fs::create_directory("wavetable");

    const double sampleRate=44100.0;
    const int duration=1;
    std::vector<float> wavetablewave;
    std::vector<float> envelope;

    using namespace indicators;
    // Hide cursor
    show_console_cursor(false);

    // Setup ProgressBar
    ProgressBar bar{
        option::BarWidth{50},
        option::Start{"["},
        option::Fill{"□"},
        option::Lead{"□"},
        option::Remainder{"-"},
        option::End{" ]"},
        option::PostfixText{"Setting up: wavetable 1/2"},
        option::ForegroundColor{Color::cyan},
        option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
    };

    // Update progress
    bar.set_progress(0);

    SignalGenerators::generatewavetable(wavetablewave,duration);
    SignalGenerators::normalize(wavetablewave);
    SignalGenerators::saveaudiofile(wavetablewave,"01-wavetablewave.aiff");
    Render::saveimagefile(wavetablewave,"02-wavetablewave.png");

    // Update progress
    bar.set_progress(50);
    bar.set_option(option::PostfixText{"Generating wavetable 2/2"});

    ADSR::Envelope env(sampleRate,duration);
    env.generateenvelope(envelope);
```

```
        env.applyenvelope(wavetablewave,envelope);
        SignalGenerators::saveaudiofile(wavetablewave,"03-final_wavetablewave.aiff");
        Render::saveimagefile(wavetablewave,"04-final_wavetablewave.png");

        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 2/2"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain)
        {
                for (uint32_t i=0;i<v.size();++i) {
                        v[i]=v[i]*gain;
                }
        }

        void normalize(std::vector<float>& v)
        {
                float max=0.0,value=0.0;
                for (uint32_t i=0;i<v.size();++i) {
                        value=v[i];
                        if (value > max) {max=value;}
                }
                // max=std::ceil(max);
                for (uint32_t i=0;i<v.size();++i) {
                        // v[i]=v[i]/max;
                        v[i]=(v[i]/max)*0.707;
                }
        }

        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
        {
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v3[i]=v1[i]+v2[i];
                }
        }
```

```

        int lfo[2048];
        float wavetable[2048][2048];

        void setuplfo()
        {
                for (int i=0;i<2048;++i)
                {
                        lfo[i]=int(2047-(1023-1023*sin((static_cast<double> (i) / 2048) * 21.5 * 2.0
PI)));
                }
        }

        void setupwavetable()
        {
                for (int i=0;i<2048;++i) {
                        wavetable[0][i]=M_2_PI*asin(sin(440*2*M_PI*i/2048));
                }

                for (int i=0;i<2048;++i) {
                        wavetable[511][i]=-2/M_PI*atan(1/tan(440*M_PI*i/2048));
                }

                for (int i=0;i<2048;++i) {
                        wavetable[1023][i]=sin((static_cast<double> (i) / 2048) * 440 * 2.0 * M_PI)
                }

                for (int i=0;i<2048;++i) {
                        wavetable[1535][i]=M_2_PI*asin(sin(440*2*M_PI*i/2048));
                }

                for (int i=0;i<2048;++i) {
                        wavetable[2047][i]=-2/M_PI*atan(1/tan(440*M_PI*i/2048));
                }

                float diff=0.0;
                float step=0.0;
                for (int i=0;i<2048;++i) {
                        diff=std::fabs(wavetable[0][i] - wavetable[511][i]);
                        step=diff/511;
                        for (int j=0;j<512;++j) {
                                if (wavetable[0][i] < wavetable[511][i]) {
                                        wavetable[j][i]=wavetable[0][i]+(step*j);
```

```
                        } else if (wavetable[0][i] > wavetable[511][i]) {
                                wavetable[j][i]=wavetable[0][i]-(step*j);
                        } else if (wavetable[0][i] == wavetable[511][i]) {
                                wavetable[j][i]=wavetable[0][i];
                        }
                }
        }

        for (int i=0;i<2048;++i) {
                diff=std::fabs(wavetable[511][i] - wavetable[1023][i]);
                step=diff/511;
                for (int j=511;j<1024;++j) {
                        if (wavetable[511][i] < wavetable[1023][i]) {
                                wavetable[j][i]=wavetable[511][i]+(step*j);
                        } else if (wavetable[511][i] > wavetable[1023][i]) {
                                wavetable[j][i]=wavetable[511][i]-(step*j);
                        } else if (wavetable[511][i] == wavetable[1023][i]) {
                                wavetable[j][i]=wavetable[511][i];
                        }
                }
        }

        for (int i=0;i<2048;++i) {
                diff=std::fabs(wavetable[1023][i] - wavetable[1535][i]);
                step=diff/511;
                for (int j=1023;j<1536;++j) {
                        if (wavetable[1023][i] < wavetable[1535][i]) {
                                wavetable[j][i]=wavetable[1023][i]+(step*j);
                        } else if (wavetable[1023][i] > wavetable[1535][i]) {
                                wavetable[j][i]=wavetable[1023][i]-(step*j);
                        } else if (wavetable[1023][i] == wavetable[1535][i]) {
                                wavetable[j][i]=wavetable[1023][i];
                        }
                }
        }

        for (int i=0;i<2048;++i) {
                diff=std::fabs(wavetable[1535][i] - wavetable[2047][i]);
                step=diff/511;
                for (int j=1535;j<2048;++j) {
                        if (wavetable[1535][i] < wavetable[2047][i]) {
                                wavetable[j][i]=wavetable[1535][i]+(step*j);
                        } else if (wavetable[1535][i] > wavetable[2047][i]) {
```

```
                                        wavetable[j][i]=wavetable[1535][i]-(step*j);
                                } else if (wavetable[1535][i] == wavetable[2047][i]) {
                                        wavetable[j][i]=wavetable[1535][i];
                                }
                        }
                }
        }

        void generatewavetable(std::vector<float>& v, int duration)
        {
                const double sampleRate=44100.0;
                setuplfo();
                setupwavetable();

                for (int i=0;i<sampleRate*duration;++i) {
                        v.push_back(wavetable[lfo[i%2048]][i%2048]);
                }
        }

        void saveaudiofile(std::vector<float>& v, std::string filename)
        {
                const std::string path="wavetable/";
                // Setup the audio file
                AudioFile<float> a;
                a.setNumChannels(1);
                a.setBitDepth(24);
                a.setNumSamplesPerChannel(44100);

                for (int i=0;i<a.getNumSamplesPerChannel();++i)
                {
                        for (int channel=0;channel<a.getNumChannels();++channel)
                        {
                                a.samples[channel][i]=v[i];
                        }
                }
                a.save(path+filename,AudioFileFormat::Aiff);
        }
}

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image)
        {
```

```
		png::rgb_pixel px(0x04,0x13,0x31);
		for (uint32_t y=0;y<image.get_height();y++) {
			for (uint32_t x=0;x<image.get_width();++x) {
				image.set_pixel(x,y,px);
			}
		}
	}

	void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
	{
		if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
		{
			png::rgb_pixel px(0x7a,0xb1,0xe3);
			image.set_pixel(x,y,px);
		}
	}

	void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
	{
		int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
		dx = x2 - x1; dy = y2 - y1;
		if (dx == 0)
		{
			if (y2 < y1) std::swap(y1, y2);
			for (y = y1; y <= y2; y++)
				drawpx(image, x1, y);
			return;
		}
		if (dy == 0)
		{
			if (x2 < x1) std::swap(x1, x2);
			for (x = x1; x <= x2; x++)
				drawpx(image, x, y1);
			return;
		}
		dx1 = abs(dx); dy1 = abs(dy);
		px = 2 * dy1 - dx1;		py = 2 * dx1 - dy1;
		if (dy1 <= dx1)
		{
			if (dx >= 0)
			{
				x = x1; y = y1; xe = x2;
```

```
			}
			else
			{
				x = x2; y = y2; xe = x1;
			}
			drawpx(image, x, y);
			for (i = 0; x<xe; i++)
			{
				x = x + 1;
				if (px<0)
					px = px + 2 * dy1;
				else
				{
					if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
					px = px + 2 * (dy1 - dx1);
				}
				drawpx(image, x, y);
			}
		}
		else
		{
			if (dy >= 0)
			{
				x = x1; y = y1; ye = y2;
			}
			else
			{
				x = x2; y = y2; ye = y1;
			}
			drawpx(image, x, y);
			for (i = 0; y<ye; i++)
			{
				y = y + 1;
				if (py <= 0)
					py = py + 2 * dx1;
				else
				{
					if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
					py = py + 2 * (dx1 - dy1);
				}
				drawpx(image, x, y);
			}
		}
```

```
        }

        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
        {
                uint32_t y=0,ox=0,oy=0;
                for (uint32_t x=0;x<image.get_width();++x)
                {
                        y=signalY[x];
                        if (x == 0) {ox=x;oy=y;}
                        drawline(image,x,y,ox,oy);
                        ox=x;oy=y;
                }
        }

        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
        {
                uint32_t halfHeight=image.get_height()/2;
                double value=0.0;
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }

                for (uint32_t i=0;i<v1.size();++i)
                {
                        value=v1[i];
                        if (value >= 0.0) {
                                v2[i]=halfHeight-(halfHeight*value);
                        } else if (value < 0.0)
                        {
                                v2[i]=halfHeight+(halfHeight*fabs(value));
                        }
                }
        }

        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
        {
                fillbackground(image);
                drawwave(image,v);
        }

        void saveimagefile(std::vector<uint32_t>& v, std::string filename)
        {
```

```
		const std::string path="wavetable/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="wavetable/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedtoimg(image,v2,v);
		renderimage(image,v);
		image.write(path+filename);
	}

	void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
v1,std::vector<uint32_t>& v2)
	{
		uint32_t height=image.get_height();
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}
		for (uint32_t i=0;i<v1.size();++i)
		{
			v2[i]=height-(height*v1[i]);
		}
	}

	void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="wavetable/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveenvelopeimage(std::vector<float>& v2, std::string filename)
	{
		const std::string path="wavetable/";
		png::image<png::rgb_pixel> image(44100,600);
		std::vector<uint32_t> v;
		normalizedenvelopetoimg(image,v2,v);
```

```
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

12-physicalmodelling_karplus_strong.cpp - 9153 bytes.

```
// compile: clang++ -std=c++20 -lpng 12-physicalmodelling_karplus_strong.cpp -o 12-p\
hysicalmodelling_karplus_strong
#define _USE_MATH_DEFINES
#include <cmath>
#include <vector>
#include <random>
#include <filesystem>
#include "indicators.hpp"
#include <png++/png.hpp>
#include "AudioFile/AudioFile.h"
#include "Envelope.hpp"

namespace Render
{
        void fillbackground(png::image<png::rgb_pixel>& image);
        void drawpx(png::image<png::rgb_pixel>& image, int x, int y);
        void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2);
        void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY);
        void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2);
        void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v);
        void saveimagefile(std::vector<uint32_t>& v, std::string filename);
        void saveimagefile(std::vector<float>& v2, std::string filename);
        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2);
        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename);
        void saveenvelopeimage(std::vector<float>& v2, std::string filename);
}

namespace SignalGenerators
{
        void gain(std::vector<float>& v, double gain);
        void normalize(std::vector<float>& v);
        void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3);
        void generatekarplusstrong(std::vector<float>& v, int duration, double frequencyInH\
```

```cpp
z, double feedback);
        void saveaudiofile(std::vector<float>& v, std::string filename);
}

int main()
{
        namespace fs = std::filesystem;
        fs::create_directory("physicalmodelling");

        const int duration=1;
        const double sampleRate=44100.0;
        const double feedback=0.2;
        std::vector<float> karplusstrong;
        int octave=3;
        int note=0;
        double frequencyInHz=float(440 * pow(2.0, ((double)((octave - 4) * 12 + note)) / 12\
.0));

        using namespace indicators;
        // Hide cursor
        show_console_cursor(false);

        // Setup ProgressBar
        ProgressBar bar{
                option::BarWidth{50},
                option::Start{"["},
                option::Fill{"□"},
                option::Lead{"□"},
                option::Remainder{"-"},
                option::End{" ]"},
                option::PostfixText{"Generate Physical Modelling 1/1"},
                option::ForegroundColor{Color::cyan},
                option::FontStyles{std::vector<FontStyle>{FontStyle::bold}}
        };

        // Update progress
        bar.set_progress(0);

        SignalGenerators::generatekarplusstrong(karplusstrong,duration,frequencyInHz,0.996);
        SignalGenerators::normalize(karplusstrong);
        SignalGenerators::saveaudiofile(karplusstrong,"01-karplusstrong.aiff");
        Render::saveimagefile(karplusstrong,"02-karplusstrong.png");

```

```
        // Update progress
        bar.set_progress(100);
        bar.set_option(option::PostfixText{"Done 1/1"});

        // Show cursor
        show_console_cursor(true);
        return 0;
}

namespace SignalGenerators
{

        class KSString
        {
        public:
            KSString(float frequencyInHz, float sampleRate, float feedback)
            {
                std::random_device rd;
                        std::mt19937 gen(rd());
                        std::uniform_real_distribution<> dis(-1.0, 1.0);

                m_buffer.resize(uint32_t(float(sampleRate) / frequencyInHz));
                for (size_t i = 0, c = m_buffer.size(); i < c; ++i) {
                    m_buffer[i] = dis(gen);
                }
                m_index = 0;
                m_feedback = feedback;
            }

            float GenerateSample()
            {
                // get our sample to return
                float ret = m_buffer[m_index];

                // low pass filter (average) some samples
                float value = (m_buffer[m_index] + m_buffer[(m_index + 1) % m_buffer.size()\
]) * 0.5f * m_feedback;
                m_buffer[m_index] = value;

                // move to the next sample
                m_index = (m_index + 1) % m_buffer.size();

                // return the sample from the buffer
```

```
            return ret;
        }

    private:
        std::vector<float>  m_buffer;
        size_t              m_index;
        float               m_feedback;
    };

    void gain(std::vector<float>& v, double gain)
    {
            for (uint32_t i=0;i<v.size();++i) {
                    v[i]=v[i]*gain;
            }
    }

    void normalize(std::vector<float>& v)
    {
            float max=0.0,value=0.0;
            for (uint32_t i=0;i<v.size();++i) {
                    value=v[i];
                    if (value > max) {max=value;}
            }
            // max=std::ceil(max);
            for (uint32_t i=0;i<v.size();++i) {
                    // v[i]=v[i]/max;
                    v[i]=(v[i]/max)*0.707;
            }
    }

    void addwaves(std::vector<float>& v1,std::vector<float>& v2,std::vector<float>& v3)
    {
            for (uint32_t i=0;i<v1.size();++i)
            {
                    v3[i]=v1[i]+v2[i];
            }
    }

    void generatekarplusstrong(std::vector<float>& v, int duration, double frequencyInH\
z, double feedback)
    {
            const double sampleRate=44100.0;
            KSString pluck(frequencyInHz,sampleRate,0.996f);
```

```
            v.resize(sampleRate);
            for (uint32_t i=0;i<v.size();++i) {
                    v[i]=0;
                    v[i]+=pluck.GenerateSample();
            }

    }

    void saveaudiofile(std::vector<float>& v, std::string filename)
    {
            const std::string path="physicalmodelling/";
            // Setup the audio file
            AudioFile<float> a;
            a.setNumChannels(1);
            a.setBitDepth(24);
            a.setNumSamplesPerChannel(44100);

            for (int i=0;i<a.getNumSamplesPerChannel();++i)
            {
                    for (int channel=0;channel<a.getNumChannels();++channel)
                    {
                            a.samples[channel][i]=v[i];
                    }
            }
            a.save(path+filename,AudioFileFormat::Aiff);
    }
}


namespace Render
{
    void fillbackground(png::image<png::rgb_pixel>& image)
    {
            png::rgb_pixel px(0x04,0x13,0x31);
            for (uint32_t y=0;y<image.get_height();y++) {
                    for (uint32_t x=0;x<image.get_width();++x) {
                            image.set_pixel(x,y,px);
                    }
            }
    }

    void drawpx(png::image<png::rgb_pixel>& image, int x, int y)
    {
```

```
            if (((x >= 0) && (x < image.get_width())) && ((y >= 0) && (y < image.get_height())\
))
            {
                    png::rgb_pixel px(0x7a,0xb1,0xe3);
                    image.set_pixel(x,y,px);
            }
    }

    void drawline(png::image<png::rgb_pixel>& image, int x1, int y1, int x2, int y2)
    {
            int x, y, dx, dy, dx1, dy1, px, py, xe, ye, i;
            dx = x2 - x1; dy = y2 - y1;
            if (dx == 0)
            {
                    if (y2 < y1) std::swap(y1, y2);
                    for (y = y1; y <= y2; y++)
                            drawpx(image, x1, y);
                    return;
            }
            if (dy == 0)
            {
                    if (x2 < x1) std::swap(x1, x2);
                    for (x = x1; x <= x2; x++)
                            drawpx(image, x, y1);
                    return;
            }
            dx1 = abs(dx); dy1 = abs(dy);
            px = 2 * dy1 - dx1;         py = 2 * dx1 - dy1;
            if (dy1 <= dx1)
            {
                    if (dx >= 0)
                    {
                            x = x1; y = y1; xe = x2;
                    }
                    else
                    {
                            x = x2; y = y2; xe = x1;
                    }
                    drawpx(image, x, y);
                    for (i = 0; x<xe; i++)
                    {
                            x = x + 1;
                            if (px<0)
```

```
                    px = px + 2 * dy1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) y = y + 1; else y = y
                    px = px + 2 * (dy1 - dx1);
                }
                drawpx(image, x, y);
            }
        }
        else
        {
            if (dy >= 0)
            {
                x = x1; y = y1; ye = y2;
            }
            else
            {
                x = x2; y = y2; ye = y1;
            }
            drawpx(image, x, y);
            for (i = 0; y<ye; i++)
            {
                y = y + 1;
                if (py <= 0)
                    py = py + 2 * dx1;
                else
                {
                    if ((dx<0 && dy<0) || (dx>0 && dy>0)) x = x + 1; else x = x
                    py = py + 2 * (dx1 - dy1);
                }
                drawpx(image, x, y);
            }
        }
    }

    void drawwave(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& signalY)
    {
        uint32_t y=0,ox=0,oy=0;
        for (uint32_t x=0;x<image.get_width();++x)
        {
            y=signalY[x];
            if (x == 0) {ox=x;oy=y;}
            drawline(image,x,y,ox,oy);
```

```
				ox=x;oy=y;
			}
	}

	void normalizedtoimg(png::image<png::rgb_pixel>& image, std::vector<float>& v1,std:\
:vector<uint32_t>& v2)
	{
		uint32_t halfHeight=image.get_height()/2;
		double value=0.0;
		if (v2.size() == 0 || v2.size() > v1.size()) {
			v2.resize(v1.size());
		}

		for (uint32_t i=0;i<v1.size();++i)
		{
			value=v1[i];
			if (value >= 0.0) {
				v2[i]=halfHeight-(halfHeight*value);
			} else if (value < 0.0)
			{
				v2[i]=halfHeight+(halfHeight*fabs(value));
			}
		}
	}

	void renderimage(png::image<png::rgb_pixel>& image,std::vector<uint32_t>& v)
	{
		fillbackground(image);
		drawwave(image,v);
	}

	void saveimagefile(std::vector<uint32_t>& v, std::string filename)
	{
		const std::string path="physicalmodelling/";
		png::image<png::rgb_pixel> image(44100,600);
		renderimage(image,v);
		image.write(path+filename);
	}

	void saveimagefile(std::vector<float>& v2, std::string filename)
	{
		const std::string path="physicalmodelling/";
		png::image<png::rgb_pixel> image(44100,600);
```

```
                std::vector<uint32_t> v;
                normalizedtoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

        void normalizedenvelopetoimg(png::image<png::rgb_pixel>& image, std::vector<float>&\
 v1,std::vector<uint32_t>& v2)
        {
                uint32_t height=image.get_height();
                if (v2.size() == 0 || v2.size() > v1.size()) {
                        v2.resize(v1.size());
                }
                for (uint32_t i=0;i<v1.size();++i)
                {
                        v2[i]=height-(height*v1[i]);
                }
        }

        void saveenvelopeimage(std::vector<uint32_t>& v, std::string filename)
        {
                const std::string path="physicalmodelling/";
                png::image<png::rgb_pixel> image(44100,600);
                renderimage(image,v);
                image.write(path+filename);
        }

        void saveenvelopeimage(std::vector<float>& v2, std::string filename)
        {
                const std::string path="physicalmodelling/";
                png::image<png::rgb_pixel> image(44100,600);
                std::vector<uint32_t> v;
                normalizedenvelopetoimg(image,v2,v);
                renderimage(image,v);
                image.write(path+filename);
        }

}
```

quiz.cpp - 338679 bytes.

```
// compile: clang++ -std=c++20 quiz.cpp -o quiz
#include <iostream>
#include <fstream>
#include <vector>
#include <algorithm>
#include <random>

namespace quiz
{
        class Quiz
        {
        public:
                Quiz(const std::string &q, const std::string &a) {_a=a;_q=q;}
                virtual ~Quiz() {}
                std::string getQ() {return _q;}
                std::string getA() {return _a;}
        private:
                std::string _q;
                std::string _a;
        };
}

std::vector<quiz::Quiz> game{
quiz::Quiz("0-5v","Denotes a range of 0 to 5 volts, which is common for gates, trigg\
ers, and modulation control voltages in modular synthesizers. Gates and triggers - w\
hich initiate events such as new notes - typically rise from 0v to 5v (0 to 10v is a\
lso common), with roughly the middle of that onset starting the event. Gates are con\
sidered high when held at 5v (or 10v), and then low when they return to 0v."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 6 decibels weaker for each octave further away you get from the cutoff fr\
equency. A 6dB/octave filter is often referred to as a “one pole” filter (as each po\
le of a filter’s design results in 6dB of attenuation), and has a relatively weak ef\
fect on the signal going through it. Low Pass Gates (LPGs) typically - but not alway\
s - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 deci\
bels for every octave above its cutoff frequency."),
quiz::Quiz("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 division\
s."),
```

```
quiz::Quiz("1 v/oct","The most common standard for controlling pitch in a modular sy\
nthesizer. Under the system, increasing the voltage going into a VCO (Voltage Contro\
lled Oscillator) 1 volt - say, from 0.5v to 1.5v - would raise its pitch by one octa\
ve."),
quiz::Quiz("1.2 v/oct","Buchla compatible synths have standardized on the 1.2 volt p\
er 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 f\
or a change in pitch of 1 semitone."),
quiz::Quiz("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."),
quiz::Quiz("1/8”","Often used to incorrectly describe the connector size commonly us\
ed in Eurorack format modules, as well as Buchla audio signals. In fact, Eurorack mo\
dules 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 phy\
sically 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.")\
,
quiz::Quiz("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 rang\
e may be varied to change the depth of their effect, so don’t get too hung up on spe\
cific voltage ranges. Pay more attention to whether they vary between 0v and some va\
lue, or swing in roughly equal amounts both above and below 0v (as 10vpp does)."),
quiz::Quiz("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 cutof\
f frequency. A 12dB/octave filter is often referred to as a “two pole” filter (as ea\
ch 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 soun\
ds when compared to those with 4-pole instruments."),
quiz::Quiz("16’","Sometimes seen on octave selector switches on oscillators. It refe\
rs to the length of an organ pipe. Longer pipes = lower pitches; 16’ is in the mid-b\
ass 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."),
quiz::Quiz("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 cutof\
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f frequency. It is often used a coded shorthand for when someone wants to refer to a\
cid-type bass lines from a TB-303 without mentioning the instrument by name."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 12 decibels weaker for each octave further away you get from the cutoff f\
requency. 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 Obe\
rheim instruments often featured 2-pole filters, often resulting in brighter sounds \
when compared to those with 4-pole instruments."),
quiz::Quiz("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 th\
at the modules are attached to."),
quiz::Quiz("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 cutof\
f frequency. This design is often used in vintage Moog and Roland synths. 4-pole fil\
ters are often associated with subjectively fatter, more “round” sounds than 2-pole \
filters – but generalizations are always dangerous."),
quiz::Quiz("24 ppqn","A common master clock division used in MIDI, DIN sync, and oth\
er systems common to electronic music and synthesizers. It means internally, 24 subd\
ivisions 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 (÷6), eighth notes\
 (÷12), eight note triplets (÷8), etc."),
quiz::Quiz("2’","Sometimes seen on octave selector switches for oscillators. It refe\
rs 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."),
quiz::Quiz("3 mm","A common screw thread size used to mount Eurorack modules. This s\
ize is most common when using module mounting rails that have been pre-drilled."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 18 decibels weaker for each octave further away you get from the cutoff f\
requency. 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."),
quiz::Quiz("3.5 mm","The standard connector size used for jacks and cables in Eurora\
ck format modular synthesizers. Note that this is slightly larger that 1/8”."),
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quiz::Quiz("303","The TB-303 Bass Line by Roland became a cult favorite in Acid Hous\
e 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;"),
quiz::Quiz("32’","Sometimes seen on octave selector switches on oscillators. It refe\
rs to the length of an organ pipe. Longer pipes = lower pitches; 32’ is the lowest s\
etting 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 h\
igher; etc."),
quiz::Quiz("3U","Refers to modules that are 3 rack units (U) high – the Eurorack sta\
ndard, which is by far the most common modular format today, even though it’s one of\
 the youngest formats."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 24 decibels weaker for each octave further away you get from the cutoff f\
requency. This design is often used in vintage Moog and Roland synths. 4-pole filter\
s are often associated with subjectively fatter, more “round” sounds than 2-pole fil\
ters – but generalizations are always dangerous."),
quiz::Quiz("4-40","A screw thread size occasionally used to mount Eurorack modules. \
This size is used by Pittsburgh Modular for their cases, for example."),
quiz::Quiz("4U","Refers to modules that are 4U (rack units) high – namely, Buchla an\
d 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.)\
"),
quiz::Quiz("4’","Sometimes seen on octave selector switches on oscillators. It refer\
s 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.")\
,
quiz::Quiz("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 fo\
llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
d is both historical and physically large, some users “5U” as a badge of honor that \
they’re traditional and cool. (And the are.) There was also a briefly popular 5U for\
mat from MOTM that used a different width and power connection. It has since been di\
scontinued, but there are still diehard MOTM format users today."),
quiz::Quiz("6 dB/oct","This format of numbers and abbreviations (dB/oct = decibels p\
er 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 (f\
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ilters) 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 alw\
ays – use 1 pole low pass filters, reducing the strength of higher harmonics by 6 de\
cibels for every octave above its cutoff frequency."),
quiz::Quiz("808","The TR-808 Rhythm Composer by Roland created all of its sounds usi\
ng analog circuitry. When it first came out, it was not well loved, as the analog so\
unds weren’t realistic enough. But later, music styles such as House and Hip-Hop ado\
pted 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 s\
nappy attack."),
quiz::Quiz("8’","Sometimes seen on octave selector switches on oscillators. It refer\
s to the length of an organ pipe. Shorter pipes = higher pitches; 8’ is typically as\
sociated 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."),
quiz::Quiz("909","The TR-909 Rhythm Composer was the follow-up to Roland’s now-rever\
ed 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-lik\
e sounds, this is the instrument it is referencing."),
quiz::Quiz("A-440","This is the frequency in hertz (cycles per second) of the A abov\
e Middle C. It is often used as a tuning reference."),
quiz::Quiz("A/B Technique","A stereo microphone placement technique in which two car\
dioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart fr\
om each other (depending on the size of the sound source) to create a left/right ste\
reo image. Also known as Spaced Pair."),
quiz::Quiz("A/D","Abbreviation of Analog-to-Digital Conversion, the conversion of a \
quantity that has continuous changes (like electrical signals) into numbers that app\
roximate those changes (i.e., computer data)."),
quiz::Quiz("Absolute Phase","This term describes a perfect polarity between an origi\
nal signal (into the microphone) and the reproduced signal (through the speaker). Wh\
en positive pressure exerted upon the microphone is translated as positive pressure \
to the loudspeaker, the two are in “absolute phase.”."),
quiz::Quiz("Absorption","In acoustics, absorption is what happens when sound waves a\
re absorbed by a surface, as opposed to bouncing off the surface (reflection). Absor\
ptive materials in a control room, for example, tend to “deaden” the sound of the ro\
om because the sound energy is absorbed rather than reflected. (See also “Reflection\
.”)"),
quiz::Quiz("AC Coupled","An AC coupled input attempts to remove any constant DC volt\
age going through it. This is useful if have an audio signal (such as the output of \
an oscillator) which is AC in nature, and you want to remove any accidental DC offse\
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t 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 A\
C coupling is a high pass filter that is attempting to remove very low frequency com\
ponents."),
quiz::Quiz("AC","Alternating Current - The type of electrical current found in stand\
ard electrical outlets and studio signals running through audio lines. In AC, the cu\
rrent “alternates” directions, flowing back and forth through the circuit. In modula\
r terms, AC refers to a voltage that alternates between positive and negative values\
 - such as the output of an oscillator."),
quiz::Quiz("Accelerometer","A device that measures the acceleration to which it is s\
ubjected and creates an electric signal to match it. In music and audio, acceleromet\
ers are found in such things as microphones and guitar pickups."),
quiz::Quiz("Acorn Tube","Named for its acorn-like shape, an acorn tube is a small va\
cuum tube used in ultra high frequency (UHF) electronics such as tube amplifiers."),
quiz::Quiz("Acoustic Amplifier","The part of a musical instrument that vibrates in r\
esponse to the initial vibration of the instrument, causing the surrounding air to m\
ove more efficiently and making the sound louder. For example: the body of an acoust\
ic guitar, the bell of a horn, a drum’s shell, and the wooden soundboard of a piano.\
"),
quiz::Quiz("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."),
quiz::Quiz("Acoustics","The science of the sound—more specifically, the science of t\
he properties and behavior of sound waves. A good understanding of acoustics is esse\
ntial to audio engineering and studio design."),
quiz::Quiz("Active Device","A component that is designed with the ability to control\
 electrical current (as opposed to a “Passive Device”). In the recording studio, act\
ive devices are generally components that include an amplifier. (See also “Passive D\
evice.”)"),
quiz::Quiz("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 desti\
nations. An active or buffered multiple is one that includes a buffer circuit betwee\
n the input and output, making sure the signal does not lose its strength or integri\
ty 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 buil\
t into their outputs, and can drive multiple modules without issue. But if you try t\
o use a passive mult to connect to, say, three oscillators, and you realize the trac\
king 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."),
quiz::Quiz("Actuator","The part of a switch that causes change of the contact connec\
tions (e.g., toggle, pushbutton, or rocker)."),
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quiz::Quiz("AD","Shorthand for a two-stage Attack/Decay envelope. This simple envelo\
pe shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 vol\
ts) 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 enve\
lope: After finishing the Attack stage, it holds at the maximum level for a specifie\
d amount of time (in contrast to an AR envelope, which holds at the maximum level fo\
r as long as the note on gate is high), and then decays back to zero. I have heard t\
here are some envelopes that a hybrid of AHD and AR in that they hold the maximum le\
vel for either the defined Hold time or the as long as the incoming gate is high;"),
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. A\
dditive synthesis is a technique that gives you direct control over each of those co\
mponent harmonics, allowing you to directly dial in the mix you want. As immediate a\
nd intuitive as that sounds on paper (or on screen), in reality it takes a lot of wo\
rk to craft the correct mixture to recreate another sound, especially since the stre\
ngth of each harmonic usually varies over time. Additive synthesis oscillators are r\
elatively rare in modular synths; two examples are the Verbos Harmonic Oscillator an\
d the Make Noise tELHARMONIC."),
quiz::Quiz("ADSR","An envelope generator with four stages: Attack, Decay, Sustain, a\
nd 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 Controlle\
d Amplifier, or the lowest frequency when connected to a voltage controlled filter o\
r oscillator) and raises to the maximum voltage it can output (typically 5 to 10 vol\
ts depending on system; it can often be set with an output level control) over a tim\
e set by the Attack control. Once it reaches that level, the output voltage immediat\
ely starts dropping to speed set by the Decay control it until it reaches the voltag\
e set by the Sustain control. If the input gate is still active, this level is maint\
ained 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."),
quiz::Quiz("AES","Audio Engineering Society."),
quiz::Quiz("AES3","(sometimes called AES/EBU) A digital audio transfer standard deve\
loped by the Audio Engineering Society and the European Broadcasting Union for carry\
ing dual-channel digital audio data between devices. AES3 is the protocol behind XLR\
 cables, as well as RCA and S/PDIF cables."),
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 re\
furbished B-stock units come up for sale every now and then. The expansion modules w\
ere, to the best of my knowledge, never released (at least not widely)."),
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 contro\
l 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 wi\
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der. Monophonic aftertouch measures one pressure value for the entire keyboard, rega\
rdless of which key(s) you are pressing; polyphonic aftertouch produces a signal for\
 each individual key. Important trivia: Touch plate keyboards actually measure the s\
urface area of the skin touching them rather than pressure or force — so you can inc\
rease or decrease the aftertouch amount by rolling between the tip and length of you\
r finger."),
quiz::Quiz("AHDSR","Attack, Hold, Decay, Sustain, and Release. This is a slightly fa\
ncier ADSR envelope that holds the voltage typically at its maximum value for a spec\
ified time after the attack is done rising and before the decay starts falling."),
quiz::Quiz("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 s\
ample rate is an audible pitch, you will also hear a mirror image of the sound’s har\
monic content reproduced started at that half-sample-rate pivot (unless some excelle\
nt filtering has taken place). (See also “Nyquist Frequency.”)."),
quiz::Quiz("Alternating Current (or AC)","The type of electrical current found in st\
andard electrical outlets and studio signals running through audio lines. In AC, the\
 current “alternates” directions, flowing back and forth through the circuit."),
quiz::Quiz("AM","Amplitude Modulation (AM) is the name given the 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 oscil\
lator with a positive voltage (say, between 0v and 5v, or maybe something smaller su\
ch as between 1v and 2v) is fed into the control input of a voltage controlled ampli\
fier to add vibrato to an audio signal passing through it. Technically, this is know\
n as a two-quadrant multiplier or modulator, as any negative swings in the modulatio\
n signal are ignored; when patching tremolo, you may need to make sure an offset vol\
tage is being added to your LFO to make sure the sound doesn’t cut out on the lower \
excursions of the LFO’s waveform."),
quiz::Quiz("Ambience","In most cases, this refers to the “atmosphere” of a certain p\
lace, like a restaurant. But in recording, it refers to the part of the sound that c\
omes from the surrounding environment rather than directly from the sound source. Fo\
r example, the sound waves coming into your ears from a cello being played are comin\
g directly from the source, but the sound of the same cello coming to you after boun\
cing off the back wall is ambient sound."),
quiz::Quiz("Ambient Field","The area away from the sound source where the reverberat\
ion is louder than the direct sound."),
quiz::Quiz("Ambient Miking","This refers to placing a microphone in the ambient fiel\
d of a room to record the ambient reverberations of the sound. The recording enginee\
r often does this in addition to direct micing of the instrument(s) to create a blen\
d or mix of direct and reverberant sound in the recording."),
quiz::Quiz("Amp","An abbreviation for “Amplifier,” “Amplitude” or “Ampere,” dependin\
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g on context."),
quiz::Quiz("Ampere","The unit of measure for electrical current, abbreviated Amp."),
quiz::Quiz("Amplifier","A device that increases the level or amplitude of an electri\
cal signal, making the resulting sound louder."),
quiz::Quiz("Amplitude Modulation","Amplitude Modulation (AM) is the name given the t\
o the technique of varying the amplitude or loudness of one signal known as the carr\
ier (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 l\
ow frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe so\
mething smaller such as between 1v and 2v) is fed into the control input of a voltag\
e controlled amplifier to add vibrato to an audio signal passing through it. Technic\
ally, this is known as a two-quadrant multiplier or modulator, as any negative swing\
s 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."),
quiz::Quiz("Amplitude","The height of a waveform above or below the zero line. In au\
dio, this usually translates to the signal strength or the volume of the sound."),
quiz::Quiz("Analog Recording","A recording of the continuous changes of an audio wav\
eform. The most common example of analog recording in a recording studio is recordin\
g on reel-to-reel magnetic tape."),
quiz::Quiz("Analog Shift Register","An Analog Shift Register (ASR) is a cross betwee\
n a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how t\
hose 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 aga\
in with this new voltage presented at the first output, while the original voltage i\
s now moved to a second output. This game of \"telephone\" is passed along for as ma\
ny stages as the ASR has – traditionally three or four."),
quiz::Quiz("Analog To Digital Converter (A/D; or ADC)","A device that translates a c\
ontinuously changing signal (analog) into numeric values that approximate those chan\
ges (digital). In audio recording, this refers to converting recorded sound from ele\
ctrical voltages to computerized data."),
quiz::Quiz("Analog","The term analog implies a signal is continuously variable, comp\
ared 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."),
quiz::Quiz("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 o\
utput: The output would be low if both inputs were high, but otherwise would be high\
."),
quiz::Quiz("AR","The two-stage Attack/Release envelope raises from 0 volts to its ma\
ximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack param\
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eter, and then stays at that value for as long as the gate signal fed into the envel\
ope 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 g\
ate signal.)"),
quiz::Quiz("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 betwee\
n your keyboard and sound module - makes it easier for you to play arpeggios: You ju\
st hold down the notes of the chord, and it automatically plays the notes one at a t\
ime, over and over again, like a step sequencer you can program on the fly just by h\
olding 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 t\
his even after you’ve released the keys."),
quiz::Quiz("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 i\
nitially 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 v\
oltage presented at the first output, while the original voltage is now moved to a s\
econd output. This game of \"telephone\" is passed along for as many stages as the A\
SR has - traditionally three or four."),
quiz::Quiz("Attack/Decay/Sustain/Release","An envelope generator with four stages: A\
ttack, Decay, Sustain, and Release. When this envelope generator receives a gate inp\
ut, it typically starts at 0 volts (which is the equivalent of silence when connecte\
d to a Voltage Controlled Amplifier, or the lowest frequency when connected to a vol\
tage controlled filter or oscillator) and raises to the maximum voltage it can outpu\
t (typically 5 to 10 volts depending on system; it can often be set with an output l\
evel 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 it unt\
il it reaches the voltage set by the Sustain control. If the input gate is still act\
ive, this level is maintained until the gate goes back to 0 volts (usually because y\
ou released the key on a controlling keyboard, etc.). At that time, the output volta\
ge then starts dropping back to 0 volts at the rate set by the Release control."),
quiz::Quiz("Attack/Decay","Shorthand for a two-stage Attack/Decay envelope. This sim\
ple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perh\
aps 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 th\
e 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 maximu\
m level for as long as the note on gate is high), and then decays back to zero. I ha\
ve 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 i\
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s high;"),
quiz::Quiz("Attack/Hold/Decay/Sustain/Release","This is a slightly fancier ADSR enve\
lope that holds the voltage typically at its maximum value for a specified time afte\
r the attack is done rising and before the decay starts falling."),
quiz::Quiz("Attack/Release","The two-stage Attack/Release envelope raises from 0 vol\
ts 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 in\
to the envelope generator stays high. Then when the gate signal goes back to zero, t\
he 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 y\
ou specify a fixed time for the level to stay at its maximum, rather than pay attent\
ion to the gate signal.)"),
quiz::Quiz("Attack","This usually refers to the first stage of an envelope that occu\
rs 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."),
quiz::Quiz("Attenuation","The reduction of electrical or acoustic signal strength. I\
n audio, attenuation is measured in decibels (dB) and is typically heard as a reduct\
ion in volume. Sound waves traveling through the air naturally attenuate as they tra\
vel away from the source of the sound. Engineers also purposefully attenuate signals\
 in the studio through gain controls or pads to prevent overload."),
quiz::Quiz("Attenuator","A control that can reduce the strength of a signal or volta\
ge going through it."),
quiz::Quiz("Attenuverter","A special version of an attenuator that can also invert t\
he polarity of the signal or voltage going through it. Most attenuverters use pass t\
hrough no signal at their center position; as you turn them clockwise, you turn up t\
he 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 po\
larity switch added on."),
quiz::Quiz("Audio Frequency Generator","The AFG (Audio Frequency Generator) is a ver\
y full-featured analog oscillator released by Livewire Electronics. It has since bee\
n discontinued, but refurbished B-stock units come up for sale every now and then. T\
he expansion modules were, to the best of my knowledge, never released (at least not\
 widely)."),
quiz::Quiz("Audio","In its broadest sense, audio is the range of frequencies we huma\
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\
."),
quiz::Quiz("Automatic Dialogue Replacement (ADR)","The process of re-recording dialo\
gue for film in a controlled environment after the film is shot, for the purpose of \
replacing poorly recorded dialogue."),
quiz::Quiz("Automatic Gain Control","A compressor with a long release time, which is\
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 used to keep the volume of the audio at a consistent level."),
quiz::Quiz("Automation","Programming certain changes to occur automatically during r\
ecording and/or playback. In the studio, engineers use automation on their consoles \
or computers so various parameters will change automatically at different times duri\
ng multitrack recording and playback. This pre-programming feature makes it easier t\
o create those changes than attempting to perform them all manually in real time."),
quiz::Quiz("Auxiliary Equipment","External signal processing devices that work along\
side the mixing console to modify the signal."),
quiz::Quiz("Auxiliary Return","(Abbreviated Aux Return or Return) The input on a con\
sole or DAW that returns the effected signal sent through the auxiliary send back in\
to the channel mix."),
quiz::Quiz("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 equi\
pment 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 th\
rough the auxiliary return."),
quiz::Quiz("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 micr\
ophone 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."),
quiz::Quiz("Background Noise","Refers to either 1) The ambient noise in a room unrel\
ated to the instrument(s) or vocal(s) being recorded; or 2) The system noise unrelat\
ed to the recorded signal. (All electronics emit a level of noise.)"),
quiz::Quiz("Baffles","Sound absorbing panels that are used to prevent sound waves fr\
om entering or leaving a space."),
quiz::Quiz("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 eve\
n out the relative levels of audio signals in the channels of stereo recording."),
quiz::Quiz("Balanced Audio","This refers to a system where three wires are used to c\
arry an audio signal: one is the ground (the 0 volt reference), the second carries t\
he audio signal as it varies above and below 0v, and the third carries an inverted c\
opy of the audio signal that goes negative while the original is going positive. Bal\
anced audio usually implies a reference signal level of +4dB (higher than line level\
; still lower than most modular synths), although microphone signals - much weaker b\
y comparison, and therefore more susceptible to outside noise - are almost always ba\
lanced as well. Modular synths tend to use unbalanced audio for their internal signa\
ls. If you require a balanced output (or input), you need a special module that conv\
erts between balanced and unbalanced audio, plus does any necessary level matching."\
),
quiz::Quiz("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 op\
posite polarities, providing protection against interference and noise in a balanced\
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 system. Examples of balanced cables include tip-ring-sleeve (TRS) stereo cables and\
 XLR cables."),
quiz::Quiz("Balanced Mixer","A circuit or device that generates the sum and differen\
ce frequencies of two input signals."),
quiz::Quiz("Balanced Modulator","Balanced or ring modulation is a special type of am\
plitude 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 a\
s the carrier. The modulator’s frequency is both added to and subtracted from the ca\
rrier’s frequency; the resulting harmonics replace the original carrier and modulato\
r."),
quiz::Quiz("Banana","An alternate type of connector (https://en.wikipedia.org/wiki/B\
anana_connector) used by 4U systems such as Buchla (control voltages) and Serge (bot\
h 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 refer\
ence. Banana connectors have an advantage in that they are usually “stackable” meani\
ng you can plug a one jack into the back of another, providing a passive multiple.")\
,
quiz::Quiz("Band Pass Filter","A device, circuit or plug-in that allows a narrow ban\
d of frequencies to pass through the circuit, rejecting or attenuating frequencies t\
hat are either higher or lower than the specified range."),
quiz::Quiz("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. T\
he exact opposite of a band pass filter."),
quiz::Quiz("Band Track","(Sometimes abbreviated “Track“) A mixdown of a song minus t\
he lead vocal and/or background vocals. In other words, a mixed track containing onl\
y the instrumental parts of the song."),
quiz::Quiz("Band","1) A range of frequencies, often identified by the center frequen\
cy of the range. 2) A group of musicians playing together."),
quiz::Quiz("Bandpass Filter","A bandpass filter (BPF) leaves the harmonics around th\
e 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 a\
ttenuated, based on the number of poles in the filter, with each pole equalling 6 de\
cibels of attenuation for each octave you get away from that center."),
quiz::Quiz("Bandwidth","In signal processing, bandwidth refers to the usable frequen\
cy range of a communication channel, measured by the difference between the device’s\
 highest and lowest usable frequencies."),
quiz::Quiz("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 sou\
nd modules as a unit."),
quiz::Quiz("Bar","In music notation, bar is another term for measure a specified per\
iod of time containing a certain number of beats, and marked by bar lines on each si\
de of the written measure."),
quiz::Quiz("Bark Scale","The human auditory (hearing) system can be thought of as co\
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nsisting of a series of bandpass filters. Interestingly, the spacing of these filter\
s do not strictly follow either a linear frequency scale or a logarithmic musical sc\
ale. The Bark Scale is an attempt to determine what the center frequency and bandwid\
th of those \"hearing filters\" are (known as critical bands)."),
quiz::Quiz("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, prev\
enting phase cancellation between the two."),
quiz::Quiz("Basic Session","The first audio recording session for recording the basi\
c tracks that serve as the song’s foundation (for example, the drums and bass)."),
quiz::Quiz("Bass Reflex","A type of loudspeaker cabinet design in which a port (open\
ing) in the speaker cabinet enhances bass frequencies. The principle is that the sou\
nd pressure generated by the back of the speaker cone inside the cabinet is routed o\
ut 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."),
quiz::Quiz("Bass","The lower range of audio frequencies up to approximately 250 Hz. \
A reference value."),
quiz::Quiz("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 samp\
le and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade d\
elays (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 multi\
plied by the time interval between samples."),
quiz::Quiz("Beaming","A phenomenon found in loudspeakers in which higher frequencies\
 are projected straight out of the loudspeaker, rather than dispersing along with th\
e lower frequencies. When you stand on-axis in front of the speaker, it sounds as th\
ough it is only reproducing the high frequencies, rather than the mids or lows. This\
 phenomenon is alleviated by routing the high frequncies through horns in the loudsp\
eaker."),
quiz::Quiz("Beat Mapping","The process of adjusting the tempo variations in a record\
ed 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 temp\
o markers. This technique is often used, for example, to reconcile a drum or bass pe\
rformance that was recorded without a click track."),
quiz::Quiz("Beat","1) The steady, even pulse in music. 2) The action of two sounds o\
r audio signals of slightly different frequency interfering with one another and cau\
sing periodic increases and decreases in volume, heard to the ear as “beats.”"),
quiz::Quiz("Beating","When two oscillators are tuned to very nearly – but not quite \
– the same frequency, the difference between them causes an interference pattern kno\
wn as beating. When the difference in frequency is below the audio rate, this can so\
und like a tremolo applied to the loudness of the combined sound."),
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quiz::Quiz("Beatmatching","A technique predominantly used by DJs to synchronize the \
tempos of two recorded tracks, generally through the use of time stretching and pitc\
h shifting tools, to create a seamless transition from one song into another."),
quiz::Quiz("Beats Per Minute (B.P.M.)","BPM (beats per minute) is the most common wa\
y of stating tempo: How many beats (typically, quarter notes) should be counted ever\
y minute. A tempo of 120 beats per minute means there would be two beats every secon\
d (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."),
quiz::Quiz("Berlin School","A particular style of electronic music popularized by th\
e 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 re\
cent versions of Berlin School music can be heard from Node and Red Shift."),
quiz::Quiz("Bi-amplification","A technique in which high and low frequencies in a sp\
eaker or speaker system are driven by two separate amplifiers."),
quiz::Quiz("Bi-Directional Pattern","A microphone pickup pattern which is most sensi\
tive to picking up sounds directly in front and back of the mic, effectively rejecti\
ng sounds coming from the sides. Also called a “figure-8 pattern.”"),
quiz::Quiz("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 trigge\
r signal in a modular synth – even if generated by analog circuitry – could be refer\
red to as a binary type signal. See the entry for Boolean for things you can do with\
 binary signals like gates and divided clocks."),
quiz::Quiz("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 (va\
rying the pitch of an oscillator both above and below the note it is supposed to be \
playing) – are bipolar in nature."),
quiz::Quiz("Bit","The smallest unit of digital information representing a single “0”\
 or “1.”"),
quiz::Quiz("Bitrate (or Bit Depth)","In digital recording, the number of computer bi\
ts used to describe each sample. The greater the bitrate, the greater the dynamic ra\
nge of the sampled sound. The quality and resolution of an audio sample are describe\
d as a combination of sample rate and bitrate. (See also “Sample Rate.”)"),
quiz::Quiz("Blending","The mixing of multiple sounds or channels together to form on\
e sound, or mixing the left and right signals together."),
quiz::Quiz("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."),
quiz::Quiz("Boolean","Boolean logic only can have two states: high or low; 1 or 0; o\
n or off."),
quiz::Quiz("Boom Stand","A microphone stand equipped with a telescoping support arm \
to hold the microphone."),
quiz::Quiz("Boom","A telescoping support arm attached to a microphone stand holding \
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the microphone."),
quiz::Quiz("Boost","To increase gain at specific frequencies with an equalizer."),
quiz::Quiz("Bouncing","(also called “Ping-Ponging” or “Ponging“) The technique of co\
mbining 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 record\
ing them onto separate tracks, or digitally through a digital audio workstation. Bou\
ncing was once used frequently by engineers to free up additional tracks for recordi\
ng, but in digital workstations where tracks are virtually unlimited, this practice \
is basically obsolete. Today, engineers typically bounce tracks for the purpose of c\
reating a preliminary or final mix of a song."),
quiz::Quiz("Boundary Microphone","An omnidirectional microphone designed to be place\
d flush against a flat surface (or boundary), effectively creating a “half-Omni” pic\
kup pattern while eliminating the danger of phase issues from reflected sounds. A po\
pular type of boundary microphone is Crown Audio’s trademark Pressure Zone Microphon\
e (PZM)."),
quiz::Quiz("BPF","A bandpass filter (BPF) leaves the harmonics around the center, co\
rner 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, b\
ased on the number of poles in the filter, with each pole equalling 6 decibels of at\
tenuation for each octave you get away from that center."),
quiz::Quiz("BPM","BPM (beats per minute) is the most common way of stating tempo: Ho\
w many beats (typically, quarter notes) should be counted every minute. A tempo of 1\
20 beats per minute means there would be two beats every second (120 beats/minute x \
1 minute/60 seconds = 2)."),
quiz::Quiz("Breathing","Pumping and Breathing – In studio jargon, an effect created \
when a compressor is rapidly compressing and releasing the sound, creating audible c\
hanges in the signal level. “Pumping” generally refers to the audible increase of so\
und 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 an\
d breathing is a sign of cheap compression or over-compression, and is usually undes\
irable, although some engineers and musicians use it on purpose occasionally to crea\
te a particular effect."),
quiz::Quiz("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-e\
ffects to the audio signal, such as phase shift."),
quiz::Quiz("Bridging","A technique of feeding a single input to both channels of an \
amplifier, then summing them into one, thereby effectively doubling the amplifier po\
wer supplied to the signal."),
quiz::Quiz("Brownian Noise","Also referred to as brown noise, technically it’s a typ\
e of noise whose power density (spectral loudness) decreases 6 dB per octave with in\
creasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a di\
stance. It can also be used a slowly changing random control voltage or modulation s\
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ignal, instead of as an audio source."),
quiz::Quiz("Buchla Bongos","This is a classic patch where a complex sound source - s\
uch 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 a\
ttack/fast decay envelope to create a nice percussive sound. The fact that the low p\
ass gate reduces the higher harmonics as its volume dies away helps tame the harmoni\
cs coming from the complex source, and give it a decay similar to a struck percussiv\
e instrument."),
quiz::Quiz("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 pow\
ers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the numbe\
r of stages multiplied by the time interval between samples."),
quiz::Quiz("Bucking","A type of phase cancellation in which two identical signals or\
 frequencies, having the same amplitude but opposite polarity, cancel one another ou\
t. 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 si\
gnal using this principle."),
quiz::Quiz("Buffered Multiple","Quite often you need to split or copy a signal to se\
nd to more than one destination. This is commonly done with a multiple, where you pl\
ug one source in, and then plug in additional patch cables to go off to multiple des\
tinations. An active or buffered multiple is one that includes a buffer circuit betw\
een the input and output, making sure the signal does not lose its strength or integ\
rity by being split too many times, and that no funny business happening on one of t\
he outputs affects any of the other connections. Some modules have good buffering bu\
ilt 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 tr\
acking 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."),
quiz::Quiz("Bulk Dump","Short for System Exclusive Bulk Dump, a method of transmitti\
ng data such as the internal parameters between MIDI devices."),
quiz::Quiz("Burst Generator","When you send this module a trigger, it outputs a stre\
am or “burst” of triggers in response. You usually have control over the number of t\
riggers, 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 u\
se to create “double pluck” triggers in response to a normal note on; and its most e\
xtreme, it is used to trigger a high-energy, chaotic stream of drum hits that may or\
 may not be in time with the music."),
quiz::Quiz("Bus Board","This simple circuit board takes the output of your modular s\
ystem's power supply and creates multiple copies of it, routed to connectors that go\
 to your individual modules."),
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quiz::Quiz("Bus","An audio pathway by which one or more signals, usually from differ\
ent sources, are routed to a designated place. Because busses are highly connected t\
o signal flow, they serve a broad range of purposes in audio applications. 2) A shor\
thand term for the signals themselves that are routed through the bus (see also “Sub\
group”)."),
quiz::Quiz("Byte","Information (data) bits in a grouping of eight. One byte = eight \
bits."),
quiz::Quiz("Cable Assembly","Cable that is ready for installation in specific applic\
ations and usually terminated with connectors."),
quiz::Quiz("Cable Harness","A grouping of cables or wires used to interconnect elect\
ronic systems."),
quiz::Quiz("Cable Sheath","Conductive protective cover that is applied to cables."),
quiz::Quiz("Cable","A group of one or more insulated conductors, optical fibers, or \
a combination of both within an enveloping jacket, typically for transmitting electr\
ical signals of different types."),
quiz::Quiz("Capacitor","An electronic device made of two plates separated by an insu\
lator, designed to store electrostatic energy. The capacitor is a key component in c\
ondenser microphones, for example."),
quiz::Quiz("Capstan","A mechanical part of a magnetic tape recorder that controls th\
e speed of the tape as it passes across the tape heads."),
quiz::Quiz("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."),
quiz::Quiz("Carbon Microphone","A microphone that uses carbon granules to convert so\
und waves to electrical impulses. The carbon element sits between two plates; as sou\
nd waves hit the carbon granules, it generates changes in resistance between the pla\
tes, affecting the electrical signal."),
quiz::Quiz("Cardioid Pattern","A microphone pickup pattern which is most sensitive t\
o sound coming from the front, less from the sides, and least from the back of the d\
iaphragm. So named because the pickup pattern is in the shape of a heart (cardio).")\
,
quiz::Quiz("Carrier","There are a few different synthesis techniques where one usual\
ly audio-rate signal varies another audio signal. For example, in frequency modulati\
on, 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."),
quiz::Quiz("Cascade","To connect or “daisy chain” two mixers so that the stereo mixi\
ng busses of the first mixer feed into the stereo busses of the second."),
quiz::Quiz("CCW","Counter-clockwise, usually in the context of rotating a control th\
e left (in the opposite direction of how a clock's hands move)."),
quiz::Quiz("CD","An abbreviation for Compact Disc, or a small optical disk with digi\
tal audio recorded on it."),
quiz::Quiz("Cent","When tuning instruments, a semitone is divided into 100 units cal\
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led 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 me\
asured in cents."),
quiz::Quiz("Center Frequency","The frequency of an audio signal that is most affecte\
d 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."),
quiz::Quiz("Channel Path","The complete signal path from the sound source to the mul\
titrack recorder (or DAW). For example, an audio signal that travels from the microp\
hone to the preamplifier, then into a channel strip on the mixing console, then is s\
ent 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 t\
he recorded signals. (See also “Monitor Path.”)"),
quiz::Quiz("Channel","1) An audio recording made on a portion of the width of a mult\
itrack 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."),
quiz::Quiz("Chaotic","Believe it or not, chaotic does not mean completely random to \
mathematicians. Chaos theory deals with systems that are random within certain bound\
aries – 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 sy\
nthesis, a chaotic system usually refers to a modulation generator that is similar t\
o 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."),
quiz::Quiz("Chase","The automatic adjusting of the speed of a recorder (or sequencer\
) to keep time with another recorder."),
quiz::Quiz("Chord Chart","A shorthand form of musical notation that provides the bas\
ic 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 struc\
ture 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.”)"),
quiz::Quiz("Chord","Three or more musical pitches sung or played together."),
quiz::Quiz("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.")\
,
quiz::Quiz("Circuit","1) One complete path of electric current. 2) Similar to defini\
tion 1, but including all audio signal paths and components to accomplish a particul\
ar audio function."),
quiz::Quiz("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\
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 without additional drivers needing to be installed. This comes up in the modular wo\
rld with MIDI to CV/Gate interfaces that use USB: If your converter is a USB Host, a\
nd you plug a class compliant USB Device such as a controller keyboard or fader pane\
l into it, the converter will recognize it."),
quiz::Quiz("Click Track","A metronome “click” fed into headphone monitors for the pu\
rpose of helping the musicians play in time with the song."),
quiz::Quiz("Clip","All active electronic circuits have a limit on how strong of a si\
gnal 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 hig\
her 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; t\
his 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.”."),
quiz::Quiz("Clock Signal","A signal sent by a device within the circuit that generat\
es 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 con\
nected devices will play in time."),
quiz::Quiz("Clock","Usually refers to the main rhythmic pulse in a system. Often, th\
e clock pulse is much faster than anything it might drive, such as a sequencer or LF\
O. The most common clock rate is 24 ppqn (pulses per quarter note), as is the case w\
ith 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 module\
s that create divisions and multiplications of the main clock to generate new clock \
signals with a relationship to the main clock."),
quiz::Quiz("Clockwise","Clockwise, as in rotating a control the the right - in the s\
ame direction as a clock's hands move."),
quiz::Quiz("Close Miking","A microphone placement technique that places the mic clos\
e to the sound source to pick up the direct sound and reject ambient sound."),
quiz::Quiz("Coaxial Cable","(abbreviated “Coax”) A two-conductor cable that consists\
 of one conductor surrounded by a shield."),
quiz::Quiz("Coincident Miking","A stereo miking technique in which two microphones a\
re placed with their heads as close to each other as possible. This prevents phase c\
ancellation problems in the mix because the distance from the sound to either microp\
hone is the same."),
quiz::Quiz("Compander","A signal processor serving as a combination compressor and e\
xpander, primarily used for noise reduction purposes in analog systems.  The audio s\
ignal is compressed prior to recording, then expanded at the reproduction stage. Com\
panding is the principle behind Dolby noise reduction systems."),
quiz::Quiz("Comparator","An electrical device that compares the level of one voltage\
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 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."),
quiz::Quiz("Comping","1) In digital audio workstations (DAWs), the process of blendi\
ng portions of multiple recorded takes to create a “compliation” track. (See also “T\
ake,” “Playlist.)  2) In jazz music performance, an abbreviation for “accompanying.”\
"),
quiz::Quiz("Complex Oscillator","This module typically has a pair of oscillators beh\
ind 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 swit\
ch them so that the first modulates the amplitude of the second, or some other varia\
tion. They may also have waveshapers built in. They are based on a popular module cr\
eated by Buchla, which is a standard of the “West Coast” approach to synthesis."),
quiz::Quiz("Compression Driver","A diaphragm that feeds a sound pressure wave into a\
 horn loudspeaker."),
quiz::Quiz("Compression Ratio","The rate by which a compressor attenuates an incomin\
g signal, measured in decibels. For example, a compression ratio of 4:1 means the co\
mpressor will only allow a 1 dB increase in the signal for every 4 dB increase in th\
e signal above the threshold."),
quiz::Quiz("Compression","1) In signal processing, the action performed by a compres\
sor (see also “Compressor”). 2) In acoustics, the increased air pressure caused by t\
he peak of a sound pressure wave, used in the context of “compression and rarefactio\
n” (see also “Rarefaction”)."),
quiz::Quiz("Compressor","A signal processor or plug-in that reduces the dynamic rang\
e of an audio signal by amplifying its quieter sections and attenuating its louder o\
nes."),
quiz::Quiz("Condenser Microphone","A microphone in which sound is converted into ele\
ctrical current through changes in a capacitor. The sound pressure waves move the di\
aphragm, producing changes in capacitance which are then changed into electrical vol\
tage."),
quiz::Quiz("Contact Microphone","A microphone designed to pick up vibrations from so\
lid objects (as opposed to vibrations in the air). Also known as a “pickup” or “piez\
o,” this microphone is often used as an acoustic guitar pickup to pick up the vibrat\
ions from the soundboard, or by experimental musicians creating “noise music” from a\
 variety of objects."),
quiz::Quiz("Control Voltage Processor","CVP is the abbreviation for a module that al\
lows 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 fr\
om an incoming voltage by running it through a comparator. Make Noise’s Maths is per\
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haps the most well known control voltage processor out there; you will also find som\
e 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 d\
o what you want (or to teach them new tricks)."),
quiz::Quiz("Control Voltage","The concept of control voltage (CV) is at the very roo\
t of modular synthesizer. The general idea is that analog voltage levels are used co\
ntrol functions and parameters of a module. For example, one control voltage may det\
ermine 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 synthes\
izer 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."),
quiz::Quiz("Controller","In the broadest sense, a controller is any device that is u\
sed 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 device\
s. Other examples of controllers in the recording studio can include monitor control\
lers, DAW controllers and DJ controllers."),
quiz::Quiz("Corner Frequency","The cutoff or corner frequency of a filter is the poi\
nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
tered” – reduced in loudness – the further above 500Hz you go."),
quiz::Quiz("Counter Clockwise","Counter-clockwise, usually in the context of rotatin\
g a control the left (in the opposite direction of how a clock's hands move)."),
quiz::Quiz("CPU","Abbreviation for Central Processing Unit, the main “brain” chip in\
 a computer (also known simply as “Processor”)."),
quiz::Quiz("Critical Distance","The distance from the sound source at which the dire\
ct sound and the reverberant sound are at equal volume. Critical distance varies acc\
ording to the space; in a room with absorbent walls, the critical distance will be f\
urther from the source, and in a reverberant room, the distance will be closer to th\
e source."),
quiz::Quiz("Crossfade","An audio editing technique in which one sound is faded out a\
s 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 reco\
rded track into a composite take. Club DJs also use crossfading to transition from o\
ne song to the next with no stops."),
quiz::Quiz("Crossover Frequency","The frequency at which the crossover stops sending\
 the signal to one speaker and starts sending it to another."),
quiz::Quiz("Crossover","An audio filter component that splits an audio signal into t\
wo or more bands or signals, usually to be fed into different components of a loudsp\
eaker system according to frequency range. (Also called a “crossover network.”)"),
quiz::Quiz("Crosstalk","The unwanted leakage of an audio signal between two audio ch\
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annels—for example, overlapping signals between channels on a mixing console, or ove\
rlapping audio between two tracks of audiotape."),
quiz::Quiz("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) Th\
e 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 fo\
r video production."),
quiz::Quiz("Cutoff Frequency","The cutoff or corner frequency of a filter is the poi\
nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
tered” - reduced in loudness - the further above 500Hz you go."),
quiz::Quiz("Cutoff Slope","The rate of reduction of the frequencies beyond the passb\
and of a filter. The slope is described as the number of dB the filter reduces the s\
ignal for each octave past the cutoff frequency."),
quiz::Quiz("CV/Gate","This is the shorthand to say a synthesizer may be controlled b\
y voltages - usually for pitch - and gate signals to indicate when a note is “on.” A\
n increasing number of controller keyboards are including CV/Gate output in addition\
 to the customary MIDI (Musical Instrument Digital Interface), making them much easi\
er to connect to a modular synthesizer, as no additional MIDI to CV interface is req\
uired."),
quiz::Quiz("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 functio\
ns and parameters of a module. For example, one control voltage may determine the pi\
tch played by an oscillator; a second control voltage may determine how loud that si\
gnal is after it’s passed through a voltage-controlled amplifier. CV is the most com\
mon shorthand to refer to control voltage - for example, when a synthesizer module s\
ays it features “CV over the filter’s resonance,” that means there is a control volt\
age input to control the amount of resonance (feedback) - not just the customary kno\
b on the front panel."),
quiz::Quiz("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 kno\
wn control voltage processor out there; you will also find some modules with CVP spe\
cifically in their name. Regardless, it’s good to have one or more of this type of m\
odule in your system to help massage voltages to get them to do what you want (or to\
 teach them new tricks)."),
quiz::Quiz("CW","Clockwise, as in rotating a control the the right - in the same dir\
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ection as a clock's hands move."),
quiz::Quiz("Cycle","One complete expression of a waveform beginning at a certain poi\
nt, progressing through the zero line to the wave’s highest and lowest points, and r\
eturning to the same value as the starting point. One complete vibration or sound wa\
ve."),
quiz::Quiz("D-Sub Connector","Abbreviation for “D-subminiature connector,” a D-sub i\
s a multipin connector that is most often used to connect a computer to a VGA monito\
r, but also used occasionally in digital audio applications in the recording studio.\
"),
quiz::Quiz("D/A","Abbreviation for Digital to Analog conversion, which changes digit\
al data numbers (digital audio signal) into discrete voltage level. The reverse proc\
ess of A/D. Also known as DAC."),
quiz::Quiz("DADSR","This is a slightly fancier take on the standard ADSR envelope ge\
nerator that introduces an initial timed delay before the initial attack stage (risi\
ng 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 n\
ote starts, and then after a pause it starts to swell into a brighter, fuller sound.\
"),
quiz::Quiz("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 secon\
d to reach the third, etc."),
quiz::Quiz("Damping Factor","Describes an amplifier’s ability to restrain the pushba\
ck motion (back-EMF) of the loudspeaker cone when the audio signal stops."),
quiz::Quiz("Damping","The reduction of energy in a vibrating system, through frictio\
n. Can refer to the reduced amplitude in an electrical signal, or the stifled vibrat\
ions of a musical instrument (for example, the damper pedal on an acoustic piano).")\
,
quiz::Quiz("DAW","An abbreviation for Digital Audio Workstation, a device or softwar\
e program designed for recording and mixing audio digitally."),
quiz::Quiz("dB","An abbreviation for decibel, a measurement ratio that compares sign\
al strengths (usually audio levels)."),
quiz::Quiz("DBX","A series of noise reduction systems, named for the company that de\
veloped them. DBX noise reduction has been less commercially successful than the mor\
e widely known Dolby systems, but is still found on occasion in recording studios.")\
,
quiz::Quiz("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, a\
s an AC coupled input would try to remove the DC component of the signal (such as it\
s sustain level) and return it to 0v."),
quiz::Quiz("DC","Electrical current that flows in a single direction, as opposed to \
Alternating Current (AC), which flows in alternating directions. Many electronic dev\
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ices run on DC, which is usually provided by battery power, USB power or an AC adapt\
er plugged into the wall. In modular terms, DC refers to a voltage that tends to sta\
y at one steady level for awhile, such as a gate output that switches between 0v whe\
n a note is off and 5 or 10v when a note is on. It can also refer to a slowly changi\
ng voltage, such as an envelope."),
quiz::Quiz("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 p\
itch that cause subtle detuning (and therefore the perception of “fatness”) when usi\
ng more than oscillator per voice."),
quiz::Quiz("De-esser","An audio compressor designed to reduce the volume of sibilant\
 sounds and frequencies, especially those produced by pronouncing the letter “s.”"),
quiz::Quiz("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."),
quiz::Quiz("Decca Tree","A stereo microphone placement technique involving three mic\
rophones (usually omnidirectional) placed in a “T” pattern. Commonly used in miking \
choirs, orchestras and other large ensembles, but variations of the Decca tree techn\
ique are also being used today in surround sound situations."),
quiz::Quiz("Decibel","(abbreviated “dB“) The ratio measurement of two levels accordi\
ng to a scale where a certain percentage change comprises one unit. Most often used \
to describe audio levels."),
quiz::Quiz("Degaussing","The process of demagnetizing an object. In the context of a\
udio, degaussing essentially erases the recording on magnetic tape."),
quiz::Quiz("Delay/Attack/Decay/Sustain/Release","This is a slightly fancier take on \
the standard ADSR envelope generator that introduces an initial timed delay before t\
he 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 mut\
ed, filtered sound when the note starts, and then after a pause it starts to swell i\
nto a brighter, fuller sound."),
quiz::Quiz("Delay","You all know what the word delay means in the normal world; it c\
an appear in different forms inside a modular synth. For example, it can refer to th\
e 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 signa\
l, such as a gate, trigger, or initial stage of an envelope so a note would start la\
ter than it was actually played. Also, 1) An process by which an audio signal is rec\
orded to a medium or device, reproduced at a time delay, then mixed with the origina\
l, non-delayed signal to create a variety of effects such as a fuller sound, echo, c\
horusing, flanging, etc.  2) A signal processor that creates delay effects."),
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quiz::Quiz("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 minima\
l tracking or production, almost like a “rough draft” of a recording."),
quiz::Quiz("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 th\
an 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 interes\
ting and “full” – and a bit more natural, as two singers or instruments can rarely h\
it 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 stu\
dio, but is often used in “double-tracking,” blending the detuned instrument/track w\
ith the original to create a fuller sound."),
quiz::Quiz("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."),
quiz::Quiz("Dialogue","The spoken word recorded in film/video sound, commercials and\
 instructional recordings."),
quiz::Quiz("Diaphragm","The part of a microphone that moves in response to sound wav\
es, converting them to electrical signals."),
quiz::Quiz("Difference","A fancy way of saying you subtracted on control voltage fro\
m another. It can also be applied to audio or harmonics."),
quiz::Quiz("Digital Audio Workstation","abbreviated DAW) A device or computer softwa\
re 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 co\
mmonly 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 in\
clude Reason, Ableton and Logic."),
quiz::Quiz("Digital Multimeter","A small device that tests electrical voltage, curre\
nt, and resistance. Multimeters are useful in recording studios for calibrating elec\
trical systems and troubleshooting problems."),
quiz::Quiz("Digital Recording","The process of converting audio signals into numbers\
 that represent the waveform, then storing these numbers as data."),
quiz::Quiz("Digital Signal Processing","(abbreviated “DSP”) Any signal processing do\
ne after an analog audio signal has been converted into digital audio."),
quiz::Quiz("Digital to Analog Converter","(abbreviated D/A) A device that converts t\
he digital data of digital audio into voltage levels that approximate the original a\
nalog audio."),
quiz::Quiz("Digital","There was a time when digital (referring to circuitry based ar\
ound 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 wor\
d among synthesists. The assumption was digital techniques created sounds that were \
more sterile, brittle, and abrasive – and just not as “authentic.” Today, digital ci\
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rcuitry is embraced in synthesizers, including modular systems. Although analog will\
 always hold a special place in our hearts, a well-implemented digital circuit can s\
ound just as good as an analog one, while digital signal processing and programming \
can create a wider range of sounds than most analog circuitry."),
quiz::Quiz("Digitally Controlled Oscillator","A DCO (Digitally Controlled Oscillator\
) is a hybrid design for an analog oscillator that – instead of using a voltage leve\
l 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, the\
re are no imperfections in pitch that cause subtle detuning (and therefore the perce\
ption of “fatness”) when using more than oscillator per voice."),
quiz::Quiz("DIN Stereo","A stereo microphone placement technique that places two car\
dioid microphones about 20cm apart and set outward from each other at a 90-degree an\
gle to create a stereo image.  Particularly for stereo miking at close ranges. (See \
also “Near-Coincident Miking.”)"),
quiz::Quiz("DIN Sync","A clock signal for controlling the tempo of sequencers, arpeg\
giators, and drum machines, distributed using cables with DIN-style connectors (yes,\
 just like old-fashioned MIDI connectors, but DIN Sync is even older). Roland pionee\
red 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 a\
t 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."),
quiz::Quiz("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 resona\
nce."),
quiz::Quiz("Diode","An electrical component that enables easy electrical current flo\
w in one direction but not the other. In the recording studio, these are commonly fo\
und in the vacuum tubes of tube amplifiers."),
quiz::Quiz("Direct Box","A small device that to converts an unbalanced, high-impedan\
ce speaker or instrument-level output to a balanced, low-impedance mic-level output.\
 Frequently used in the signal path connecting electric instruments “directly” to th\
e mixing console, as opposed to miking them acoustically. Also called “direct inject\
ion box” or “DI box.”"),
quiz::Quiz("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 cha\
nging voltage, such as an envelope. (abbreviated “DC“) Electrical current that flows\
 in a single direction, as opposed to Alternating Current (AC), which flows in alter\
nating directions. Many electronic devices run on DC, which is usually provided by b\
attery power, USB power or an AC adapter plugged into the wall."),
quiz::Quiz("Direct Injection","(abbreviated “DI”) The process of sending an electric\
al audio signal directly from an instrument to the mixing console through the use of\
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 electric pickups or direct boxes, as opposed to using a microphone."),
quiz::Quiz("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. T\
his feature is often used to send a “dry” signal to a monitor mix or a recording dev\
ice."),
quiz::Quiz("Direct Sound","The sound that reaches a microphone or a listener’s ear w\
ithout hitting or bouncing off any obstacles (as opposed to reflected or ambient sou\
nd)."),
quiz::Quiz("Directional Pattern","1) In microphones, a term meaning the same thing a\
s “Pick Up Pattern,” a description of the area in which a microphone is most sensiti\
ve 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."),
quiz::Quiz("Dispersion (also Dispersion Angle)","The area that is effectively covere\
d 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 s\
tanding directly in front of the speaker. Each speaker has both a horizontal and ver\
tical dispersion angle."),
quiz::Quiz("Distant Miking","The technique of placing a microphone far from the soun\
d source in order to pick up a combination of the direct and reflected sounds."),
quiz::Quiz("Distortion","Refers to the deforming of a waveform at the output of a de\
vice as compared with the input, usually due to overload, creating a distorted or “d\
irty” signal. While electrical or audio distortion is typically unwanted and avoided\
, it is frequently used in controlled situations in audio to create certain desirabl\
e effects, particularly with electric guitars and amplifiers."),
quiz::Quiz("Diversity","1) In audio settings: the use of two or more antennas in a w\
ireless receiver system to prevent dropouts in the audio from a wireless microphone.\
 2) In other settings: the embracing of the uniqueness of all individuals."),
quiz::Quiz("Dolby","The brand name of a manufacturer of noise reduction systems and \
other audio systems, to improve performance and fidelity of audio recording, playbac\
k, and transmission."),
quiz::Quiz("Doppler Effect","The phenomenon in which the human ear perceives a chang\
e in the frequency (pitch) of a sound while the sound source is in motion. As the so\
und source approaches, the sound waves travel a shorter distance to the ear, increas\
ing 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 frequenci\
es. 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 bet\
ween the sound source and the listener (or microphone) and creating its signature wa\
rbling vibrato effect."),
quiz::Quiz("Double","1) To record a second performance closely matching the first pe\
rformance, for the purpose of blending the two tracks. 2) To use a delay line with m\
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edium delay to simulate double tracking."),
quiz::Quiz("Driver","1) A transducer in a loudspeaker that converts electrical signa\
ls into sound pressure waves. 2) A computer program that controls an attached device\
 or piece of hardware."),
quiz::Quiz("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 unwan\
ted dip in audio, a crackle or a pop."),
quiz::Quiz("Drum Machine","An electronic device containing synthesized and/or sample\
d drum sounds in its memory, along with an internal sequencer that can be programmed\
 to play drum patterns or loops."),
quiz::Quiz("Drum Pattern","A specific sequence of drum sounds played by a drummer or\
 sequenced into a drum machine for use in a song."),
quiz::Quiz("Dry","A sound with no effects is referred to as \"dry\"; a sound with ef\
fects (such as reverb) mixed is referred to as \"wet.\" Effects units or mixers ofte\
n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
d and the fully-effected sound."),
quiz::Quiz("DSP","Any signal processing done after an analog audio signal has been c\
onverted into digital audio."),
quiz::Quiz("Dub (or Dubbing)","1) To copy a recording. 2) To record in real time wit\
h 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 ele\
ctronic music."),
quiz::Quiz("Ducking","A compression-based audio effect in which an audio signal is r\
educed proportionately by the presence of another audio signal, sometimes accomplish\
ed 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 e\
ngineer “ducks” specific sounds to make room for others in the track; for example, w\
hen a bass guitar signal triggers a slight reduction in the level of drums or guitar\
s. (See also “Sidechain.”)"),
quiz::Quiz("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 t\
hey could play two separate notes as once. Some MIDI interfaces for modular synths i\
nclude 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;"),
quiz::Quiz("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."),
quiz::Quiz("Dynamic Microphone","(Also called Moving Coil Microphone) A microphone i\
n which sound pressure waves are converted to an electrical audio signal by an induc\
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tion coil moving within a magnetic field—a process often compared to a loudspeaker w\
orking in reverse. Dynamic microphones are less sensitive than condenser microphones\
, but can be effective for miking louder sound sources or for close-miking applicati\
ons."),
quiz::Quiz("Dynamic Processing/Dynamic Signal Processing","The process of automatica\
lly changing the level (or gain) to alter the level relationship of the loudest audi\
o to the softest audio. Dynamic processors include compressors, limiters, expanders \
and gates."),
quiz::Quiz("Dynamic Range","1) The ratio (in dB) between the loudest peak and the so\
ftest level of a song or recording. 2) The ratio (in dB) between the softest and lou\
dest possible levels a device or system can provide without distortion."),
quiz::Quiz("Early Reflections","The first sound waves that reach a listener’s ear af\
ter bouncing off a surface in the room, usually heard almost immediately after the i\
nitial sound. The first stage of reverberation."),
quiz::Quiz("East Coast Synthesis","This blanket term is applied to most common synth\
esizer 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 osci\
llators 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 subt\
ractive synthesis, as the filter reduces (subtracts) harmonics that came from the os\
cillators. East Coast synthesizers also regularly have organ-style black & white key\
boards, and four stage ADSR type envelopes. Today it's common to mix both East Coast\
 and West Coast approaches in the same system."),
quiz::Quiz("Echo Chamber","An enclosed room designed with reflective, non-parallel s\
urfaces for the purpose of creating acoustic echoes (reverberation)."),
quiz::Quiz("Echo","The distinct repetition of an initial sound, caused by the reflec\
tion of the sound waves upon a surface. We recognize a sound as an echo when the dis\
tance between the source and the reflection is far enough apart that we can detect t\
he time delay between one and the other. Essentially, reverberation is the combinati\
on of many echoes occurring too rapidly to hear each individually. In the studio, ec\
hoes can be reproduced acoustically or simulated by a digital signal processor."),
quiz::Quiz("Edit","To change one or more parameters of a recorded sound after the fa\
ct. 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 g\
raphically; changing the sequence of playback; and many others. Analog editing would\
 typically involve splicing the magnetic tape on which the audio signals were record\
ed. These days, almost all editing in the studio is done via computer using a digita\
l audio workstation (DAW)."),
quiz::Quiz("Effect Loop","Sometimes you might want to send a signal outside your mod\
ular system, process it through an external effects device, and bring it back into y\
our 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\
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 to be much higher than those used by external effect equipment, so a modular effect\
 loop will usually have level matching circuitry as well."),
quiz::Quiz("Effects Processor","(Also called Guitar Processor) A device that adds au\
dio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, o\
verdrive, amplifier simulation, etc. Effects processors can occur as individual effe\
cts boxes or multi-sound pedal boards (see also “Foot Pedals,” “Foot Switches”) adde\
d 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."),
quiz::Quiz("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 m\
ultitrack recording."),
quiz::Quiz("Effects","1) Various ways an audio signal can be modified by adding some\
thing 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 fi\
lm or video."),
quiz::Quiz("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 (Vo\
ltage Controlled Filter). To do this, and envelope generator creates a voltage that \
typically rises from zero volts to some maximum level, and back down again. You cont\
rol how long this takes, usually in various stages: an attack stage as it goes from \
zero to max, a decay stage as if 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 sust\
ain, and then (usually after a key has been released and the corresponding gate sign\
al has gone back to zero) from the sustain level back to zero over a duration known \
as its release."),
quiz::Quiz("Electret Microphone","A variation of condenser microphone that uses an e\
lectret instead of a capacitor. (Also called “Electret Condenser Microphone.”) Becau\
se the electret is permanently polarized, an electret microphone does not require an\
 external power source as a standard condenser microphone does."),
quiz::Quiz("Electret","A dielectric plate that is designed with permanent polarity, \
allowing it to function similarly to a magnet. (“Electret” comes from the words “ele\
ctricity” and “magnet.”) Used in some microphone types in place of a capacitor (cond\
enser)."),
quiz::Quiz("Electromagnetic Field","(Abbreviated EMF) A field of magnetic energy put\
 out because of current traveling through a conductor."),
quiz::Quiz("Electromagnetic Interference (EMI)","The bane of audio professionals eve\
rywhere, EMI is a type of interference caused by nearby electromagnetic activity, wh\
ich can be picked up by audio cables and equipment, causing unwanted noise, hum or b\
uzz in audio systems. Common causes of EMI in audio systems may include high-current\
 power lines, fluorescent lighting, dimmer switches, computers, video monitors and r\
adio transmitters."),
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quiz::Quiz("Electrons","Negatively charged particles revolving around the nucleus of\
 an atom. Electrical current is generated by electrons moving along a conductor, lik\
e a metallic wire."),
quiz::Quiz("Emphasis","This word can have two meanings. In a normal audio context, i\
t usually means some form of high frequency boost, as emphasizing the higher harmoni\
cs 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 se\
tting boosts (emphasizes) the harmonics at the cutoff or corner frequency."),
quiz::Quiz("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 p\
erform 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."),
quiz::Quiz("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 Ampl\
ifier), as well as how its frequency content or timbre changes over time when connec\
ted to a VCF (Voltage Controlled Filter). To do this, and 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 if falls back down from maximum to eit\
her zero (in the case of an AD, or Attack/Decay envelope) or an intermediate level k\
nown as the sustain, and then (usually after a key has been released and the corresp\
onding gate signal has gone back to zero) from the sustain level back to zero over a\
 duration known as its release."),
quiz::Quiz("Envelope Tracking","This describes the main action of an envelope follow\
er: a module or section of a module that follows the loudness of a signal and output\
s a voltage that corresponds to - tracks - that input."),
quiz::Quiz("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 describe\
s how a sound or audio signal varies in intensity over a period of time."),
quiz::Quiz("Equal Loudness Contours","A drawing of several curves showing how loud t\
he tones of different frequencies would have to be played for a person to say they w\
ere of equal loudness. (See also “Fletcher-Munson Curves.”)"),
quiz::Quiz("Equalizer","An audio signal processor that uses one or more filters to b\
oost or cut the amplitude (volume) of certain frequencies within the sound. The unde\
rlying principle is to balance or “equalize” the frequency response of the audio sys\
tem, 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.")\
,
quiz::Quiz("Eurorack","Eurorack is arguably the most popular format of modular synth\
esizer today, with over 100 manufacturers and over 1000 modules available. It was cr\
eated by Doepfer Musikelektronik in 1995, basing its size off the Eurorack format fo\
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r lab equipment. Some users will try to tell you that Eurorack doesn't “sound” as go\
od as other formats, but that's just based on a few substandard manufacturers or mod\
ules; there's nothing inherent to the standard that makes a huge difference in the f\
inal sound (no; the difference between 12 and 15 volt power supplies is not enough t\
o most ears)."),
quiz::Quiz("Expander","A signal processor (or plug-in) that performs the opposite fu\
nction of a compressor, expanding the dynamic range of an audio signal rather than c\
ompressing it. It accomplishes this by further reducing the amplitude of signals tha\
t drop below a set threshold."),
quiz::Quiz("Expansion Ratio","The rate by which an expander attenuates an incoming s\
ignal, measured in decibels. For example, an expansion ratio of 2:1 means the expand\
er 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.\
"),
quiz::Quiz("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 a\
n 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 i\
n the same increase of cycles per second."),
quiz::Quiz("Fade","A gradual reduction of the level of the audio signal, or a gradua\
l change of level from one pre-set level to another."),
quiz::Quiz("Fader","A control which adjusts the level (gain or attenuation) of an in\
coming signal to a channel or grouping of channels on a console."),
quiz::Quiz("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 fa\
r field begins around 3 feet from the sound source. Audio engineers often use both n\
ear field and far field monitoring when fine-tuning a mix. (See also “Critical Dista\
nce,” “Near Field.”)"),
quiz::Quiz("Feed","To send an audio or control signal to."),
quiz::Quiz("Feedback Control","The control on a delay line or delay effects device t\
hat controls the amount of feedback into the system."),
quiz::Quiz("Feedback","The return of a portion of the output signal back into the in\
put 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 i\
s picking up the sound from a nearby speaker, generating a loud, oscillating frequen\
cy 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."),
quiz::Quiz("Fidelity","A term describing how accurately a sound is reproduced from i\
ts original source."),
quiz::Quiz("Figure-8 Pattern","A microphone pickup pattern which is most sensitive t\
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o picking up sounds directly in front and back of the mic, effectively rejecting sou\
nds coming from the sides."),
quiz::Quiz("Filter","A module that reduced or removes certain frequencies and harmon\
ics from the sound that is passed through it. In a synthesizer, the most typical fil\
ter types are low pass (passes all of the harmonics below its cutoff or corner frequ\
ency untouched, and then reduces the level of higher harmonics the further you go ab\
ove that cutoff frequency), high pass (passes all harmonics above its cutoff frequen\
cy untouched, and reduces the level of progressively lower harmonics below the cutof\
f), 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 no\
tch (harmonics right around the cutoff frequency are reduced or cut out entirely; ot\
hers above or below are allowed to live)."),
quiz::Quiz("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 inpu\
t signal is delayed by a very small amount (longer than a chorus effect; shorter tha\
n an echo effect) and mixed in with the original. When the delay is constant, the re\
sult 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 th\
e flange of one of the tape reels to delay it."),
quiz::Quiz("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 “swi\
rling” sound. This was originally accomplished in analog tape recording by playing t\
he original tape and the copy on two tape machines simultaneously, then physically p\
ressing on the flange of one of the machines to alter the timing of the duplicate tr\
ack. These days, most flanging is done through delay boxes or digital plug-ins."),
quiz::Quiz("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 nota\
tion, an “accidental” mark that instructs the player to play/sing the note one-half \
step lower."),
quiz::Quiz("Fletcher-Munson Curves","Also known as “Equal Loudness Contours,” a set \
of graphical curves plotted to illustrate how the human ear responds to different fr\
equencies at different volume levels. Named after the two researchers who first plot\
ted the curves. (See also “Equal Loudness Contours.”)"),
quiz::Quiz("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 ou\
tput 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."),
quiz::Quiz("Floating Unbalanced Line","A connection “workaround” in which an unbalan\
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ced output is connected to a balanced input by modifying the connections in the line\
 to resemble a balanced line, alleviating unwanted hum or buzz."),
quiz::Quiz("Fly In","To add sounds into a mix or recording that have no synchronizat\
ion."),
quiz::Quiz("Flying Bus","This is a very simple type of power distribution or bus boa\
rd that typically uses a ribbon cable with multiple connectors along its length to t\
ake the output of your power supply and distribute it to your individual modules. Th\
ey're cheap and easy to install and use, but in a few cases might be a cause of nois\
e being shared between modules."),
quiz::Quiz("FM","Frequency modulation (FM for short) refers to a synthesis technique\
 where the pitch of an oscillator is varied (modulated) very quickly - at audio rate\
s - 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."),
quiz::Quiz("FOH","In live audio settings, the location in a venue opposite the stage\
, where live audio for the show is controlled and mixed."),
quiz::Quiz("Foldback","A stage monitoring system used in live audio. A set of on-sta\
ge 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 t\
o play. This mix is usually different from the FOH (front-of-house) mix that the aud\
ience 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 freq\
uently 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 goo\
d mix for the audience. For this reason, many live audio systems now use in-ear moni\
toring as an alternative to stage monitors to control the onstage noise and reduce t\
he risk of feedback."),
quiz::Quiz("Foot Pedal","An effects device controlled by a musician with his foot."),
quiz::Quiz("Foot Switch","A switch placed on the floor and pressed by a musician to \
do various functions."),
quiz::Quiz("Force-Sensing Resistor","In modular systems, an FSR (Force-Sensing or -S\
ensitive Resistor) usually takes the form of a circular pad that you press on to var\
y a parameter. It acts as a resistor that decreases in resistance the harder you pre\
ss."),
quiz::Quiz("Formant","Many instruments based on vibrating tubes - including our own \
vocal tract - have certain frequencies that they like to vibrate or “resonate” at. W\
hen you send a sound down these tubes, they will accentuate the frequency of that so\
und (or some of its harmonics) to match these resonate frequencies. Each of these re\
sonant frequencies is known as a formant of that instrument. A common way of synthes\
izing 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 i\
s 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 reso\
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nances that occur as a result of the physical structure of the sound source. These f\
requencies are what create timbre, that element of sound that creates the specific s\
ound of a guitar, a flute, a male or female voice, etc."),
quiz::Quiz("Format","1) One of many different media used to store and reproduce audi\
o, whether in the recording studio or for listening purposes. Examples include curre\
ntly 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 reco\
rding to analog tape, such as number of tracks, width, spacing and order. 3) To prep\
are a hard drive or memory card for use, usually erasing all existing data in the pr\
ocess."),
quiz::Quiz("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-qua\
drant 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 usua\
lly follow. Namely, the modulation frequency is both added to and subtracted from th\
e carrier's frequency; the resulting harmonics replace the original carrier and modu\
lator. Say the carrier was a sine wave (only the fundamental harmonic present) at 60\
0Hz, and the modulator was a sine wave at 100Hz. The result would be a tone that had\
 frequency components at 500 and 700Hz."),
quiz::Quiz("FracRack","A less-common format of modular synthesizers put forward by P\
AiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5" (3.8 cm) wi\
de by 3U, or 5.25" (13.3 cm) high."),
quiz::Quiz("Fractional Rack","A less-common format of modular synthesizers put forwa\
rd 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."),
quiz::Quiz("Frequency Modulation (FM) Synthesis","A method of sound synthesis in whi\
ch the frequencies generated by one oscillator (the carrier) are altered by the outp\
ut of one or more additional oscillators (operators) to create a diversity of harmon\
ically rich sounds."),
quiz::Quiz("Frequency Range","1) The range of frequencies over which an electronic d\
evice puts out a useful signal (see also "Bandwidth"). 2) The range of frequencies t\
hat 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 al\
so hear within a certain frequency range."),
quiz::Quiz("Frequency Response","The range between high and low frequencies that a c\
omponent of an audio system can adequately handle, transmit or receive."),
quiz::Quiz("Frequency-Agile","In wireless microphone systems, frequency-agile descri\
bes the ability of the system to operate on a choice of different RF frequencies wit\
hin a certain bandwidth. Frequency-agile systems are preferred for live touring and \
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in areas with high concentrations of radio signals (like large cities) because the R\
F frequency of the device can be changed to avoid interference."),
quiz::Quiz("Frequency-Shift Key (FSK)","A now out-of-date protocol in which a sync t\
one is recorded onto a spare track of a multi-track tape recorder to enable electron\
ic 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.”)"),
quiz::Quiz("Frequency","The number of occurrences of a particular event within a cer\
tain amount of time. In audio and acoustics, frequency specifically refers to the nu\
mber of complete cycles a vibration or waveform makes in a second, measured in cycle\
s per second, or Hertz (Hz). In sound, frequency determines what we hear as pitch. T\
he longer the wavelength, the fewer the cycles per second, and the lower the pitch."\
),
quiz::Quiz("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\
."),
quiz::Quiz("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 a\
cts as a resistor that decreases in resistance the harder you press."),
quiz::Quiz("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 pat\
h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\
rd into either the input or output side; when it is “half-normalled,” the path chang\
es 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 correspond\
ing output."),
quiz::Quiz("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."),
quiz::Quiz("Function Generator","The term function generator can have two meanings i\
n 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 sin\
e or creating a rise & fall in response to a trigger."),
quiz::Quiz("Fundamental","(Also called fundamental frequency or first harmonic) The \
lowest frequency present in the sounding of a note by musical instrument or voice.")\
,
quiz::Quiz("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."),
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quiz::Quiz("Gain Reduction","The action of a compressor or limiter in regulating the\
 amplitude of the audio signal."),
quiz::Quiz("Gain Structure","A term that describes the interconnection of multiple c\
omponents in an audio system, and the amount of gain increase or reduction that occu\
rs at each point. A configuration with a good gain structure means that the componen\
ts are working properly together to provide optimal gain with minimal distortion or \
noise."),
quiz::Quiz("Gain","The amount of increase in audio signal strength, often expressed \
in dB."),
quiz::Quiz("Gate Detector","This is one of the main signal types that are passed aro\
und 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 controll\
er), or when a sequencer jumps to the next “stage” or note. A gate typically stays a\
t that level for the duration of the note (i.e. while the key is being held down), a\
nd suddenly drops or “goes low” to its resting level - typically 0 volts, but someti\
mes -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 De\
cay stages; while the gate remains high, the envelope stays at its Sustain level, an\
d when the gate goes low again, the envelope moves onto its Release stage."),
quiz::Quiz("Generation Loss","The amount of clarity lost when recorded audio is copi\
ed, due to added noise and distortion."),
quiz::Quiz("Generation","A term used to describe the number of times that the record\
ed audio signal has been copied."),
quiz::Quiz("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 volta\
ge 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 sl\
ew generator, slew limiter, slope generator, or lag. Some use the terms glide, gliss\
ando, and portamento interchangeably, but if you want to split musical hairs, a glis\
sando (gliss) is a different effect where the intermediate notes are more distinct -\
 such as played rapidly in order - rather than slurred through."),
quiz::Quiz("Golden Section","(also called Golden Ratio) A ratio of height to width t\
o length, where the width is approximately 1.6 times the height, and the length appr\
oximately 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, inc\
luding room dimensions and studio design (to achieve “optimal acoustics” in the room\
), and even in the design of certain acoustic instruments."),
quiz::Quiz("Granular Synthesis","Granular synthesis can be thought of as particle th\
eory applied to sound. The concept is that a sound can be broken down into very smal\
l “grains” - typically 1-50 or 100 msec in duration. These tiny snippets are then pl\
ayed back to reproduce the original sound, or to create new sounds by changing the s\
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peed, 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 ca\
n range from audio processing tricks such as changing speed without changing pitch a\
nd vice versa, to creating psychedelic “clouds” of sound (and indeed, there is a pop\
ular module called Clouds)."),
quiz::Quiz("Graphic Equalizer","A type of equalizer that can adjust various frequenc\
ies of the incoming signal using sliders that are assigned to specific frequency ban\
ds. (See also “Equalizer.”)"),
quiz::Quiz("Ground Lift Plug","An adapter that enables a three-prong power cord to p\
lug into two-prong outlet. Some engineers wrongly use this plug to interrupt the gro\
und connection and prevent buzz, but it is a VERY unsafe practice to break the groun\
d connection using this plug without grounding the unit by another means."),
quiz::Quiz("Ground Lift Switch","A switch that breaks the connection between the gro\
und point in one circuit and the ground point in another circuit, for the purpose of\
 eliminating hum or buzz caused by ground loops."),
quiz::Quiz("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-freque\
ncy hum that can be annoying at best and cause damage to gear at worst. The best res\
olution 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 man\
agement of electricity."),
quiz::Quiz("Group (or Grouping)","A number of input channels on a console that can b\
e controlled and adjusted as a single set before sending the combined signal to the \
master output. Sometimes also called “Submix,” “Bus” or just “Group.”"),
quiz::Quiz("Group Delay","In audio, group delay is a phenomenon within all electroni\
c audio devices (e.g., speakers, amplifiers) in which different frequencies in the s\
ignal are output at slight delays from one another. In simpler terms, lower frequenc\
ies 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 a\
ny configuration is to keep the group delay as small as possible; in cases of extrem\
ely poor configurations, the delays between highs and lows can be audible."),
quiz::Quiz("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.\
"),
quiz::Quiz("Guitar Processor","A device that adds audio effects to a direct guitar s\
ignal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, \
etc. Effects processors can occur as individual effects boxes or multi-sound pedal b\
oards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between t\
he guitar and the console. They can also be found as presets in guitar amplifiers, o\
r even as digital plug-ins within a DAW."),
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quiz::Quiz("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 t\
han 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 direc\
tion of the sound. This is what helps us distinguish the original sound source witho\
ut being confused by reflections and reverberations off of nearby surfaces. Understa\
nding the Haas effect is particularly useful in live audio settings, especially in l\
arge venues where loudspeakers are time-delayed to match the initial sound waves com\
ing from the source."),
quiz::Quiz("Half Step","A change in pitch equivalent to adjacent keys on a piano. Al\
so known as a “semitone.”"),
quiz::Quiz("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 pat\
h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\
rd into either the input or output side; when it is “half-normalled,” the path chang\
es 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 correspond\
ing output."),
quiz::Quiz("Half-Wave Rectifier","A half-wave rectifier passes only positive voltage\
s, and replaces anything negative with 0v. In other words, anything “below zero” is \
clipped off."),
quiz::Quiz("Hall Program","A setting of a digital delay/reverb effects unit that app\
roximates concert halls. Hall programs are characterized by pre-delay of up to 25 ms\
."),
quiz::Quiz("Hard Knee","In compression, refers to a more abrupt introduction of comp\
ression of the signal once the sound level crosses the threshold. (See also “Knee.”)\
"),
quiz::Quiz("Hard Sync","This is the most common type of oscillator sync where the sl\
ave oscillator will reset its waveform whenever it receives a sync pulse. If the typ\
e of sync is not specified, then it’s probably hard sync."),
quiz::Quiz("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 chan\
ging the instrument’s timbre."),
quiz::Quiz("Harmonic","A single harmonic is the purest sound possible: It contains n\
o 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. M\
ost of the time, overtones have a very specific pitch relationship to each other. Th\
e 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, a\
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nd spaced out as integer multiples of the fundamental: two times its frequency, thre\
e times, four times, and so forth. The first few harmonics happen to have a nice mus\
ical spacing: an octave; an octave and a fifth; two octaves. But the higher they get\
, the less musical they may seem."),
quiz::Quiz("Harmonics","Whole number multiples of the fundamental frequency that occ\
ur naturally within the playing of a tone. Mathematically, if the fundamental freque\
ncy is x, the harmonics would be 2x, 3x, 4x, etc. For example, if the fundamental fr\
equency 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 timbr\
e of an instrument or voice."),
quiz::Quiz("Head","In tape recording, an electromagnetic transducer that magneticall\
y 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 purp\
oses. An erase head creates a powerful electromagnetic field to the tape to erase pr\
evious signals from the tape."),
quiz::Quiz("Headroom","The difference in dB between normal operating level and clipp\
ing level in an amplifier or audio device.  Also describes the difference in dB betw\
een the peak levels of a recording and the point at which the signal distorts. (Also\
 called “Margin.”)"),
quiz::Quiz("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."),
quiz::Quiz("Hertz","(Abbreviated Hz) 1) The unit of measurement for frequency, speci\
fically, the number of complete wave cycles that occur in a second (cycles per secon\
d). 1 Hz = 1 complete wave per second. 2) A popular rental car company (not typicall\
y used in recording except for transport to the studio)."),
quiz::Quiz("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."),
quiz::Quiz("Hi-Z","(abbreviated Hi-Z) Described as an impedance or resistance of sev\
eral thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more o\
hms. (See also “Impedance.”)"),
quiz::Quiz("High (gate)","When a gate signal is at the voltage level (typically 5 vo\
lts, although it can be more) that indicates it is “on” – such as when a note is bei\
ng held down on a keyboard controller – it is said that the gate is high."),
quiz::Quiz("High Impedance","(abbreviated Hi-Z) Described as an impedance or resista\
nce of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000\
 or more ohms. (See also “Impedance.”)"),
quiz::Quiz("High Pass Filter","An audio filter that attenuates signals below a certa\
in frequency (the cut-off frequency) and passes signals with frequencies that are hi\
gher."),
quiz::Quiz("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” i\
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n an audio signal."),
quiz::Quiz("High-Pass Filter","The high pass filter (HPF) design passes harmonics ab\
ove its cutoff or corner frequency untouched, and reduces the level of lower harmoni\
cs depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) hi\
gh 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 bel\
ow the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pa\
ss 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."),
quiz::Quiz("Horizontal Pitch","HP = Horizontal Pitch. In the Eurorack format for syn\
thesizer modules, the width of a module is defined as the number of hp (horizontal p\
itch) units. Each hp is 0.2" (0.5 cm). Most modules are even numbers of hp wide, alt\
hough some are odd numbers. Also, modules tend to be ever so slightly less than exac\
tly some multiple of 0.2" wide, just to make sure you don't run into problems with e\
ver so slightly too wide modules overlapping."),
quiz::Quiz("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."),
quiz::Quiz("House Sync","A reference signal such as SMPTE time code that is used to \
keep all devices in the room in sync."),
quiz::Quiz("HP","HP = Horizontal Pitch. In the Eurorack format for synthesizer modul\
es, the width of a module is defined as the number of hp (horizontal pitch) units. E\
ach 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 multi\
ple of 0.2" wide, just to make sure you don't run into problems with ever so slightl\
y too wide modules overlapping."),
quiz::Quiz("HPF","The high pass filter (HPF) design passes harmonics above its cutof\
f 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 filte\
r, harmonics one octave below the cutoff frequency (in other words, one half the cut\
off 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 ar\
e typically used to create bright sounds where the higher harmonics are much stronge\
r than the fundamental and lower harmonics - for example, the sound of a harpsichord\
."),
quiz::Quiz("Hum","1) The low-frequency pitch that occurs when power line current is \
accidently induced or fed into electronic equipment. The hum reflects the fundamenta\
l 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."),
quiz::Quiz("Hybrid Power Supply","A hybrid power supply uses a lower weight, more ef\
ficient switching power supply to perform most of the drop in voltage - say, from 12\
0v AC to 15v DC - and then uses a linear power supply for the remaining much smaller\
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 drop, such as from 15v to 12v. These are becoming the preferred design in many modu\
lar synthesizer enclosures. Shortcomings with the power supply - too noisy, or not e\
nough - tend to be at the cause of many unexpected problems in modular synthesizers.\
"),
quiz::Quiz("Hypercardioid","A variation of the cardioid microphone pick up sensitivi\
ty pattern in which the shape of the optimal pickup area is tighter and more directi\
onal than cardioid. Hypercardioid microphones are most sensitive directly on-axis in\
 front of the microphone, and begins rejecting sounds between 90-150 degrees off-axi\
s, depending on the tightness of the pattern."),
quiz::Quiz("Hz/V","A system where a change of 1 volt at the input results in a chang\
e in pitch of a fixed number of hertz (cycles per second), rather than a fixed music\
al interval."),
quiz::Quiz("Hz","An abbreviation for the term Hertz, or the unit of frequency."),
quiz::Quiz("IADSR","This is an Attack/Decay/Sustain/Release (ADSR) envelope generato\
r that allows you to start the attack phase at an initial level - the “I” - rather t\
han the customary 0 volts. The envelopes in the Prophet VS, as well as a module from\
 Ladik, have this capability."),
quiz::Quiz("IC","Integrated Circuit - A miniature circuit of many components set on \
semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”\
"),
quiz::Quiz("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 f\
ools our ears into thinking the sound source is in emanating from a specific point i\
nstead 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 com\
ing from performers on the stage, rather than from the speakers."),
quiz::Quiz("Impedance","Refers to the resistance of a circuit or device to alternati\
ng 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 to\
gether. Impedance is abbreviated by the letter Z, and measured in ohms (W)."),
quiz::Quiz("In Line Console","An audio mixing console that is designed and configure\
d so each channel strip can be used for both recording and monitoring functions duri\
ng multitrack recording. This configuration is in contrast to split mixing consoles,\
 which requires separate channels on the board for recording and monitoring function\
s."),
quiz::Quiz("In Phase","The desirable situation in which two or more devices (and the\
ir respective audio signals) are on the same side of the polarity spectrum, producin\
g waveforms that do not conflict or cancel each other out."),
quiz::Quiz("In Port","A jack on a MIDI device or computer that will accept an incomi\
ng data signal."),
quiz::Quiz("Inductance","A characteristic of electrical conductors in which electric\
al charge (voltage) is produced or stored magnetically due to the natural resistance\
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 to change in the electrical current. Inductance is an electromagnetic principle tha\
t 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)."),
quiz::Quiz("Inductor","A device (usually a coil of wire) that converts electrical en\
ergy into stored magnetic energy as electrical current passes through it. Commonly f\
ound in a variety of audio applications such as guitar pickups and loudspeakers."),
quiz::Quiz("Infinite Baffle","A loudspeaker mount or enclosure designed so that soun\
d waves coming from the front theoretically do not reach the back, preventing the so\
und waves from cancelling each other out. The term “infinite” comes from the idea th\
at 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 mo\
unting the speaker on a wall of an enclosed room, or building it inside a sealed cab\
inet large enough to prevent rear sounds from affecting the cone from the back."),
quiz::Quiz("Initial/Attack/Decay/Sustain/Release","This is an Attack/Decay/Sustain/R\
elease (ADSR) envelope generator that allows you to start the attack phase at an ini\
tial level – the “I” – rather than the customary 0 volts. The envelopes in the Proph\
et VS, as well as a module from Ladik, have this capability."),
quiz::Quiz("Input / Output (I/O)","I/O – An abbreviation for “Input/Output.” In audi\
o, it refers to any device, program or system involving the transferring of electric\
al/audio signals or data."),
quiz::Quiz("Input Impedance","The opposition to current flow by the first circuits o\
f a device."),
quiz::Quiz("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)."),
quiz::Quiz("Input","The jack or physical location where a device receives a signal. \
Also refers to the incoming signal itself."),
quiz::Quiz("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 outp\
ut. Commonly used to add processing such as reverb, compression or EQ to a channel o\
r group of channels."),
quiz::Quiz("Instrument Amplifier","A device that has a power amplifier and speaker t\
o reproduce the signal put out by an electric instrument."),
quiz::Quiz("Instrument Out Direct","Feeding the output of an electric instrument (li\
ke an electric guitar) directly to the recording console or tape recorder, as oppose\
d to miking the amplifier."),
quiz::Quiz("Insulator","A substance such as glass, air, plastic, etc., that will (fo\
r all practical purposes) not conduct electricity."),
quiz::Quiz("Integrated Circuit","Integrated Circuit (Abbreviated “IC”) – A miniature\
 circuit of many components set on semiconductor material, used in electronics. A fa\
ncy term for “chip” or “microchip.”"),
quiz::Quiz("Integrator","This function smoothens out an incoming signal so that the \
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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 d\
etail on its uses) or less often as a lag generator or processor."),
quiz::Quiz("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 c\
omputer or other electronics. There are many examples of interfaces in professional \
audio situations, including MIDI (Musical Instrument Digital Interface); audio inter\
faces 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, m\
ix and playback. Even the mixing console is an interface of sorts, connecting the ma\
ny elements of the control room."),
quiz::Quiz("Intermodulation (IM) Distortion","Distortion caused by two or more audio\
 signals of different frequencies interacting with one another. The sum and differen\
ce of the frequencies produce new (usually unwanted frequencies) that didn’t exist i\
n any of the original frequencies."),
quiz::Quiz("Inverse Square Law","A mathematical rule that describes an inverse relat\
ionship between one quantity and the square of another quantity. In plain English, o\
ne 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 obst\
ructions, the sound pressure level will drop by half (6dB) each time the distance fr\
om the sound source is doubled. (This equation is quite useful to audio engineers tr\
ying to provide sound in open-air settings, for example.)"),
quiz::Quiz("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 (f\
or example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pola\
rizer, as it changes the polarity (+ versus -) of a signal. A control voltage invert\
er is often combined with an offset voltage to adjust the output voltage into the de\
sired 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 vol\
tages, you would then need to add 8 volts to that result to get an upside-down (inve\
rted) envelope that still had an overall range of 0 to +8v."),
quiz::Quiz("Inverting Mixer","Most signal mixers make an effort to keep the same pol\
arity of a signal as it passes through the mixer. However, some mixers may invert th\
e 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 addin\
g one waveform or filter mode output out of phase with another coming from the same \
oscillator or filter."),
quiz::Quiz("Isolation","The process of containing sound within a certain area so tha\
t 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 o\
ther sounds away from instrument mics during live recording sessions."),
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quiz::Quiz("IV Cable","You often need to send one signal to multiple destinations. O\
ptions 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 intravenou\
s or “IV” drip), in addition to having 3.5mm plugs at either end."),
quiz::Quiz("Jack","That hole you plug your patch cables into on the face of your syn\
thesizer modules? That’s called a jack. The size and type of jack – 3.5mm, banana, o\
r 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, i\
t’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.)"),
quiz::Quiz("Jam Sync","A process available on some clock or syncing devices which re\
ads 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."),
quiz::Quiz("Karplus Strong","This is a physical modeling synthesis algorithm designe\
d 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, altho\
ugh it can be a high frequency chirp or other sound – is sent to both the output, an\
d 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 o\
f 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 cha\
llenge to patch yourself and play with the results."),
quiz::Quiz("Key","1) In music, the note scale in which a piece of music is written o\
r played, identified by the first note (tonic) of the scale, as in, “Key of C.” 2) T\
he control of a dynamics processing device by an external audio signal through the u\
se of a side chain. 3) A digital or data code that unlocks the use of a device or so\
ftware. Example: Pro Tools is licensed through an iLok ID via the use of a physical \
USB key."),
quiz::Quiz("Keyboard Controller","A piano-styled keyboard that sends out MIDI signal\
s to control other MIDI devices. Most keyboard instruments are equipped with MIDI co\
ntrol capabilities, but dedicated MIDI keyboard controllers emit no audio signals, o\
nly MIDI data."),
quiz::Quiz("Keyboard Tracking","Most modular synths follow a strict relationship bet\
ween 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” (con\
trol voltage) to make it clear they are not attenuated or otherwise modified when co\
ntrolling a function on a module."),
quiz::Quiz("Keyboard","Any musical instrument or computer controlled by pressing a k\
ey."),
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quiz::Quiz("Keytar","A strap-on, lightweight, portable keyboard meant to allow keybo\
ardists the same freedom (not to mention posturing opportunities) as guitarists."),
quiz::Quiz("Kick Drum","The bass drum on a trap drum set, so called because it is pl\
ayed with a kick pedal."),
quiz::Quiz("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 d\
escribing audio frequency ranges or digital sampling rates."),
quiz::Quiz("Knee","A function on a compressor that determines how abruptly or gradua\
lly 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” refe\
rs to an abrupt activation of the compressor, while “soft knee” refers to a more gra\
dual change."),
quiz::Quiz("Krell Patch","Recreating this patch is a challenge many modular musician\
s 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 no\
te has a random pitch, envelope, and duration."),
quiz::Quiz("Lag Generator","This function smoothes out an incoming signal so that th\
e change in voltage level cannot exceed a certain number of volts per second. This c\
auses the result to “lag behind” changes in the input. It is sometimes called a slew\
 limiter or technically as an integrator."),
quiz::Quiz("Layering","Refers to almost any blending of similar multiple musical par\
ts or sounds at once, often combined on one channel or assigned to one controller. I\
n audio recording, layering usually involves recording similar takes of the same ins\
trument or vocal (or duplicating parts with slight delays or chorusing effects) to c\
reate a fuller, richer sound than the vocal/instrument by itself. In sound design, i\
t also refers to blending multiple samples (example: two or more drum sounds) to cre\
ate a fuller sound."),
quiz::Quiz("Lead Sheet","A shorthand form of music notation (similar to a chord char\
t) that displays the basic essential elements of a song so musicians can follow alon\
g without the full notation of every note or expression. Lead sheets most commonly i\
nclude a melody line written in music notation with chord changes above the staff, a\
nd lyrics below it. (See also “Chord Chart.”)"),
quiz::Quiz("Leakage","Sounds from other instruments and sound sources that were not \
intended to be picked up by the microphone."),
quiz::Quiz("Level","The amount of signal strength; the amplitude, especially the ave\
rage amplitude."),
quiz::Quiz("LFO","This module produces repetitive, cycling waves ranging in frequenc\
y 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 lo\
udness of a signal), vibrato (when controlling the pitch of a signal), repetitive fi\
lter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an \
oscillator, and more."),
quiz::Quiz("Limiter","A type of compressor that sharply reduces (limits) the gain of\
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 the signal when the audio level reaches a certain threshold, typically used to prev\
ent overload and signal peaking. A compressor effectively becomes a limiter when its\
 ratio is 10:1 or higher. (See also “Compressor.”)"),
quiz::Quiz("Line Input","Line Input (“Line In”) – An input designed to take a line l\
evel signal."),
quiz::Quiz("Line Level","Most consumer and lower-cost professional audio equipment u\
se a signal level reference known as line level or –10dBV (decibel volts). The most \
common connectors are RCA (phono) or 3.5mm, although 1/4” is also used; the signal i\
s “unbalanced” (it uses two wires: signal and ground). In the line level standard, a\
 sine wave that varies between +/–0.447 volts is considered to be at –10dBV. By cont\
rast, a typical oscillator signal in a modular synthesizer is +/–5 to +/–8 volts. As\
 a result, you will need either an output module in your modular synth or one heckuv\
a 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 sig\
nal to get it up to modular standards to process in your modular synth."),
quiz::Quiz("Line Output","Line Output (“Line Out”) – Any output that sends out a lin\
e level signal, such as the output of a console that feeds a recorder."),
quiz::Quiz("Linear FM","This is often the preferred input response for frequency mod\
ulating (FM’ing) an oscillator, as the result stays in tune while you change the mod\
ulator."),
quiz::Quiz("Linear Power Supply","A linear power supply design takes a higher incomi\
ng 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 signa\
l, at the cost of increased heat and weight (they're not very efficient). Many are m\
oving to a hybrid power supply that combines a switcher with a small linear supply o\
r regulator to get the best of both worlds."),
quiz::Quiz("Linear VCA","A linear voltage-controlled amplifier (VCA) uses a simple m\
athematical 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 lou\
d. 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 exponenti\
al output to a linear VCA, then you will get the desired aural result. Confusing? Th\
at’s why it’s great when an envelope generator or VCA has a switch or control to var\
y 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 swi\
ng as the input. Again, this is fine for altering control voltages, but not for mixi\
ng audio signals; in that case you want a mixer with exponential controls."),
quiz::Quiz("Linear VCO","A linear voltage-controlled oscillator (VCO) follows the vo\
lts/hertz (v/Hz) standard; more common is the exponential volts/octave (v/oct) stand\
ard. The exception is frequency modulation (FM), where a linear control voltage inpu\
t is often preferred to recreate classic style FM as it does not change the fundamen\
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tal pitch of the carrier oscillator."),
quiz::Quiz("Live Recording","A recording session where all the musicians are playing\
 at once with no overdubbing."),
quiz::Quiz("Live Room","The large, main room of the recording studio where most of t\
he instruments and/or vocalists perform. So called, not just because there is room f\
or live performances, but because the room has been acoustically treated to produce \
a pleasing amount of live reverberation."),
quiz::Quiz("Live","1) A term describing a space with a reverberant or reflected soun\
d. In a “live” space, the sound waves are active or “live.”  2) Occurring in real ti\
me, as opposed to previously recorded."),
quiz::Quiz("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 m\
odule when the keyboard is played; “Local Off” disconnects it. “Local Off” is freque\
ntly used to prevent unwanted looping of MIDI messages in some configurations, or wh\
en controlling the internal module via another controller."),
quiz::Quiz("Logic Functions","In a modular synth, control voltages tend to be contin\
uous in nature, while gate and trigger signals are binary: on or off; high or low. T\
his 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). Anoth\
er is AND: If and only if signal A and signal B are both, then output a high gate (o\
n); otherwise, output a low gate (off). These are great functions for combining beat\
 triggers from different timing sources."),
quiz::Quiz("Logic","Binary or Boolean logic is a way of combining gate signals (on o\
r off voltages) to create new outputs. Each section of a logic module typically incl\
udes 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 signa\
l if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all ha\
ve gate ons). Adding an “N” to the front of a function’s name says “not” this functi\
on – in other words, a NOR function would only output a high signal if all inputs we\
re low (not input 1 nor input 2 are high)."),
quiz::Quiz("Loop","1) Effectively, any piece of music or data that repeats endlessly\
. Before digital audio and sampling, loops were created by looping tape. Today, loop\
s 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) A\
nother term for antinode, or the points of maximum displacement of motion in a vibra\
ting stretched string or a sound wave.  (See also “Standing Wave.”)"),
quiz::Quiz("Looping","Sometimes it’s useful to have a module loop or repeat its func\
tions. For example, an envelope generator that can be set to loop becomes a low freq\
uency oscillator: as it attacks to a maximum value and decays back to zero, it start\
s that attack phase again. Quite often you want a note sequencer to loop: When it re\
aches the last note in the sequence, it would be useful for it to then look back to \
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or return to the first note and start over. Audio recorders with looping features ar\
e also popular for live performance."),
quiz::Quiz("Loudness","A term referring to how the human ear perceives incoming soun\
d 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 relationshi\
p. In acoustic terms, volume is more about the amplitude of the sound waves, while l\
oudness describes how our ears hear the intensity of those waves."),
quiz::Quiz("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” o\
r “on” signal of +5 volts)."),
quiz::Quiz("Low Frequency Oscillator","This module produces repetitive, cycling wave\
s ranging in frequency from the low end of the audio spectrum to as slow as many sec\
onds or even minutes per cycle. They are used to produce effects such as tremolo (wh\
en controlling the loudness of a signal), vibrato (when controlling the pitch of a s\
ignal), repetitive filter wah-wah effects, pulse width modulation to vary the wavesh\
ape of a pulse in an oscillator, and more."),
quiz::Quiz("Low Impedance","(abbreviated Lo-Z) Described as impedance of 500 ohms or\
 less. (See also “Impedance.”)"),
quiz::Quiz("Low Pass Filter","The low pass filter (LPF) design passes harmonics belo\
w 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 cu\
toff (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 w\
eaker high harmonics – especially as a note fades to silence."),
quiz::Quiz("Low Pass Gate","By strict definition, a low pass gate (LPG) is a low pas\
s filter whose cutoff frequency goes down into the subsonic range as its control vol\
tage 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 ei\
ther 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."),
quiz::Quiz("Low-Frequency Oscillator (LFO)","A circuit that emits low-frequency elec\
tronic 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 paramet\
ers in the audio signal, such as pitch or volume. LFOs are frequently used in sample\
rs, synthesizers and signal processors to create such effects as vibrato, tremolo, a\
nd phasing."),
quiz::Quiz("low-pass-filter","An audio filter or device that attenuates signals abov\
e a certain frequency (the cut-off frequency) and passes signals with frequencies th\
at are lower than the cut-off."),
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quiz::Quiz("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 lo\
ws” in an audio signal."),
quiz::Quiz("LPF","The low pass filter (LPF) design passes harmonics below its cutoff\
 or corner frequency untouched, and reduces the level of lower harmonics depending o\
n 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 freq\
uency) are reduced in level by 12 dB; harmonics two octaves above the cutoff (four t\
imes 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 h\
armonics - especially as a note fades to silence."),
quiz::Quiz("LPG","By strict definition, a low pass gate (LPG) is a low pass filter w\
hose 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. S\
ome 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 (hi\
gher harmonics are filtered more) as it falls to silence. This mimics the way many n\
atural sounds work."),
quiz::Quiz("M2.5","A common screw thread size used to mount Eurorack modules. This s\
ize is most common when using a system of loose nuts that slide along the rails that\
 the modules are attached to."),
quiz::Quiz("M3","A common screw thread size used to mount Eurorack modules. This siz\
e is most common when using module mounting rails that have been pre-drilled."),
quiz::Quiz("Magnetic Tape","Recording tape consisting of a plastic strip coated by m\
agnetic materials, finely ground iron oxide (rust) particles. Commonly used for anal\
og recording."),
quiz::Quiz("Magnetism","A natural attractive energy of iron based-materials toward o\
ther iron-based materials."),
quiz::Quiz("MArF","The rare Buchla Model 248 MArF (Multiple Arbitrary Function Gener\
ator) is a cross between a sequencer and an envelope generator (both described elsew\
here in this glossary) in that it typically contains 16 or 32 stages (sometimes refe\
rred 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 creat\
ed. Later on, Buchla used the term MARF to describe the multi-step envelopes in inst\
ruments such as the Buchla 400."),
quiz::Quiz("Margin","See “Headroom.”"),
quiz::Quiz("Masking","The characteristic of hearing by which loud sounds prevent the\
 ear from hearing softer sounds of similar frequency. Also refers to the obscuring o\
f softer sounds by louder ones."),
quiz::Quiz("Master","1) The main output control of a console or DAW, setting the lev\
el of the mixed signal as it leaves the console. (Also called “master fader.”) 2) Th\
e final-mixed original recording from which copies are made."),
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quiz::Quiz("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."),
quiz::Quiz("Measure","The grouping of a number of beats in music. (See also “Bar.”)"\
),
quiz::Quiz("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 (1,000,0\
00 bytes, MB)."),
quiz::Quiz("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 t\
he number of beats in a measure."),
quiz::Quiz("Mic / Line Switch","Mic, Mike – Abbreviations for “microphone.”"),
quiz::Quiz("Microphone (Mic) Input","The input of a console or other device designat\
ed for a microphone signal."),
quiz::Quiz("Microphone (Mic) Level","The very low audio voltage level emitted by a s\
tudio microphone. The signal must go through a preamplifier to be increased to line \
level before entering the console. (See also “Line Level,” “Preamplifier.”)"),
quiz::Quiz("Microphone (Mic) Pad","A setting on a microphone or preamp, or a separat\
e adapter/connector, that reduces the level of the microphone signal before it enter\
s the preamplifier to prevent overload."),
quiz::Quiz("Microphone","A transducer which converts sound pressure waves into elect\
rical signals."),
quiz::Quiz("Mid-Side Miking (M/S)","(Abbreviated M/S) A stereo coincident microphone\
 placement technique in which one cardioid pattern microphone is aimed directly at t\
he sound source, and a bi-directional microphone placed sideways and as close as pos\
sible to the first mic."),
quiz::Quiz("MIDI Clock","A clock signal conveyed by MIDI that is used by the connect\
ed sequencers and musical devices to stay in sync with one another. Not to be confus\
ed with MIDI time code (MTC), MIDI clock is tied to the Beats-Per-Minute (BPM) tempo\
, advancing 24 steps per quarter note."),
quiz::Quiz("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."),
quiz::Quiz("MIDI Interface","A device that converts a MIDI signal into the digital f\
ormat of a computer so it can store and use the MIDI signal."),
quiz::Quiz("MIDI over Bluetooth","Bluetooth Low Energy (BLE) is a wireless connectio\
n specification supported by the majority of mobile computing devices. BLE (also cal\
led Bluetooth SMART) can extend battery life for mobile devices using connected acce\
ssories (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 f\
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rom different manufacturers to interoperate. The Specification for MIDI over Bluetoo\
th Low Energy (BLE-MIDI) is based on Apple's implementation which appeared in iOS8 a\
nd OSX 10.10, so that products from early adopters would remain compatible with the \
industry standard."),
quiz::Quiz("MIDI Sample Dump Standard (SDS)","A sub-protocol that was added into MID\
I to enable the transfer of digitally recorded samples between instruments, storage \
units or sound modules without converting them to analog."),
quiz::Quiz("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 serie\
s of timed steps. MIDI sequencers can exist on board MIDI controllers, keyboards or \
workstations, as standalone devices, or as computer software."),
quiz::Quiz("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 al\
ternative to “daisy chaining” devices, which can cause slight delays in the MIDI sig\
nal."),
quiz::Quiz("MIDI Thru","A port that puts out a MIDI signal that is the same as the i\
ncoming MIDI signal, effectively relaying the signal to another device without alter\
ing or changing it.  (Many MIDI devices have three MIDI ports: In, Out and Thru.)"),
quiz::Quiz("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 SMT\
PE code (usually in relation to video). (See also “SMPTE Time Code.”)"),
quiz::Quiz("MIDI","Short for Musical Instrument Digital Interface. MIDI is a common \
language to connect one synthesizer to another, and synthesizers to a computer. Alth\
ough 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 connect\
or is required: the traditional 5-pin DIN, or a USB computer-style connection."),
quiz::Quiz("Mids","Abbreviation for “mid-range frequencies,” the audio frequencies f\
rom about 250 Hz through 6000 Hz. Meant in the context of “highs, mids and lows” in \
an audio signal."),
quiz::Quiz("Mini Keys","A number of keyboard controllers and even keyboard synths us\
e a key size that is much smaller than a typical piano key. Mini keys is the term co\
mmonly used (sometimes derisively, although the space and cost savings can be quite \
significant) to refer to this hardware choice."),
quiz::Quiz("Mix Down","Mixdown or Mix Down - The processes of creating a final mix b\
y 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”\
)."),
quiz::Quiz("Mix","1) The blending of audio signals together into one composite signa\
l. 2) Can also refer to the blending of a portion of an effected audio signal back i\
nto the direct signal."),
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quiz::Quiz("Mixer","This module combines signals together. You may use a mixer to co\
mbine 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 wa\
nt linear level controls plus additional functions to invert and offset the voltages\
 going through it."),
quiz::Quiz("Modular","A modular synth breaks down the main components of a synthesiz\
er – the tone-generating oscillators, the tone-modifying filters, the amplitude-shap\
ing VCAs, and the modulation sources that create envelopes, tremolos, and more – int\
o individual modules you can purchase and install. At the most basic level, this all\
ows you to play mix-and-match in building your own custom synth."),
quiz::Quiz("Modulation Noise","Noise that is present only when the audio signal is p\
resent."),
quiz::Quiz("Modulation","When you vary a parameter of a synthesizer module using vol\
tage 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 w\
ah-wah effect, it is said that the LFO is modulating the cutoff. When an envelope ge\
nerator causes a voltage controlled amplifier (VCA) to open up to allow a sound to b\
ecome suddenly loud, and then fades it back down to silence, you can also say the en\
velope 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."),
quiz::Quiz("Modulator","We touched on the general subject of modulation and modulato\
rs in the definition above. However, quite often when someone uses the term modulato\
r, they’re usually discussing a synthesis techniques where one usually audio-rate si\
gnal “modulates” (varies) another audio signal. For example, in frequency modulation\
 (FM) synthesis, the modulator (or modulating oscillator) varies the frequency (pitc\
h) of the main signal generator (oscillator), called the carrier. In ring, balanced,\
 or amplitude modulation, the modulator is varying the loudness of the carrier signa\
l. So the term modulator is a way to make it clear which component you’re talking ab\
out in one of these patches: not the main tone generator, but the module that is dri\
ving that generator crazy."),
quiz::Quiz("Module","A self-contained group of circuits and controls. In the recordi\
ng studio, modules are often contained in interchangeable housing for installation o\
n 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 modul\
es."),
quiz::Quiz("Monaural (Mono)","(Abbreviated “Mono”) Describing an audio signal coming\
 through a single, as opposed to stereo, which is two channels. (See also “Monophoni\
c.”)"),
quiz::Quiz("Monitor Mix","A mix of the live and/or recorded audio signals that is fe\
d to the musicians so the 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 (oft\
en controlled by a separate, smaller console) and do not affect the FOH mix (in live\
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 audio) or the signal going into the multitrack recorder/DAW. In live performance se\
ttings, the monitor mix is often controlled by a separate audio engineer running a s\
eparate sound board."),
quiz::Quiz("Monitor Mixer Section","Monitor Section/Monitor Mixer Section – The sect\
ion of the console that is used to create a rough mix so the engineer can hear what \
is being recorded without effecting the levels being fed to the multitrack recorder \
or DAW."),
quiz::Quiz("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.”)"),
quiz::Quiz("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 t\
o 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 performe\
rs to hear a mix of the tracks while they are performing."),
quiz::Quiz("Monophonic","(Abbreviated “Mono”) 1) A single sound source or single-cha\
nnel 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 play\
s one pitch (or “voice”) at a time. (See also “Voice.”)"),
quiz::Quiz("Morphing","In the context of a modular synth, morphing refers to an osci\
llator 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 digit\
al oscillators which internally crossfade (or in some cases, switch) from one wavesh\
ape to another, although it is sometimes applied to analog oscillators that give you\
 real time control over waveshapes."),
quiz::Quiz("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-sty\
le transistor ladder filter that can be low pass or high pass, and one AD or AR enve\
lope generator. It also has a very capable step sequencer plus a miniature one-octav\
e 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 exp\
and a basic modular system. I have an online introductory course to the Mother-32 co\
ming out this spring, and will have a course plus ongoing weekly series on adding di\
fferent modules to this starter system."),
quiz::Quiz("Moving Coil Microphone","A microphone in which sound pressure waves are \
converted to an electrical audio signal by an induction coil moving within a magneti\
c field—a process often compared to a loudspeaker working in reverse. Dynamic microp\
hones are less sensitive than condenser microphones, but can be effective for miking\
 louder sound sources or for close-miking applications."),
quiz::Quiz("Moving Fader Automation","A feature in some consoles in which fader chan\
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ges can be pre-programmed to occur automatically during playback of a multitrack rec\
ording."),
quiz::Quiz("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 fo\
llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
d is both historical and physically large, some users “5U” as a badge of honor that \
they’re traditional and cool. (And the are.) There was also a briefly popular 5U for\
mat from MOTM that used a different width and power connection. It has since been di\
scontinued, but there are still diehard MOTM format users today."),
quiz::Quiz("Multi-Tap Delay","A delay works by in essence putting audio in one end o\
f 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 wh\
en 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."),
quiz::Quiz("Multimeter","A small device that tests electrical voltage, current, and \
resistance. Multimeters are useful in recording studios for calibrating electrical s\
ystems and troubleshooting problems."),
quiz::Quiz("Multiple Arbitrary Function Generator","The rare Buchla Model 248 MArF (\
Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelop\
e generator (both described elsewhere in this glossary) in that it typically contain\
s 16 or 32 stages (sometimes referred to as “segments”), and a rate control to inter\
polate between these stages. This means very complex envelope shapes and other contr\
ol 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."),
quiz::Quiz("Multiple","Quite often you need to split or copy a signal to send to mor\
e than one destination. This is commonly done with a multiple (“mult” for short) whe\
re you plug one source in, and then plug in additional patch cables to go off to mul\
tiple destinations."),
quiz::Quiz("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 possiblity of one input being switched between mult\
iple outputs. A sequential switch is a type of multiplexor, as it chooses among mult\
iple inputs to decide which one to send to the output (or the other way around). The\
re are some modules that do this at audio rate, using an oscillator's output to swit\
ch between variations of another waveshape to create complex, chopped mixtures of th\
ose waveforms."),
quiz::Quiz("Multitimbral","Refers to the ability of a synthesizer or module to play \
several different sounds, patches or “timbres” at once."),
quiz::Quiz("Multitrack Recording","Also called tracking or multitracking) The heartb\
eat of the recording studio, multitrack recording is process of recording a collecti\
ve of sound sources onto separate tracks, each with its own audio channel, then comb\
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ining 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 o\
nto multitrack analog tape, but today they can also be recorded digitally as separat\
e audio files into a digital audio workstation (DAW)."),
quiz::Quiz("Multitrack Tape","A piece/reel of magnetic tape which can be used to sto\
re two or more discrete signals in sync with each other."),
quiz::Quiz("Musical Instrument Digital Interface (MIDI)","Short for Musical Instrume\
nt Digital Interface. MIDI is a common language to connect one synthesizer to anothe\
r, 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 t\
o watch out for is what type of connector is required: the traditional 5-pin DIN, or\
 a USB computer-style connection."),
quiz::Quiz("Mute Switch","A switch on a console or other piece of audio equipment th\
at 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 ca\
n cut the signal for that channel."),
quiz::Quiz("Mute","Sometimes you need to silence or disconnect a signal. A circuit t\
hat allows you to do so is called a mute."),
quiz::Quiz("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 us\
ed instead. Nanowebers per meter of tape effectively describes the signal strength t\
hat is being recorded to tape."),
quiz::Quiz("Narrowband Noise","Noise (random energy) that occurs over a limited freq\
uency range."),
quiz::Quiz("Near Field","The area between 1-5 feet from the sound source. Studio mon\
itors are generally considered “near-field” speakers because they are meant to be li\
stened to at close range. (See also “Far Field.”)"),
quiz::Quiz("Near-Coincident Miking","A stereo miking technique in which two micropho\
nes are placed near each other at an outward angle to create a stereo image (as oppo\
sed 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).\
"),
quiz::Quiz("Negative Feedback","A portion of the output signal that is fed back to t\
he input of an amplifier with its phase inverted from the original output signal. Th\
is has a dampening effect on the output, effectively cancelling out a portion of the\
 volume."),
quiz::Quiz("Noise Floor","The level of the noise present below the audio signal, mea\
sured in dB. Every electronic device emits a minimum level of noise, even when no au\
dio is traveling through it; this is described as its noise floor. Generally speakin\
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g, 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 record\
ing is the sum of all the noise generated by connected devices. The objective is alw\
ays to keep the noise floor as low as possible."),
quiz::Quiz("Noise Gate","A gate that is used reduce audible noise by automatically t\
urning off an audio channel when the signal is not present."),
quiz::Quiz("Noise Reduction","Any of a number of processes to remove noise from a si\
gnal, device or system."),
quiz::Quiz("Noise","Describes any unpleasant, objectionable or unintended sound freq\
uencies 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 ot\
herwise silent passages. (See also “Noise Floor.”) Bad connections, improper groundi\
ng, radio interference and other issues can also cause introduce noise into the sign\
al. Engineers may also deliberately run a noise signal through a sound system for te\
sting purposes. (See also “White Noise, “Pink Noise.”)"),
quiz::Quiz("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 ve\
rsion at any time. (Sometimes also called “Nonlinear editing.”)"),
quiz::Quiz("Nondirectional","In microphones, picking up evenly from all directions."\
),
quiz::Quiz("Normalize","To apply a fixed amount of gain to audio so that the highest\
 peak is set at the highest acceptable recording level."),
quiz::Quiz("Normalled","The power of modular synthesizers is that you can patch a si\
gnal to flow the way you prefer through your system. This can also be a time-consumi\
ng 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 conn\
ections 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 con\
nections are often referred to as being normalled. For example: An internal noise so\
urce may normally be connected to one channel of a mixer that appears before the fil\
ter, but if you insert a patch cable into a jack usually labeled external input, thi\
s “normalled” connection is broken and replaced by your external connection."),
quiz::Quiz("Notch Filter","This is a particular type of filter mode where audio freq\
uencies 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, whic\
h only passes harmonics around the cutoff frequency. Notch filters tend to have a su\
btle effect on the sound; moving (modulating) the cutoff frequency can result in a w\
eak phasing sort of sound. Notch filters are often used in sound systems to weaken o\
r 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 fil\
ter in a patch to hollow out a sound, leaving room in the harmonic spectrum for othe\
r sounds to exist with less competition, or just to create a sound more likely to ca\
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tch the ear because something that is expected is instead missing."),
quiz::Quiz("Notch","A narrow band of audio frequencies."),
quiz::Quiz("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 t\
o tape."),
quiz::Quiz("Nybble","Nybble (or Nibble) – One half byte of computer data, or 4 bits.\
"),
quiz::Quiz("Nyquist Frequency","In digital recording, the highest frequency that can\
 be recorded and reproduced properly, equivalent to a one-half the sampling rate. (F\
or example, with the common sampling rate of 44,100 kHz per second, the Nyquist freq\
uency would be 22,050 kHz.) Aliasing begins to occur with frequencies that exceed th\
is threshold. (See also “Aliasing.”)"),
quiz::Quiz("Nyquist Rate","he lowest sampling rate that can be used to record and re\
produce a given audio signal, equivalent to twice the highest frequency. If the high\
est frequency found in an analog signal or sound is 18,000 kHz, theoretically the si\
gnal 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 inver\
se principle of the Nyquist Frequency. (NOTE: the sample rate of 44,100 kHz/second i\
s considered the standard sample rate because it easily covers the upper range of hu\
man hearing, which is about 20,000 kHz.)"),
quiz::Quiz("Octave Divider","A module that creates a new tone one or two octaves bel\
ow the fundamental harmonic – the “pitch” – of the sound coming into it, to emphasiz\
e the bass. Sometimes also known as a suboctave or sub bass function."),
quiz::Quiz("Octave","An octave is a typical musical internal. For example, all of th\
e “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 su\
btracting 1 volt to get a one octave change in pitch; some oscillators also have oct\
ave 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 oc\
taves). Suboctave or subharmonic generators divide the input pitch by 2 or 4 to crea\
te new waveforms that are one or two octaves lower in pitch, which adds bass."),
quiz::Quiz("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-a\
xis.)"),
quiz::Quiz("Offset Time","1) The SMPTE time that will trigger a MIDI sequencer to be\
gin. 2) The amount of position difference needed to get two reels to play the music \
in time."),
quiz::Quiz("Offset","In simple terms, Offset modules usually add or subtract a volta\
ge from a signal passing through - such as shifting a 0 to +10v signal to instead va\
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ry between -5 and +5 volts."),
quiz::Quiz("Ohm's Law","The mathematical relationship between voltage, current and r\
esistance."),
quiz::Quiz("Ohm","The unit used to measure the amount of opposition (impedance) to e\
lectrical current flow in a signal or device. (See also “Impedance.”)"),
quiz::Quiz("Omni Mode","A setting that enables a MIDI device to recognize and respon\
d to all MIDI channels at once."),
quiz::Quiz("Omni","A prefix meaning “all.”"),
quiz::Quiz("Omnidirectional Pattern","In microphones, picking up evenly from all dir\
ections (sometimes also called “Nondirectional”). 2) In speakers, sending out the si\
gnal evenly in all directions."),
quiz::Quiz("On Axis","The position directly in front of the diaphragm of a microphon\
e, in line with its movement."),
quiz::Quiz("Open Circuit","An electrical circuit that is disconnected, interrupted o\
r incomplete, preventing the flow of electricity."),
quiz::Quiz("Operating Level","(Sometimes called “Reference Level“) The maximum level\
 that should not be exceeded in normal operation."),
quiz::Quiz("Operational Amplifier","(Abbreviated “Op Amp“) An amplifying circuit use\
d in most audio and electronic devices."),
quiz::Quiz("Operational Transconductance Amplifier","An OTA (operational transconduc\
tance 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, yo\
u 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 voltag\
e to resistance device that makes it possible to add voltage control to circuits suc\
h as filters. In general, when someone touts they have an OTA based filter, they usu\
ally mean it has a “warm” sound...unless it’s an MS-20 filter clone, in which case i\
t’s thinner and more edgy. In reality, using an OTA is more about convenience of des\
ign than creating a specific sound."),
quiz::Quiz("Operator","There are a few different synthesis techniques where one usua\
lly audio-rate signal does something to another audio signal. For example, in freque\
ncy modulation (FM), a second signal (called the modulator) varies the frequency (pi\
tch) of the main signal, called the carrier. These two signals or oscillators are of\
ten 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."),
quiz::Quiz("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 inve\
rted output: it would only be on (high) if all inputs were low (not input 1 nor inpu\
t 2 are high). An XOR (Exclusive OR) would only output a high signal if one of the i\
nputs was high, but not if both inputs were high (or low). Finally, an XNOR is the i\
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nvert of an XOR function."),
quiz::Quiz("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 oscill\
ator because its output varies up and down (oscillates) in voltage in a repeating pa\
ttern. This pattern is referred to as its waveshape (such as a square wave, that alt\
ernates 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 t\
hat vibrates back and forth on a guitar, a drum head that vibrates up and down, or t\
he vibrations in the reed of a woodwind instrument. The vibrations of a modular synt\
h’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.)"),
quiz::Quiz("Oscilloscope","This is a piece of test equipment that displays voltage f\
luctuations as graphical waveforms. A ’scope can run at a wide range of frequencies,\
 displaying slowly changing voltages like LFOs or envelopes, or quickly changing vol\
tages 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 jo\
b to your computer, or scopes as modules."),
quiz::Quiz("OTA","An OTA (operational transconductance amplifier) circuit is one tha\
t 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, an\
d is also known for a “warm” sound. Therefore, you may find it in VCAs (voltage cont\
rolled 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 i\
t possible to add voltage control to circuits such as filters. In general, when some\
one touts they have an OTA based filter, they usually mean it has a “warm” sound...u\
nless it’s an MS-20 filter clone, in which case it’s thinner and more edgy. In reali\
ty, using an OTA is more about convenience of design than creating a specific sound.\
"),
quiz::Quiz("Out of Phase","1) Being similar to another signal in amplitude, frequenc\
y and wave shape but being offset in time by part of a cycle. 2) Having the opposite\
 polarity."),
quiz::Quiz("Outboard Equipment","Equipment that is used with, but is not a part of, \
a console."),
quiz::Quiz("Output Impedance","The opposition to the flow of electrical current by t\
he output circuits of an amplifier (or other device)."),
quiz::Quiz("Output Level","The signal level at the output of a device."),
quiz::Quiz("Output","1) The jack or physical location of where a device sends out a \
signal. 2) The signal put out by a device."),
quiz::Quiz("Overdubbing","The process of recording an additional musical performance\
 over an existing recording, usually on its own track. Overdubbing has become a comm\
on recording technique with the advent of multitrack recording, first on multitrack \
analog tape, and more recently via computers and Digital Audio Workstations (DAWs)."\
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),
quiz::Quiz("Overtone","Any harmonic in a tone except the fundamental frequency. (See\
 also “Partial.”)"),
quiz::Quiz("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 somet\
imes 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."),
quiz::Quiz("Pan (Panning)","The process of “placing” a particular sound within the s\
tereo field. This is accomplished by controlling the balance of the signal between t\
he left and right speakers so the ear hears the sound as coming from a particular po\
int 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."),
quiz::Quiz("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 ster\
eo (or surround) space by controlling how much of the signal is sent to each speaker\
 or channel."),
quiz::Quiz("Parallel Jacks","Several jacks that are wired so that each connection is\
 wired to the corresponding connection of other jacks."),
quiz::Quiz("Parallel Port","A connector that is able to transmit and receive digital\
 data at the same time though different pins."),
quiz::Quiz("Parameter","Parameter is the fancy name given to any value or property o\
r control of a synthesizer module that you’re trying to change. For example, an osci\
llator’s parameters typically include its pitch and the width of its pulse wave. A f\
ilter’s parameter will include its cutoff frequency (pitch), the amount of resonance\
 (feedback), and possibly other controls such as a blend between its different outpu\
ts. Parameter was a popular term to describe a value you could change in software, a\
nd it’s been carried over by some to hardware modular synths."),
quiz::Quiz("Parametric Equalization","An equalizer in which all parameters of equali\
zation can be adjusted to any amount, including the center frequency, the amount of \
boost or cut, and the bandwidth."),
quiz::Quiz("Paraphonic","A paraphonic synth is one where all of the notes being play\
ed go through a single filter (VCF) and amplifier (VCA). This was a popular scheme i\
n the early days of polyphonic synths in that a separate oscillator (or organ-like f\
requency divider, in the case of “string synths” and the such) was used for each not\
e played, but they were mixed before all going to the filter and amp to articulate t\
he note(s). It was not uncommon for some monophonic synths to allow two to four inde\
pendent notes to independently control the pitch of its oscillators, while still goi\
ng 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 n\
otes will be re-articulated together."),
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quiz::Quiz("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."),
quiz::Quiz("Pass Band","The frequency range of signals that will be “passed” by a fi\
lter, rather than reduced."),
quiz::Quiz("Passive Device","A component that does not generate or control electrica\
l current (as opposed to an “Active Device”). In audio applications, this usually re\
fers to a piece of gear that does not include an amplifier as part of its design. Fo\
r example, active speakers are self-powered, while passive speakers require an exter\
nal amplifier in order to reproduce sound. (See also “Active Device.”)"),
quiz::Quiz("Passive","Means no active (i.e. connected to a power supply) electronics\
 are involved - such as sending a signal straight through a potentiometer control, i\
nstead of using op amps and other electronics to create a mixer circuit around it. P\
assive is cheap and easy, and does not add noise to a signal. But passive electronic\
s cannot buffer one signal from another (meaning they might interact in undesirable \
ways), and cannot boost, offset, or invert a signal."),
quiz::Quiz("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 an\
d outputs of the console and components in the studio, used for the purpose of organ\
izing, managing and regulating signal flow."),
quiz::Quiz("Patch Cable","The cables used to connect together the different inputs a\
nd 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 physical\
ly connect two callers together using electrical cables. As different modular format\
s often use different connector standards, you need to make sure the connectors at t\
he 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)."),
quiz::Quiz("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 us\
ed to make connections in the patch bay (hence the name); however, patch cords facil\
itate 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."),
quiz::Quiz("Patch Field","A panel or component containing a series of jacks with con\
nections for most of the inputs and outputs of the console and components in the stu\
dio, used for the purpose of organizing, managing and regulating signal flow."),
quiz::Quiz("Patch Librarian","A computer program allowing for the storing of sound p\
atches outside of a synthesizer via MIDI."),
quiz::Quiz("Patch Panel","A panel or component containing a series of jacks with con\
nections for most of the inputs and outputs of the console and components in the stu\
dio, used for the purpose of organizing, managing and regulating signal flow."),
quiz::Quiz("Patch","The shorthand term used to refer how a series of modules are int\
erconnected to create a sound, derived from the fact that patch cords are used to co\
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nnect the modules together. 1) To route or reroute the signal in an audio system (su\
ch as a console) by using short cables with plugs inserted into jacks. 2) A sound se\
tting or program on a synthesizer."),
quiz::Quiz("Path","Short for Signal Path, the way in which current does or may trave\
l in a circuit or through a device."),
quiz::Quiz("PCM","Pulse Code Modulation - A process by which analog signals are tran\
slated to digital code. This is done by taking samples of the amplitude of the analo\
g signal at regular rapid intervals, then translating it into binary numbers as a di\
gital 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 a\
udio."),
quiz::Quiz("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 di\
fferences to avoid problems with the patent used by Yamaha's FM synths of the era. I\
ntriguingly, it did a good job at mimicking many "analog" synth effects including th\
e sound of a resonant filter."),
quiz::Quiz("Peak Filter","An EQ circuit/filter that boosts or cuts the middle (cente\
r frequencies in an audio signal, as opposed to high-pass or low-pass filters. (NOT \
to be confused with amplitude peaks.)"),
quiz::Quiz("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 Mete\
r.")"),
quiz::Quiz("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 wav\
e. (See also "Peak Value.")"),
quiz::Quiz("Peak Value","eak Value (also called Peak Level) – The measure of the max\
imum positive or negative value (amplitude) of a waveform at any moment. In audio, t\
his is visually depicted as the farthest point of the waveform above or below the ze\
ro axis."),
quiz::Quiz("Pedal Board","A board with several guitar pedals attached and inter-conn\
ected so that a guitar player can conveniently activate a number of different effect\
s."),
quiz::Quiz("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."),
quiz::Quiz("Phase Addition","The increased audio energy that happens when waveforms \
are in similar phase relationships, resulting in an increase in volume up to twice w\
hat it should be."),
quiz::Quiz("Phase Cancellation","The opposite of phase addition, this is the reducti\
on of energy that occurs when two similar waveforms that are out of phase with one a\
nother and begin cancelling each other out, either greatly reducing or eliminating t\
he 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."),
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quiz::Quiz("Phase Distortion Synthesis","Phase Distortion synthesis was used by Casi\
o originally in the 80s in the CZ line of synths. It is related to FM (frequency mod\
ulation), 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” sy\
nth effects including the sound of a resonant filter."),
quiz::Quiz("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 ad\
ding harmonics to the sound."),
quiz::Quiz("Phase Lock","Any of a number of processes used to help synchronize signa\
ls or devices by correcting phase differences. For example, in analog tape machines,\
 phase locking helps to keep multiple machines synced together by sensing phase diff\
erences in the playback of pilot tunes by the two machines and adjusting the speed t\
o eliminate the phase difference. In synthesizers, phase locking controls one tone g\
enerator so that it begins its waveform in phase with the signal from another tone g\
enerator. Phase-locked loops (PLL) are reference signals used in the clock functions\
 of electronic devices."),
quiz::Quiz("Phase Locked Loop","A phase locked loop is, in essence, an oscillator th\
at 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 tha\
t 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 incom\
ing pitch - perhaps to drive a special circuit such as a switched-capacitor filter."\
),
quiz::Quiz("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 ba\
ck 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 wavefor\
m. 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."),
quiz::Quiz("Phase Reversal","A change in a circuit to get the waveform to shift by 1\
80 degrees."),
quiz::Quiz("Phase Shift","A delay introduced into an audio signal measured in degree\
s delayed."),
quiz::Quiz("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 harmo\
nics 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, an\
d different harmonic components of the original sound cancel each other out (see Pha\
se), 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."),
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quiz::Quiz("Phase-Locked Loop","PLL -  Any of a number of processes used to help syn\
chronize signals or devices by correcting phase differences. For example, in analog \
tape machines, phase locking helps to keep multiple machines synced together by sens\
ing phase differences in the playback of pilot tunes by the two machines and adjusti\
ng the speed to eliminate the phase difference. In synthesizers, phase locking contr\
ols 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 c\
lock functions of electronic devices."),
quiz::Quiz("Phase","A measurement (expressed in degrees) of the time difference betw\
een 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) c\
an be defined by the phase. For example, if you are one-quarter of the way through a\
 waveform’s cycle, your phase is 90°."),
quiz::Quiz("Phasing","An effects sound created by varying the phase shift of an audi\
o signal, then mixing it with the direct signal."),
quiz::Quiz("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 i\
t is perceived to be as loud as a 1000-Hz tone playing at 60dB."),
quiz::Quiz("Phone Plug","A plug (or its mating jack) with a diameter of 1/4 inch and\
 a length of I 1/4 inches used for interconnecting audio."),
quiz::Quiz("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."),
quiz::Quiz("Physical Modeling","One approach to (often digital) synthesis is to recr\
eate 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 t\
o create sounds. There are a handful of modules available which perform this modelin\
g to create their sounds."),
quiz::Quiz("Pickup Pattern","The shape of the area in front of or around the microph\
one from where it evenly picks up sound. Many use this term interchangeably with “po\
lar pattern,” but a polar pattern gives more detail about microphone sensitivity. (S\
ee also “Polar Pattern.)"),
quiz::Quiz("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 “Conta\
ct Microphone.”"),
quiz::Quiz("Pinch Roller","A rubber (or plastic) wheel on a tape recorder that pinch\
es the tape between it and the capstan, allowing the capstan to pull the tape."),
quiz::Quiz("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 an\
alog by playing the tracks through the console and recording them onto separate trac\
ks, or digitally through a digital audio workstation. Bouncing was once used frequen\
tly by engineers to free up additional tracks for recording, but in digital workstat\
ions where tracks are virtually unlimited, this practice is basically obsolete. Toda\
y, engineers typically bounce tracks for the purpose of creating a preliminary or fi\
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nal mix of a song."),
quiz::Quiz("Pink Noise","A noise signal similar to white noise, containing all audib\
le 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 o\
ctave. As each higher octave has double the frequency of the octave below it which s\
preads out the energy over a wider range of frequencies, pink noise tends have a mor\
e 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) a\
nd therefore tends to sound very bright."),
quiz::Quiz("Pitch Bend","A mechanism on a synth, keyboard or controller that can cau\
se the pitch of the note to move up or down by a small amount."),
quiz::Quiz("Pitch to Voltage Converter","A device that detects the frequency of an a\
udio waveform and changes it into a control voltage, which is in turn fed to an osci\
llator that produces a pitch at the same frequency."),
quiz::Quiz("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-Convert\
er.”)"),
quiz::Quiz("Pitch-to-Voltage Converter","A device that detects the frequency of an a\
udio waveform and changes it into a control voltage, which is in turn fed to an osci\
llator that produces a pitch at the same frequency."),
quiz::Quiz("pitch","1) The perception of frequency by the ear (a higher or lower ton\
e 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."),
quiz::Quiz("Plate Reverb","A device that produces artificial reverberation by sendin\
g vibrations across a metal plate via a transducer similar to a speaker driver. Phys\
ical plate reverbs today are considered a vintage form of artificial reverb; nowaday\
s, most plate reverb effects are emulated digitally by plugins or reverb units."),
quiz::Quiz("Playback Head","A transducer that converts magnetic flux recorded on tap\
e into an audio signal for playback."),
quiz::Quiz("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."),
quiz::Quiz("Playback","1) The reproduction of recorded audio. 2) In motion picture o\
r video production, the reproduction of the music over loudspeakers so the performer\
s/musicians can perform in time to the music for the camera."),
quiz::Quiz("Playlist","1) See “Take.” 2) A user-defined selection of songs; a featur\
e available on most streaming and digital media players."),
quiz::Quiz("PLL","A phase locked loop is, in essence, an oscillator that tries to ma\
tch 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 h\
igher than the incoming reference signal – such as a timing module that can create a\
n output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a v\
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ery high frequency oscillator that is locked to a multiple of an incoming pitch - pe\
rhaps to drive a special circuit such as a switched-capacitor filter."),
quiz::Quiz("Plug","A connector, usually on a cable, that mates with a jack."),
quiz::Quiz("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 i\
n dB of sound arriving from different directions. Similar to “Pickup pattern,” but m\
ore specific. 2) In speakers, a graphic display of the speaker’s dispersion of sound\
."),
quiz::Quiz("Polarity","The direction of current flow or magnetizing force."),
quiz::Quiz("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 pol\
arizer, as it changes the polarity (+ versus -) of a signal. A control voltage inver\
ter is often combined with an offset voltage to adjust the output voltage into the d\
esired 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. Sinc\
e some modules such as voltage controlled amplifiers usually expect only positive vo\
ltages, you would then need to add 8 volts to that result to get an upside-down (inv\
erted) envelope that still had an overall range of 0 to +8v."),
quiz::Quiz("Polarizing Voltage","In condenser and electret microphones, the introduc\
tion of a small amount of electrical current to create the magnetism by which the ca\
pacitor converts audio signals to electrical current. In condenser microphones, pola\
rizing voltage is provided externally (see also “Phantom Power”); in electret microp\
hones, the polarizing voltage is permanently impressed on the condenser during manuf\
acturing."),
quiz::Quiz("Pole Pieces","Iron or other magnetic material that conducts magnetic for\
ce for use in transducers like record heads, playback heads, microphones, speakers, \
etc."),
quiz::Quiz("Pole","This is a technical term that helps describe the design of a filt\
er. Each pole of a filter attenuates frequencies beyond its cutoff or corner frequen\
cy 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 cuto\
ff frequency by 24 dB; frequencies two octaves above the cutoff are attenuated by 48\
 dB and so forth."),
quiz::Quiz("Polyphonic","The term \"polyphonic\" refers to a synthesizer that can pl\
ay 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 t\
he same time. A term commonly used to describe synths and keyboards. (See also “Voic\
e.”)"),
quiz::Quiz("Ponging (Bouncing)","The technique of combining and mixing multiple trac\
ks 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, o\
r digitally through a digital audio workstation. Bouncing was once used frequently b\
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y engineers to free up additional tracks for recording, but in digital workstations \
where tracks are virtually unlimited, this practice is basically obsolete. Today, en\
gineers typically bounce tracks for the purpose of creating a preliminary or final m\
ix of a song."),
quiz::Quiz("Pop Filter","A device that is placed over a microphone or between the mi\
crophone and vocalist to prevent loud “pop” sounds created by the vocalist’s breath \
directed toward the microphone."),
quiz::Quiz("Port","1) A connection point in computer or electronic device for transm\
itting and receiving digital data, similarly to how a jack receives and transmits au\
dio signals. 2) An opening or vent in a speaker case that resonates with air movemen\
t in the speaker, used in bass reflex speakers and woofers to enhance low frequencie\
s."),
quiz::Quiz("Portamento","A pitch change that smoothly glides from one pitch to anoth\
er. Also refers to the synthesizer mode or MIDI command that allows or causes this t\
o happen."),
quiz::Quiz("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 effec\
ts.  NOTE: We mention film in this context because film post-production includes a l\
ot of audio work (e.g., voiceovers, foley, audio mixing and editing) to the point th\
at many audio engineers are involved in film post-production as a full-time career."\
),
quiz::Quiz("Post Roll","A segment of blank tape (or track silence, on a DAW) that ru\
ns past the end of the recording. (See also “Pre-Roll.”)"),
quiz::Quiz("Post-Fader","Refers to an aux send position or setting that places the s\
end after the channel fader within the signal path. Sending a signal post-fader mean\
s the fader itself affects the level of the send signal, as opposed to pre-fader. (S\
ee also Pre-Fader.)"),
quiz::Quiz("Post","Refers to an aux send position or setting that places the send af\
ter 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 als\
o Pre-Fader.)"),
quiz::Quiz("Pot","Often thought of as a fancy word for “knob,” a potentiometer is ba\
sically 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 t\
o an aux send or bus. Potentiometers can be knobs or faders, meaning that almost eve\
ry control on a console that isn’t a button or switch is a potentiometer. However, m\
any engineers commonly refer to faders as “faders” and knobs as “pots.”"),
quiz::Quiz("Potentiometer","(Abbreviated “Pot“) Often thought of as a fancy word for\
 “knob,” a potentiometer is basically any mechanism that controls input or output vo\
ltage 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 f\
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aders, 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.”"),
quiz::Quiz("Power Amplifier","(abbreviated “Power Amp”) A device that amplifies a li\
ne level signal to drive a speaker or set of speakers.  (See also “Line Level.”)"),
quiz::Quiz("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 con\
nectors that go to your individual modules."),
quiz::Quiz("PPQN","When you send a clock signal (usually a gate signal or other elec\
trical 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 t\
erms of how many pulses there are per quarter note – PPQ or PPQN for short. If the c\
lock is just happening every quarter note, then the clock speed is 1 PPQN; in the ca\
se of DIN Sync (a popular standard among early Roland synths, with DIN being the typ\
e 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)."),
quiz::Quiz("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 (S\
ee also “Pre-Fader,” “Post-Fader.”)"),
quiz::Quiz("Pre Emphasis","A boosting of high frequencies during the recording proce\
ss to keep the audible signal above the noise floor."),
quiz::Quiz("Pre Fader","Refers to an aux send position or setting that places the se\
nd 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."),
quiz::Quiz("Pre-Delay","A parameter on a reverb unit or plugin that determines the a\
mount of time (delay) between the original dry sound and the early reflections of re\
verberation. This feature is often used to simulate the natural acoustic properties \
of a room, but can also be used to create interesting unnatural effects."),
quiz::Quiz("Pre-Echo","(Also called “Forward Echo”) A compression artifact that ofte\
n occurs in digital audio in which an “echo” of a sound (or part of a sound) is hear\
d ahead of the sound itself, often due to the data inconsistencies in certain compre\
ssed 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 g\
enerally an unwanted problem that requires additional signal processing to resolve—b\
ut in some cases it can also be used on purpose as a sound effect (not to be confuse\
d with “Reverse Echo”)."),
quiz::Quiz("Pre-Fade Listen (PFL)","A function on the channel strip of a mixer or DA\
W that allows a channel signal to be heard and often metered before the channel fade\
r."),
quiz::Quiz("Preamplifier (Preamp)","A low-noise amplifier designed to take a low-lev\
el signal (for example, from a microphone) and bring it up to normal line level befo\
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re sending it into the mixing console."),
quiz::Quiz("Precedence Effect (Haas Effect)","Simply stated, a factor in human heari\
ng in which we perceive the source of a sound by its timing rather than its sound le\
vel. 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 refl\
ection or reproduction from another source. The reflection of the sound must be at l\
east 10dB louder than the original source, or delayed by more than 30ms (where we ca\
n 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 con\
fused by reflections and reverberations off of nearby surfaces. Understanding the Ha\
as 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."),
quiz::Quiz("Precision Adder","Synthesizers are very sensitive to unintentional varia\
tions in pitch control voltage - any error can result in the oscillators under contr\
ol going out of tune. Therefore, whenever you add together pitch control voltages in\
side a modular synth, you really should be using a precision adder that precisely ad\
ds together the pitch voltages without introducing an error. Ordinary mixers might s\
lightly attenuate or amplify a voltage passed through them, which in most cases woul\
d create tuning errors."),
quiz::Quiz("Premix","1) The process of mixing a set of tracks as group, then managin\
g 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 n\
ot necessarily pre-recorded. (See also “Bouncing.”) 2) An important part of film pos\
t-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."),
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.")\
,
quiz::Quiz("Presence","1) In amplification and mixing, the boosting of upper-mid fre\
quencies 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.”\
"),
quiz::Quiz("Preset","A factory programmed set of parameters on a synth, signal proce\
ssor, plug-in or other electronic device."),
quiz::Quiz("Pressure Microphone","(Also called “pressure operative microphone”) - A \
microphone whose diaphragm responds to incoming sound wave pressure as it works agai\
nst the normal or controlled air pressure inside the microphone case. This design ma\
kes the diaphragm sensitive to pressure regardless of direction, giving it an omnidi\
rectional pickup pattern. (See also “Omnidirectional Pattern.”)"),
quiz::Quiz("Pressure Sensitivity (Aftertouch)","A feature in some keyboard instrumen\
ts by which applying additional pressure to a key after it has been pressed can acti\
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vate an additional MIDI control command. a synthesizer or Keyboard Controller of Aft\
er Touch (a control or operational function of a synthesizer where pressing a key af\
ter it has been pressed, and before it is released, will activate a control command \
that can be set by the player)."),
quiz::Quiz("Pressure Zone Microphone (Boundary Microphone)","An omnidirectional micr\
ophone 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 tradem\
ark Pressure Zone Microphone (PZM)."),
quiz::Quiz("Pressure-Gradient Microphone","(Also called “Velocity Microphone“) A mic\
rophone whose diaphragm is exposed front and back, with diaphragm movement being cau\
sed by the pressure difference between its front and back. This creates a bi-directi\
onal or “figure-8” pickup pattern (See also “Bi-Directional Pattern.”)"),
quiz::Quiz("Pressure","Some keyboards measure how hard you press down on the keys, a\
nd convert this to a voltage (or other control signal such as MIDI, which can then b\
e converted into a control voltage) that you can use to add expression to a note, su\
ch 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 r\
ather than pressure or force - so you can increase or decrease the aftertouch amount\
 by rolling between the tip and length of your finger."),
quiz::Quiz("Print Through","The unwanted transfer of magnetic flux from one layer of\
 analog tape to another."),
quiz::Quiz("Pro Tools","Avid’s trade name for its digital audio workstation (DAW) th\
at has become an industry standard in professional recording studios."),
quiz::Quiz("Producer","In music, the producer is the director of an audio recording \
project; the person responsible for getting a final product of desired quality withi\
n a budget."),
quiz::Quiz("Production Studio","Broadly speaking, any space dedicated to production \
within the arts, for example, film/video, animation or post production. In the conte\
xt 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."),
quiz::Quiz("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."),
quiz::Quiz("Program Change","A MIDI message that tells the receiving device to chang\
e presets."),
quiz::Quiz("Programmable","Able to have the parameters changed by the user, especial\
ly in a computer controlled device."),
quiz::Quiz("Prompt","A set of instructions for the user to follow, which appears on \
a computer screen."),
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quiz::Quiz("Protocol","In digital and information technology, a set of rules governi\
ng the structuring and transmitting of data in a standardized format so all related \
devices can properly interpret the data."),
quiz::Quiz("Proximity Effect","The natural boost in the microphone’s output for bass\
 frequencies as the mic is placed closer to the sound source."),
quiz::Quiz("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 p\
sychological and emotional factors into account. This branch of science is helpful t\
o audio engineers in understanding how the brain interprets various sounds and frequ\
encies."),
quiz::Quiz("Puck","Any circular piece of metal, fiber, rubber, etc., which drives so\
mething from a rotating power source. A common example in the recording studio is th\
e puck in a rotating Leslie speaker."),
quiz::Quiz("Pulse Code Modulation (PCM)","A process by which analog signals are tran\
slated to digital code. This is done by taking samples of the amplitude of the analo\
g signal at regular rapid intervals, then translating it into binary numbers as a di\
gital 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 a\
udio."),
quiz::Quiz("Pulse Per Quarter Note","When you send a clock signal (usually a gate si\
gnal or other electrical pulse) around a modular synth to move sequencers through th\
eir steps and the such, it’s good to know how fast that clock is pulsing. This is us\
ually defined in terms of how many pulses there are per quarter note – PPQ or PPQN f\
or 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)."\
),
quiz::Quiz("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 port\
ion narrower and vice versa). The act of varying the width of the resulting pulse wa\
ve 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 re\
ferred 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, mor\
e lush feel, or of giving a digital sound more analog properties."),
quiz::Quiz("Pulse","Pulse has a couple of different meanings in a modular synth. Whe\
n you alter the shape of a square wave so that one portion is narrower than the othe\
r, it is referred to as 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 refer\
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red to as a pulse. 1) The steady beat in music based on its tempo, whether audible o\
r perceived. 2) A type of sound wave commonly created and manipulated by synthesizer\
s whose waveform is characterized by sharp rises and drops in amplitude like a squar\
e wave, but whose peaks are shorter than its troughs, giving the wave a pulse-like f\
eel. Also called “Pulse Wave.”"),
quiz::Quiz("Pumping and Breathing","In studio jargon, an effect created when a compr\
essor is rapidly compressing and releasing the sound, creating audible changes in th\
e signal level. “Pumping” generally refers to the audible increase of sound levels a\
fter 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, alth\
ough some engineers and musicians use it on purpose occasionally to create a particu\
lar effect."),
quiz::Quiz("Punch In / Punch Out Recording","The process of activating and/or deacti\
vating the record function on tape or DAW during playback of a passage, usually as t\
he performer plays/sings along. This can be used either as a method of doing quick o\
verdubs, or as a way of getting a better take on a certain passage without having to\
 start the track from the beginning."),
quiz::Quiz("Pure Tone","A tone consisting of only the fundamental frequency with no \
overtones or harmonics, graphically represented as a simple sine wave."),
quiz::Quiz("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 (to\
uch 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 con\
vert that pitch to a corresponding voltage that can drive a VCO in unison with the o\
riginal sound."),
quiz::Quiz("PWM","Most oscillators that output a square waveform also have an additi\
onal control voltage input that sets the width of the top portion of the “square” wa\
ve (obviously, making the top portion wider makes the bottom portion narrower and vi\
ce versa). The act of varying the width of the resulting pulse wave creates a sort o\
f Doppler shift; varying the width back and forth – for example, by modulating the p\
ulse width with a low frequency oscillator – creates a chorusing effect that can sou\
nd like a detuned pair of oscillators. The resulting effect is referred to as pulse \
width modulation."),
quiz::Quiz("PZM","Abbreviation for Crown Audio’s Pressure Zone Microphone. (See also\
 “Boundary Microphone.”)"),
quiz::Quiz("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, th\
e broader the bandwidth curve of frequencies that will be boosted or cut. If you com\
e from the pro audio world, you may be used to Q referring to the width or narrownes\
s of a peak or notch filter. In a synthesizer filter, when you increase the resonanc\
e (feedback), a peak forms around the cutoff frequency of the filter’s curve or shap\
e. The higher the resonance, the higher and narrower this peak. As a result, some us\
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ed to use the audio term Q to refer to the resonance amount, although you don’t hear\
 that term used nearly as much today."),
quiz::Quiz("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."),
quiz::Quiz("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."),
quiz::Quiz("Quantization Distortion","Quantization Distortion/Quantization Error – T\
he effective “error in translation” between an analog signal and its sampled counter\
part due to the rounding of a large number of analog values to the nearest digital q\
uantity. This often results in additional random frequencies in the sound, often hea\
rd as noise."),
quiz::Quiz("Quantization Noise","The modulation noise in a signal resulting from qua\
ntization error. "),
quiz::Quiz("Quantization","1) In digital music, the process of adjusting the rhythmi\
c performance of music by moving the notes to precise locations on the time line, ef\
fectively “rounding” the note occurrences to the nearest defined increment. 2) In an\
alog-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)."),
quiz::Quiz("Quantizer","A quantizer auto-corrects the input voltage to the nearest d\
esired 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, bu\
t quite often they are standalone modules."),
quiz::Quiz("Rack Ears","Rack Ears/Rack Flanges – Mounting brackets that can are atta\
ched to equipment so it can be mounted in a standard equipment rack."),
quiz::Quiz("Rack Mounted","Describing outboard gear that can be housed in an equipme\
nt rack."),
quiz::Quiz("Rack Rash","When you mount a module into a case, the head of the screw o\
r bolt used to mount the module can scratch the faceplate of the module. These scrat\
ches are referred to as rack rash. You can almost never see it when you mount a modu\
le, as the scratches are behind the screw or bolt head, but nonetheless some will pa\
y more for a used module that is unscratched. So buy a bag of plastic washers and pu\
t them behind the screw or bolt head just to remove another reason for someone to no\
t buy your used module."),
quiz::Quiz("Rack Unit","Rack-mounted equipment usually follows a standard set of dim\
ensions, including 19” (48.3 cm)for width, and a “rack unit” (or U for short) for he\
ight equaling 1.75” (4.4 cm) per U. Many common modular synthesizer formats follow t\
he rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25” o\
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r 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (\
sometimes referred to as MU for Moog Unit)."),
quiz::Quiz("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 d\
escribes the area where sound arrives at the microphone, while a radiation pattern d\
escribes how sound is dispersed from the loudspeaker."),
quiz::Quiz("Radiation","The angle and pattern of coverage of a speaker."),
quiz::Quiz("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 rai\
ses 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."),
quiz::Quiz("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 h\
ard-drive storage memory where projects are saved and recalled). In the recording st\
udios, 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 MID\
I instruments)."),
quiz::Quiz("Random Note Generator","A device that generates random pitches at a set \
rate, used in synthesizers."),
quiz::Quiz("Random","Most voltages moving around inside a modular synth are very pur\
poseful in their variations: the repeating waveforms of an audio rate or low frequen\
cy oscillator; the rising then falling voltages of an envelope generator. However, i\
t can also be useful to have randomly wandering voltages to create everything from s\
ubtle variations in pitch to wildly varying volumes or filterings. Noise is an examp\
le of an audio-rate random signal."),
quiz::Quiz("Rap","To perform a spoken rhythmic part to a music or percussion perform\
ance."),
quiz::Quiz("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.”)"),
quiz::Quiz("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 bur\
st of four sixteenth notes appearing instead for one or more stages."),
quiz::Quiz("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."),
quiz::Quiz("Rated Load Impedance","The input impedance, or opposition to current flo\
w by an input of a device, that a piece of equipment is designed to feed."),
quiz::Quiz("RCA Plug","(Also called Phono Plug) A common audio connector found on mo\
st stereo systems with a center pin as one connection and an outer shell as the seco\
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nd connection."),
quiz::Quiz("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)."),
quiz::Quiz("Read","To retrieve information bits from a storage device; in digital au\
dio, the reproduction of digital signals."),
quiz::Quiz("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 virt\
ual musical instruments which can be played live or used with other sequencing softw\
are."),
quiz::Quiz("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 electrol\
ytic capacitor to be 25 years. Therefore, serious vintage synth owners “recap” (repl\
ace the age-sensitive capacitors in) their older equipment."),
quiz::Quiz("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."),
quiz::Quiz("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."),
quiz::Quiz("Record Monitor","On some tape machines, a switch position that allows th\
e VU meter and sound output of the tape machine electronics to monitor the input sig\
nal to the tape machine."),
quiz::Quiz("Record Ready","A control state of a multitrack tape recorder where the d\
esignated track will begin recording when the record function of the tape recorder i\
s activated."),
quiz::Quiz("Recording Bus","A bus that sends a mix signals from the console channels\
 to the multitrack recorder or DAW. (See also “Bus.”)"),
quiz::Quiz("Recording Session","A bloc of time in which music is being recorded in t\
he studio."),
quiz::Quiz("Rectifier","A circuit that makes sure a voltage stays only positive or n\
egative. In power supplies, it is used to remove the negative component of AC voltag\
e, or to protect you from plugging in module's power connector backwards. As a modul\
e, a half-wave rectifier passes only positive voltages and replaces anything negativ\
e with 0v; a full-wave rectifier takes any negative voltages and inverts them so the\
y become positive. This effectively doubles the frequency of many simple waveforms, \
like the triangle and sine."),
quiz::Quiz("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 increas\
ing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distanc\
e. It can also be used a slowly changing random control voltage or modulation signal\
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, instead of as an audio source."),
quiz::Quiz("Reel","1) The hub and flanges onto which analog tape is spooled; recordi\
ng and playback involves unspooling the tape from one reel and onto another. 2) Some\
times 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 inten\
ded 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 delivere\
d on reels."),
quiz::Quiz("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 Leve\
l.”"),
quiz::Quiz("Reference Tone","A single-frequency tone (often at 1000 kHz) used to cal\
ibrate the levels of sound equipment; the tone used to set reference level. (See als\
o “Test Tones.”)"),
quiz::Quiz("Reflected Sound","Sound that reaches a microphone or listener after one \
or more reflections from surrounding surfaces."),
quiz::Quiz("Reflection","In acoustics, the bouncing of sound waves off of a flat sur\
face, as opposed to absorption. Reflection can have a great impact on how we perceiv\
e the collective sound; reflected sounds from a distance is perceived as echo, while\
 reverberation is created from thousands of reflections. (See also “Absorption,” “Ea\
rly Reflection,” “Echo,” “Reverberation.”)"),
quiz::Quiz("Regeneration","Regeneration can have a couple of different meanings insi\
de a synth, both meaning feedback. An echo unit can feed some of its output back int\
o its input, causing the delayed signal to be repeated again; this is sometimes refe\
rred to as regeneration. Also, very rarely you will hear resonance in a filter refer\
red to as regeneration."),
quiz::Quiz("Regulated Power Supply","A device to supply power to electronic equipmen\
t whose output voltage will not fluctuate when more equipment is turned on, or if th\
ere is a change in voltage of the power line. A regulated power supply is designed t\
o protect sensitive electronics from destructive power surges."),
quiz::Quiz("Relay","An electromagnetically activated switch that connects or disconn\
ects two terminals when a control voltage is applied."),
quiz::Quiz("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 designa\
ted threshold."),
quiz::Quiz("Release","This refers to the final stage of an envelope that typically f\
alls back to zero volts, usually resulting in silence. It is often used in the conte\
xt of talking about an Attack/Release (AR) or Attack/Decay/Sustain/Release (ADSR) en\
velope generator, but can refer to any final stage of an envelope."),
quiz::Quiz("Remote","1) A device that controls the functions of another device wirel\
essly. 2) Describing on-site recording, as opposed to recording in the studio."),
quiz::Quiz("Reset","The Reset input on a module accepts a trigger or gate signal, an\
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d 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 se\
quencer, it means go back to the first stage. In the case of an envelope, it means g\
o back to the start of the attack. In the case of a gate delay, it means to re-start\
 the timer for the delay."),
quiz::Quiz("Residual Magnetization","The amount of magnetism left in a magnetic mate\
rial after the magnetizing force is removed. Residual magnetism can accumulate in ta\
pe machines over time, either creating distortions and noise in the sound output or \
partially erasing the tape."),
quiz::Quiz("Residual Noise","The noise level left on recording tape after it has bee\
n erased."),
quiz::Quiz("Resistance","The opposition of a substance to the flow of electrical cur\
rent, measured in ohms."),
quiz::Quiz("Resistor","An electrical component with a specific amount of resistance \
to electrical current, used within the circuit to regulate the flow of current."),
quiz::Quiz("Resonance","The natural tendency of physical substances to vibrate with \
more energy at certain frequencies. The principle of resonance is a key element in t\
he design of acoustic instruments; for example, the hollow chamber of a guitar or vi\
olin 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 re\
sult is an increased boost in the harmonics right around the filter’s cutoff or corn\
er 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. The\
refore, the term resonance is often used to refer to a filter’s feedback amount."),
quiz::Quiz("Resonant Frequency","A frequency at which a physical item vibrates natur\
ally."),
quiz::Quiz("Resonate","To vibrate at the resonant frequency. Also refers to the ling\
ering reverberation that causes a sound to continue after the sound source has stopp\
ed. This continuing sound is due to the sympathetic resonance of nearby objects."),
quiz::Quiz("Resonator","Many acoustic instruments include a body or sound chamber th\
at “resonates” – sympathetically vibrates at, or reinforces – one or more frequencie\
s. To simulate this effect in modular synths, you can get a specialized filter or eq\
ualization module that boosts the sound at typically three or so user-definable freq\
uencies, each usually within a narrow band. This is one of the secrets of synthesizi\
ng real-world sounds or spaces."),
quiz::Quiz("Reverb (Reverberation)","1) Short for “Reverberation.” (See “Reverberati\
on.”) 2) A signal processor or plug-in that creates artificial reverb to a signal.")\
,
quiz::Quiz("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 reve\
rb time is measured between the point at which the sound source stops and the point \
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at which the reverberation levels fall by 60 dB."),
quiz::Quiz("Reverb","Short for reverberation. This is an effect device that mimics b\
eing 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 dimensio\
n to what might otherwise be a stark sound. There are relatively few modules that im\
plement a reverb effect, and even fewer that allow you to voltage control some of it\
s parameters (the ErbeVerb being the most famous); many just use an external reverb \
effect."),
quiz::Quiz("Reverberant Field","Describes the space that is far enough from the soun\
d source that the reverberations are louder than the direct sound."),
quiz::Quiz("Reverberation Chamber","A device built to simulate room reflections."),
quiz::Quiz("Reverberation Envelope","The attack, decay, sustain and release of the r\
everberation volume; or how fast the reverberation reaches peak level and its rate o\
f decay."),
quiz::Quiz("Reverberation","The persistence of a sound after the source stops emitti\
ng it, caused by many discrete echoes arriving at the ear so closely spaced in time \
that the ear cannot separate them."),
quiz::Quiz("RF Interference","The unwanted noise introduced into electronics, circui\
ts and/or audio systems by the presence of RF signals. RF interference in a system c\
an result in humming, buzzing, static or even the reproduction of radio transmission\
s."),
quiz::Quiz("RF Signals","RF Signals (or RF) - Short for Radio Frequency Signals, ele\
ctromagnetic waves that carry wireless radio and television signals. The vast majori\
ty of RF signals exist at frequencies higher than 100 kHz."),
quiz::Quiz("Rhythm Section","The musical instruments in a band or ensemble that are \
responsible for playing rhythmic parts rather than melody parts. In contemporary mus\
ic, rhythm sections typically consist of drums and bass, along with some combination\
 of percussion, piano/keyboard and/or guitars."),
quiz::Quiz("Ribbon Controller","This is a long strip that is capable of measuring th\
e position where you press it along its length, and the pressure used to press it. I\
t can be used as an alternate keyboard or as a pitch bend controller, with the posit\
ion determining pitch. Shorter versions also appeared sometimes as alternate control\
lers on synthesizers, such as the Yamaha CS-80."),
quiz::Quiz("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-di\
rectional or figure-8 pattern."),
quiz::Quiz("Riff","A short melody repeatedly played in a tune often with variation b\
etween vocal lines."),
quiz::Quiz("Ring Modulator","Balanced or ring modulation is a special type of amplit\
ude modulation, where one bipolar (swinging both above and below 0 volts) signal - t\
he modulator - is used to vary the amplitude of a second bipolar signal, known as th\
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e carrier. The modulator’s frequency is both added to and subtracted from the carrie\
r’s frequency; the resulting harmonics replace the original carrier and modulator.")\
,
quiz::Quiz("Ringing Out a Room","The process of identifying and compensating for pro\
blem 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 f\
requencies."),
quiz::Quiz("Rise Time","The rate at which an audio waveform makes a sudden increase \
to a higher amplitude."),
quiz::Quiz("RMS Meter","A meter that recognizes and responds to the effective averag\
e, the RMS level, or the effective average value of an AC waveform, rather than to t\
he peak level. (See also “Root-Mean-Square,” “Peak Meter.”)"),
quiz::Quiz("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."),
quiz::Quiz("Room Equalization","In live audio, an equalizer inserted in the monitor \
system that attempts to compensate for frequency response changes caused by room aco\
ustics."),
quiz::Quiz("Room Sound","The natural ambience of a room, including the reverberation\
 and background noise."),
quiz::Quiz("Room Tone","The natural background noise occurring in a room without mus\
ic playing or people speaking. In recording audio for film and TV, on-set sound mixe\
rs capture a take of room tone for the purpose of providing continuity between clips\
 of dialogue during post-production."),
quiz::Quiz("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 th\
e peaks. (See also “RMS Metering,” “Peak Metering.”)"),
quiz::Quiz("Rotating Head","A circular head with two (or more) gaps that rotates aga\
inst the direction of tape motion at a slight angle to the tape travel."),
quiz::Quiz("Rumble","A low-frequency noise, typically caused by earth/floor vibratio\
n or by uneven surfaces in the drive mechanism of a tape recorder or playback unit."\
),
quiz::Quiz("Rythm Tracks","The recording of the rhythm instruments in a music produc\
tion."),
quiz::Quiz("S-trig","Some systems - such as the original Moog modular - use an s-tri\
gger (switch or shorting trigger) instead of a normal gate, which was a wire that wa\
s shorted to 0 volts ground, like the closing of a switch wired to ground. You canno\
t interconnect these two systems without some form of conversion between the two, wh\
ich can be as simple as a special cable."),
quiz::Quiz("S/H","A sample and hold (S/H) module has two inputs: a signal that is be\
ing sampled, and a trigger input that indicates when the first input should be sampl\
ed. When a trigger is received, the current voltage at the first input is sampled (m\
easured) and held (stored), and presented at the output. This stable voltage is held\
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 until a new trigger is received. Sample and holds are most often associated with cr\
eating 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, whic\
h is then dutifully sent to the output."),
quiz::Quiz("S/PDIF","Abbreviation for “Sony/Phillips Digital Interface,” a protocol \
for sending and receiving digital audio signals using a common RCA connector."),
quiz::Quiz("Safety Take (ST)","An additional take of audio captured for good measure\
 after a take of acceptable quality has been recorded."),
quiz::Quiz("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.\
"),
quiz::Quiz("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 shoul\
d 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 volta\
ge is held until a new trigger is received. Sample and holds are most often associat\
ed 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 v\
alue, which is then dutifully sent to the output."),
quiz::Quiz("Sample Dump Standard (SDS)","See “MIDI Sample Dump Standard.”"),
quiz::Quiz("Sample Rate Conversion","The conversion of digital audio taken at one sa\
mple rate to a different sample rate without first converting the signal to analog."\
),
quiz::Quiz("Sample Rate","This is a specification of digital audio: How fast the ind\
ividual measurements (samples) that reconstruct a sound are recorded or played back.\
 The bandwidth of that audio file (which corresponds to the highest frequency that c\
an 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 sa\
mple 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 resolu\
tion of a digitally reproduced sound are described as a combination of sample rate a\
nd bitrate. (See also “Bitrate.”)"),
quiz::Quiz("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 pur\
pose of reproducing and manipulating the sound digitally."),
quiz::Quiz("Sampler","A device that records and plays samples, often with features f\
or editing, manipulating and storing the samples."),
quiz::Quiz("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 inst\
ead held at that limit. However, saturation usually implies a more rounded, shaped a\
pproach to that clipping limit, resulting in a more pleasing (or at least less annoy\
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ing) 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 proc\
essing. 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 exces\
s of sound level. This creates some distortion that some audiophiles describe as “an\
alog 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 co\
lor and warmth in the sound that engineers often find appealing. 3) A digital plugin\
 that emulates tape or tube saturation."),
quiz::Quiz("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 “R\
amp Wave.”)"),
quiz::Quiz("Sawtooth","One of the most common waveforms produced in a synthesizer. T\
his ramp-shaped wave contains both even and odd harmonics, strongest at the fundamen\
tal frequency (the note being played) and diminishing at the higher frequencies. The\
 result is very bright, loud, “brassy” sound."),
quiz::Quiz("Schmitt Trigger","This is a type of gate detector that looks at a varyin\
g input signal and outputs either a “high” (typically 0, 10, or even 15 volts) signa\
l or a “low” signal (typically 0 volts). When the input goes above one reference thr\
eshold – 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."),
quiz::Quiz("scope","This is a piece of test equipment that displays voltage fluctuat\
ions as graphical waveforms. A ’scope can run at a wide range of frequencies, displa\
ying slowly changing voltages like LFOs or envelopes, or quickly changing voltages l\
ike oscillators and noise. Oscilloscopes used to be bulky pieces of external equipme\
nt, but now you can get USB scopes that offload the display portion of the job to yo\
ur computer, or scopes as modules."),
quiz::Quiz("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 pa\
rts. At a later date the final vocal track is overdubbed. 3) The action of a musicia\
n or disc jockey quickly moving a record back and forth on a turntable reproducing t\
he stylus motion to create a rhythm pattern of sound."),
quiz::Quiz("Scrubbing","The action or function of shuttling a piece of recorded audi\
o back and forth while monitoring it, typically to locate a certain point in the rec\
ording. In earlier days, scrubbing was done with reel-to-reel analog tape by manuall\
y turning the reels to pull the tape across the playhead. Today, scrubbing is primar\
ily done digitally on a DAW by dragging the cursor back and forth across the wavefor\
m."),
quiz::Quiz("Second Engineer","An assistant recording engineer."),
quiz::Quiz("SEM","The Oberheim SEM (Synthesizer Expander Module) was one of their ea\
rliest products. It was an entire synthesizer voice – two oscillators, two simple en\
velopes, VCA, and a very popular two-pole state variable filter design with a knob t\
hat crossfaded between low pass, notch, and high pass outputs plus a separate bandpa\
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ss 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 t\
he original Oberheim synth."),
quiz::Quiz("Semi-modular","The components of a semi-modular synth - such as the osci\
llator, filter and amplifier - are pre-wired behind the front panel in what the manu\
facturer considers to be a typical, logical way. However, they also provide patch po\
ints 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 w\
ho are new to modular synthesis dip their toe in the water by getting a semi-modular\
 synth, and then expanding it with additional modules."),
quiz::Quiz("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 d\
ivision is 1/12 of an octave."),
quiz::Quiz("Send Level","A control determining the signal level sent to a send bus."\
),
quiz::Quiz("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 o\
f another microphone from the same sound level. 2) In music, describes the artistic \
persona in general."),
quiz::Quiz("Sequence","1) A pre-programmed set of musical events, such as pitches, s\
ounding 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 ref\
erring to a segment of music in general."),
quiz::Quiz("Sequencer","The most common type of sequencer you’re going to see in a m\
odular synth contains a row of knobs (also known as steps or stages) that may each b\
e set to output a different voltage. A sequencer then goes through steps one at a ti\
me. This is most often used to create repetitive musical lines where each note has t\
he same duration, which is popular in trance-like forms of music as well as the clas\
sic Berlin School style (70s-era Tangerine Dream and Klaus Schulze; current Red Shif\
t and Node). A computerized device or software that can be programmed to play a step\
ped order of musical events, including playing of pitches, sounding of samples, and \
rests."),
quiz::Quiz("Sequential Switch","This module comes in a few different forms; in the m\
ost common, a few different inputs are routed to one output (although they are usual\
ly 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 t\
o 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 mig\
ht directly route a series of inputs to corresponding outputs (with usually a summed\
 output as well)."),
quiz::Quiz("Serial Data","A digital data stream where individual bits are transmitte\
d one after another over a single connection (as opposed to “parallel data,” in whic\
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h multiple bits can be sent at once). Most data connections in the recording studio \
transmit serial data—for example, USB, Firewire and MIDI."),
quiz::Quiz("Series Connection","Connecting devices (especially circuit elements) so \
that the electrical signal flows from one thing to the next, to the next, etc."),
quiz::Quiz("Set Up","The positioning of microphones, instruments, connections and mo\
nitoring in the studio, as well as the controls and levels on consoles, DAWs, etc., \
in preparation for recording."),
quiz::Quiz("Shelf Filter","A name for the circuit in an equalizer used to obtain the\
 shelf."),
quiz::Quiz("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 i\
t; a low-frequency shelf does the same for signals at and below the set frequency.")\
,
quiz::Quiz("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 electromagn\
etic interference."),
quiz::Quiz("Shielded Cable","Cable that has a shield around an inner conductor or in\
ner conductors."),
quiz::Quiz("Shock Mount","An elastic mount on microphone stand that reduces the impa\
ct of unwanted vibrations that may affect the stand (for example, floor vibrations f\
rom footsteps)."),
quiz::Quiz("Short Circuit","A direct connection between two points in a circuit that\
 (usually) should not be connected."),
quiz::Quiz("Short Delay","Delay times under 20 milliseconds."),
quiz::Quiz("Shortest Path","A technique in recording that routes the signal through \
the least amount of active (amplified) devices during recording."),
quiz::Quiz("Shotgun Microphone","A microphone with a long line filter, a tube that a\
coustically cancels sound arriving from the side, to make the microphone pick up muc\
h 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."),
quiz::Quiz("Sibliance","Energy from a voice centered around 7 kHz, caused by pronoun\
cing “s”, “sh” or “ch” sounds."),
quiz::Quiz("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 sidechaini\
ng is in compressors, particularly in ducking effects where the presence of a partic\
ular audio signal triggers the compression of another audio signal. (See also “Ducki\
ng.”)"),
quiz::Quiz("Signal Flow","1) In the general sense, the path that an audio signal tra\
vels from the sound source to the system output. (For example, from the vocalist’s v\
oice into the microphone, through the cables, into the preamp, out of the preamp int\
o the console, through all inserts and buses, and output into the DAW for recording.\
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) 2) Signal flow is often specifically meant to refer to the routing of an audio sig\
nal through the console, from input to output."),
quiz::Quiz("Signal Processing","The practice of altering the character or sound of a\
n audio signal through a variety of devices or plug-ins, such as equalizers, compres\
sors, reverb units, etc."),
quiz::Quiz("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."),
quiz::Quiz("Signal","1) In audio, an alternating current (or voltage) matching the w\
aveform of, or being originally obtained from, a sound pressure wave. 2) Also in aud\
io, an alternating current (or voltage) between 20 Hz and 20,000 Hz. 3) A digital au\
dio bit stream."),
quiz::Quiz("Sine Wave","1) In the general sense, the path that an audio signal trave\
ls from the sound source to the system output. (For example, from the vocalist's voi\
ce 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 signa\
l through the console, from input to output."),
quiz::Quiz("Sine","This is the purest waveform: It contains only the fundamental har\
monic, and no higher harmonics. As a result, it's a great wave to use to create a su\
b 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 modul\
ation (AM), or wavefolding which add or shift harmonic content."),
quiz::Quiz("Slap Echo (also called Slapback)","A single, distinct echo of a sound, w\
hich can result naturally from higher frequencies reflecting off a non-absorbent wal\
l, or artificially reproduced by a signal processing unit or plugin. Slap echo creat\
es a "live" sounding effect similar to what you would hear in an arena."),
quiz::Quiz("Slate","Slate (Slating) – 1) In video/film, the identification of a scen\
e and take at the beginning of the clip for the purpose of video editing. This is do\
ne by presenting the scene/take in written form in front of the camera on a clapboar\
d, 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 tr\
ack. 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 v\
isually on the screen."),
quiz::Quiz("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 instr\
ument that is being operated remotely by MIDI information sent from another device."\
),
quiz::Quiz("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 re\
sult, it is sometimes called a lag generator or processor, or more technically as an\
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 integrator."),
quiz::Quiz("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 nu\
mber of nuts are inserted into these channels, which can then be slid to any positio\
n to accommodate the mounting hole spacing of your modules. In a Eurorack case, thes\
e nuts tend to have a 2.5mm or 3mm hole and corresponding thread."),
quiz::Quiz("Slope Generator","A slope generator creates ramps: rising or falling vol\
tages. It is essentially a gate generator and a slew limiter (see above) wired toget\
her 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 gener\
alized control voltage functions – even creating a sawtooth or triangle wave oscilla\
tor – some companies such as Buchla and Serge referred to by its elemental function \
of generating sloping voltage changes."),
quiz::Quiz("Slope","Most filters typically have a cutoff or corner frequency they ar\
e tuned to. It then reduces (filters) the frequency spectrum of a signal going throu\
gh it so that it 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 h\
ave slopes that are defined multiples of 6 decibels (dB) weaker for each octave furt\
her away you get from the cutoff frequency. For example, a low-pass filter (LPF) wit\
h a slope of 24 dB/octave would attenuate harmonics one octave above its cutoff freq\
uency by 24 decibels."),
quiz::Quiz("Smart FSK (Frequency-Shift Key)","Smart FSK – An updated form of Frequen\
cy-Shift Key (FSK) sync that enables MIDI devices to sync to analog tape recorders a\
nd/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 be\
at for MIDI sequencers/devices at any point in the recording. This enables the devic\
e to start playing at exactly the right place and tempo no matter where you start th\
e tape. (See also “Frequency-Shift Key.”)"),
quiz::Quiz("SMPTE Time Code","(Abbreviated “SMPTE“) A standardized timing and sync s\
ignal 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 purp\
oses in audio recording environments. Many audio professionals simply refer to this \
time code as “SMPTE.”"),
quiz::Quiz("SMPTE","1) Abbreviation for Society of Motion Picture and Television Eng\
ineers. 2) See “SMPTE Time Code.”"),
quiz::Quiz("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."),
quiz::Quiz("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.”)"),
quiz::Quiz("Soft Knee","In compression, refers to the gradual introduction of compre\
ssion of the signal once the sound level crosses the threshold. (See also “Knee.”)")\
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quiz::Quiz("Software Instrument (Virtual Instrument)","One of a number of software-b\
ased synthesizers, samplers or sound samples that are stored and accessed via comput\
er and performed by an external MIDI controller, rather than in a standalone synthes\
izer or module. Because of the wide versatility available from these instruments, a \
growing number of composers and electronic musicians are working with virtual instru\
ments that can be stored in hard drives, rather than purchasing stacks of keyboards \
and modules."),
quiz::Quiz("Soldering","The action of making connections with solder, a soft metal a\
lloy that is used to bond two metal surfaces by melting. In audio settings, solderin\
g 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 b\
uying new ones or sending them off for repair."),
quiz::Quiz("Solid State","In electronics, refers to the use of transistors and semic\
onductors (solid materials) in the building of electronic devices, as opposed to tub\
es. In the recording studio, solid state amplifiers have different properties than t\
ube amps, and each has its own advantages and disadvantages. A more recent applicati\
on of solid state construction is in computer devices, particularly solid state hard\
 drives (SSD), which transfer data more quickly than conventional spinning disc driv\
es, and are less prone to breakage."),
quiz::Quiz("Solo Switch","A switch that activates the solo function on a console or \
DAW."),
quiz::Quiz("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 featur\
ed above other instruments."),
quiz::Quiz("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.\
"),
quiz::Quiz("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."),
quiz::Quiz("Sound Effects (SFX)","Sounds other than dialogue, narration or music tha\
t are added to audio, usually in the context of film/video."),
quiz::Quiz("Sound File","A digital audio recording that can be stored in a computer \
or on a digital storage medium (such as a hard disk)."),
quiz::Quiz("Sound Modeling","A technique that recreates a sound without directly mod\
eling the physical device. An example is additive synthesis, which uses a combinatio\
n of sine waves and noise to recreate sounds."),
quiz::Quiz("Sound Module","An electronic instrument (tone generator, synth or sample\
r playback unit) that has no playable interface, but instead responds to incoming MI\
DI 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 configurat\
ion. Today, sound modules can also occur as software versions or plugins to be acces\
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sed on a computer."),
quiz::Quiz("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 pe\
rceive SPL in terms of amplitude, volume or loudness of the sound."),
quiz::Quiz("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."),
quiz::Quiz("Sound Source","The origin of a sound, whose vibrations create sound wave\
s."),
quiz::Quiz("Sound Wave","(Also called “Sound Pressure Wave”) A wave caused by a vibr\
ation that results in slight variations in air pressure, which we hear as sound."),
quiz::Quiz("Soundtrack","1) Broadly speaking, refers to any/all audio that accompani\
es an instance of visual media, whether music, dialogue or SFX. 2) In more common te\
rms, refers to the musical score and/or licensed music synced to a film, video, TV p\
rogram or video game."),
quiz::Quiz("Source of Uncertainty","This was the name for the Buchla 265 and 266 mod\
ules that create random control voltages. Its name is often used for random source m\
odules that follow or are inspired by the original Buchla template."),
quiz::Quiz("Spaced Pair","(Also called “A/B Technique“) A stereo microphone placemen\
t technique in which two cardioid or omnidirectional microphones are spaced somewher\
e between 3-10 feet apart from each other (depending on the size of the sound source\
) to create a left/right stereo image."),
quiz::Quiz("Speaker","A device that converts electrical signals to sound; more techn\
ically, a transducer that changes an electrical audio signal into sound pressure wav\
es."),
quiz::Quiz("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, liquid\
s and gases, and though we usually think of sound as traveling through the air, diff\
erences in temperature, air pressure and humidity can also affect how fast sound tra\
vels.  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 ai\
r at sea level at a temperature of 15 degrees Celsius (59 degrees Fahrenheit). By co\
ntrast, at 70 degrees Fahrenheit under standard atmospheric conditions, the speed of\
 sound is about 344 m/s, or 770 mph."),
quiz::Quiz("Splicing","Historically, the act of attaching previously cut pieces of a\
udio tape or film in precise locations by applying a special kind of adhesive tape o\
n the back. This is/was done for the purpose of shortening sections of audio or edit\
ing film. Today, splicing has become a very simple process by editing sections of au\
dio or video digitally with a DAW or film editing software."),
quiz::Quiz("Splitter","The short definition is something that can divide a signal in\
to two or more copies, such as a splitter cable where two outputs are wired to one i\
nput. For a deeper discussion, see the entry on multiple, as there are ways of going\
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 about this beyond simple wiring."),
quiz::Quiz("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 o\
ften pleasing chorusing sound. Depending on the module, you might even be able to sp\
read these tones to form intervals, triads, and chords."),
quiz::Quiz("Spring Reverb","A device that simulates reverberation by creating vibrat\
ions within a metal spring by attaching it to a transducer and sending the audio sig\
nal through it. A pickup at the other end converts those vibrations into an electric\
al signal which is mixed with the original audio signal. While the physical spring r\
everbs still exist, most studios emulate spring reverb with the use of plug-ins or h\
ardware reverb units."),
quiz::Quiz("Square wave","This is a common waveform produced by a synthesizer’s osci\
llator. 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 positiv\
e voltage). Aside from being a really easy waveshape to generate with analog circuit\
ry, it has an interesting harmonic series: it has a strong fundamental, then gradual\
ly weaker odd harmonics: a component at three times the fundamental frequency, one a\
t fives time 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 harmo\
nics 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."),
quiz::Quiz("Stackable Cable","Many banana style cables are constructed that each plu\
g 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."),
quiz::Quiz("Stage Monitor","A speaker on the stage that enables performers to hear t\
hemselves and to hear what the other musicians are playing on stage."),
quiz::Quiz("Stage","1) The partially enclosed or raised area where live musicians pe\
rform. 2) In reverberation effects devices, an echo added before the reverberation t\
o 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 sequenc\
er, 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 attac\
k, where the envelope generally rises from 0 volts to the highest voltage it can out\
put - is a stage. You might also hear it used to describe the number of sample stage\
s in a BBD (Bucket Brigade Delay), described elsewhere."),
quiz::Quiz("Standard Operating Level","A reference voltage level or maximum average \
level that should not be exceeded in normal operation."),
quiz::Quiz("Standing Wave","An unwanted sound wave pattern that often occurs when th\
e sound wave bounces between two reflective parallel surfaces in a room, and the ref\
lected waves interfere with the initial wave coming from the sound source, in which \
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the combined wavelength of the affected frequency is effectively the length of the r\
oom. This creates the audible illusion that the wave is standing still, so the frequ\
ency is amplified to an unwanted level in certain parts of the room while nearly abs\
ent in others. Standing waves are most common in square or rectangular rooms with pa\
rallel surfaces, so acoustic designers try to prevent these waves by installing abso\
rptive materials or introducing other items to offset the parallel surfaces."),
quiz::Quiz("Step Mode","A setting in a sequencer or DAW in which notes are input man\
ually, one note or step at a time."),
quiz::Quiz("Step Sequencer","This usually refers to a type of sequencer where you st\
ep to and pause on a stage, enter the note (and possibly the duration) for that stag\
e, move on to the next step, and so forth."),
quiz::Quiz("Step","Step is often used interchangeably with stage (see above), especi\
ally when talking about sequencers."),
quiz::Quiz("Stereo Image","The audible perception of stereo, in which different soun\
ds sources appear to be coming from far left, far right or any place in between."),
quiz::Quiz("Stereo Micing","Placement of two or more mics so that their outputs comb\
ine to create a stereo image."),
quiz::Quiz("Stereo","A recording or reproduction of at least two channels where posi\
tioning of instrument sounds left to right can be perceived."),
quiz::Quiz("Strike","This term appears on several Make Noise modules, although it ha\
s 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 l\
ight source such as an LED, allowing a voltage to be turned into a resistance to con\
trol a parameter) may have a strike input. When you flash an LED at a light sensitiv\
e resistor, it does not change the resistance instantaneously and stay there - inste\
ad, there is some delay as it glides to the desired resistance. When you turn the LE\
D off, the resistance may not go instantaneously to full; instead it might take a br\
ief 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 equipmen\
t and clean up after a recording session."),
quiz::Quiz("Subcode","Additional information bits that are recorded alongside digita\
l audio, used for control and playback purposes."),
quiz::Quiz("Subframe","A unit smaller than one frame in SMPTE time code."),
quiz::Quiz("Subgroup","A number of input channels on a console that can be controlle\
d and adjusted as a single set before sending the combined signal to the master outp\
ut. Sometimes also called “Submix,” “Bus” or just “Group.”"),
quiz::Quiz("Subharmonic","A circuit that divides the fundamental harmonic of the inc\
oming sound to produce lower frequencies, and therefore subharmonics. The most commo\
n is an octave divider or sub bass circuit that divides creates a subharmonic by div\
iding the fundamental by 2 (some can also create a subharmonic two octaves below the\
 fundamental by dividing it by 4)."),
quiz::Quiz("Submaster / Sub-Master","The fader which controls the combined level of \
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sound from several channels during mixdown or recording."),
quiz::Quiz("Submix","See “Subgroup.”"),
quiz::Quiz("Suboctave","A module that creates a new tone one or two octaves below th\
e 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 sin\
e-like, or that more closely resembles the original waveform."),
quiz::Quiz("Subtractive Synthesis","The most common synthesis technique: You start w\
ith one or more oscillators outputting waveforms with a large number of harmonics, a\
nd 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 ad\
ds articulation to the note by varying its loudness. An old-school method of sound s\
ynthesis in which sounds are designed and created by generating harmonically rich wa\
veforms, then filtering out unwanted harmonics to arrive at the desired sound."),
quiz::Quiz("Sum","To sum is a fancy way of saying you added two (or more) things tog\
ether; the sum is the result. It usually is used in the context of adding together c\
ontrol 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."),
quiz::Quiz("Summing","The process of blending two or more signals into one mixed sig\
nal. 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."),
quiz::Quiz("Super-Cardioid Pattern","A very tight cardioid microphone pattern with m\
aximum sensitivity on axis and the least amount of sensitivity approximately 150 deg\
rees off-axis."),
quiz::Quiz("Surround Sound","A technique of recording and playback in which the list\
ener 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. Surroun\
d sound can occur in various formats with different numbers of speakers arrayed thro\
ugh the room. Surround sound today is most commonly used in film and TV production."\
),
quiz::Quiz("Sustain","This is a common stage of an envelope generator where a voltag\
e – usually being sent to a filter’s cutoff frequency or an amplifier’s level – is b\
eing 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 co\
ntrol for how long the sustain stage should last. Envelopes that contain sustain sta\
ges include the ADSR (Attack/Decay/Sustain/Release) and AR (Attack/Release, which us\
ually assumes a sustain stage)."),
quiz::Quiz("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 a\
vailable simultaneously. Another side effect is that they are not prone to oscillati\
ng at high feedback (resonance) settings, although some have certainly figured out h\
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ow to make this happen. The Oberheim SEM (Synthesizer Expander Module) filter is per\
haps the most famous state variable design."),
quiz::Quiz("Sweetening","A vague term referring to the fine-tuning of audio in the p\
ost-production stage of recording. Effectively, any small “tweaks” to to make the au\
dio sound better is considered sweetening."),
quiz::Quiz("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. Y\
ou cannot interconnect these two systems without some form of conversion between the\
 two, which can be as simple as a special cable."),
quiz::Quiz("Switch","A device that makes and/or breaks electrical connections."),
quiz::Quiz("Switchable Pattern Microphone","A microphone having the capability of tw\
o or more pickup patterns, which can be toggled by use of a switch on the microphone\
."),
quiz::Quiz("Switching Power Supply","A switching power supply starts by directly con\
verting 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 switc\
hed voltage is then smoothed out to create a constant DC supply at the desired volta\
ge. 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 (t\
his is why they are often surrounded by a shielding cage). Many are moving to a hybr\
id power supply that combines a switcher with a small linear supply or regulator to \
get the best of both worlds."),
quiz::Quiz("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 tr\
acks stay in sync with each other through all stages of post-production."),
quiz::Quiz("Sync24","Sync24 is an alternate name used for the Roland-created standar\
d 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."),
quiz::Quiz("Sync","Sync can have two different meanings, depending on whether we're \
talking about oscillators or about clock signals. Some oscillators support a mode wh\
ere they reset their waveshapes to the beginning when they receive a signal from ano\
ther oscillator. If there is not a precise octave relationship between the two oscil\
lators, 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” sou\
nds 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 patter\
ns, it is common to send a master timing or sync signal around the modular for all t\
he 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\
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 two recording/playback devices timed to a sync signal like SMPTE Time Code, synchro\
nizing 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.”"\
),
quiz::Quiz("Synthesizer Expander Module","The Oberheim SEM (Synthesizer Expander Mod\
ule) was one of their earliest products. It was an entire synthesizer voice – two os\
cillators, two simple envelopes, VCA, and a very popular two-pole state variable fil\
ter design with a knob that crossfaded between low pass, notch, and high pass output\
s plus a separate bandpass setting – in a cube-like case. Most often today, when a m\
odular manufacturer uses the magic letters \"SEM\", they're referring to a filter me\
ant to emulate that in the original Oberheim synth."),
quiz::Quiz("Synthesizer","A musical instrument that uses electrical oscillators to g\
enerate tones artificially, either to simulate the sounds of other instruments or to\
 create other sounds not possible with other instruments."),
quiz::Quiz("System Exclusive","System Exclusive (SysEx) – A MIDI message that will o\
nly be recognized by a unit of a particular manufacturer."),
quiz::Quiz("Tach/Tachometer","In analog tape recording, a device on the recorder tha\
t measures and regulates tape speed by emitting pulses as the tape moves across the \
head."),
quiz::Quiz("Tails Out","A method of winding audio tape so that the end of the last r\
ecorded selection is at the outside of the reel."),
quiz::Quiz("Take Notation","Writing down the takes of the tune being recorded on a t\
ake sheet or on the track log with comments. Take notation was/is recommended for an\
alog tape recording, but in most studios, this function is now accomplished on the D\
AW."),
quiz::Quiz("Take","The recording that is done between one start and stop of a tape r\
ecorder or DAW."),
quiz::Quiz("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 ha\
ve been used as an effect for guitars, but they can be used to modify other instrume\
nts, as well."),
quiz::Quiz("Talkback","A microphone in the control room carried on a separate circui\
t from the recorded channels, allowing the engineer to communicate with the musician\
s in the live room or sound booths through the monitoring system."),
quiz::Quiz("Tape Delay","A signal processing technique for creating artificial delay\
 or echoes by manipulating time delays with analog tape machines. This technique beg\
an 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 coul\
d adjust the length of the delay by adjusting the distance between the record and pl\
ayback heads. Today, most tape delay effects in the studio are simulated digitally t\
hrough plug-ins in a DAW."),
quiz::Quiz("Tape Guide","Any stationary or rotating device which directs the tape pa\
st the heads on a tape machine, or from one reel to the other."),
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quiz::Quiz("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 mos\
t of the noise floor that occurs in analog recording, and can be reduced by using ta\
pe constructed of finer magnetic particles. (See also “Noise Floor.”)"),
quiz::Quiz("Tape Loop","A length of tape with the ends spliced together so that the \
recording will play continuously."),
quiz::Quiz("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 h\
igh-frequency signals recorded above the tape hiss."),
quiz::Quiz("Telephone Filter","A filter used to simulate the audio heard through a t\
elephone receiver by removing signals at frequencies below 300 Hz and above 3500 Hz.\
"),
quiz::Quiz("Tempo Mapping","The act of programming a sequencer or DAW to follow the \
tempo variations of a recorded performance. Unlike beat mapping or beatmatching, bot\
h of which effectively adjust the recording to fit a set tempo, tempo mapping adjust\
s the tempo of the project (especially the MIDI instruments) to match the natural te\
mpo nuances of the recorded material. (See also “Beat Mapping,” “Beatmatching.”)"),
quiz::Quiz("Tempo","The rate at which the music moves, measured in Beats Per Minute \
(BPM)."),
quiz::Quiz("Terminal","1) A point of connection between two wires, including the plu\
g on the end of a cable, and the jack on a piece of equipment. 2) Refers to the keyb\
oard and monitor of a computer that enable the user to enter information and to acce\
ss data."),
quiz::Quiz("Test Oscillator","A device that generates audio waveforms at various fre\
quencies for testing purposes."),
quiz::Quiz("Test Pressing","One of a few initial vinyl record copies pressed from th\
e first stamper made, which is listened to and visually inspected to approve the qua\
lity before more copies are pressed."),
quiz::Quiz("Test Tones","1) A recording of several single-frequency tones at the beg\
inning of a tape reel at the magnetic reference level that will be used to record th\
e program. 2) Artificially generated tones that are used to calibrate an audio syste\
m."),
quiz::Quiz("Thin Sound","A vague term describing an audio signal that that is lackin\
g in certain frequencies, especially on the low end. Over-filtering a signal with an\
 EQ can produce a thin sound, for example."),
quiz::Quiz("Threaded Inserts","A common system for mounting modules into a case is c\
alled sliding rails or nuts. A number of nuts are inserted into these channels, whic\
h 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 mount\
ing hole spacing. They may be drilled for 2.5 or 3 mm screws; pay attention when buy\
ing the rails or a case that has them pre-installed."),
quiz::Quiz("Three-To-One Rule","A principle of microphone placement that says when m\
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ultiple mics are used at once, the distance between microphones should be at least t\
hree 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."),
quiz::Quiz("Three-Way Speaker","A speaker system that has separate speakers to repro\
duce the bass, mid-range and treble frequencies."),
quiz::Quiz("Threshold of Hearing","Described as the sound pressure level at which pe\
ople can hear only 50 percent of the time."),
quiz::Quiz("Threshold","A threshold is generally a voltage level a signal needs to c\
ross before a module takes an action. For example, when the output of an envelope fo\
llower (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 indic\
ating a note has started. When the output of the envelope follower falls before a th\
reshold (which may be the same or different than the note-on threshold), then the ga\
te 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."),
quiz::Quiz("Throat","In a speaker, the small opening in a horn or in a driver throug\
h which the sound wave passes from the driver to the horn."),
quiz::Quiz("Through-Zero Frequency Modulation","TZFM is the abbreviation for Through\
-Zero Frequency Modulation. Think of a patch where you feed the output of one oscill\
ator (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 i\
s added to the normal pitch control voltage for the carrier, and the pitch of the ca\
rrier goes up. As the waveform output of the modulator goes below zero, it is subtra\
cted from the normal pitch control voltage, and the pitch goes down. But what happen\
s 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 mo\
dulation, the carrier will start playing backwards – in essence, a negative frequenc\
y. This generally produces a more pleasing result, and is a desirable characteristic\
 for an oscillator."),
quiz::Quiz("Throw","1) In speakers and in microphones, describes the amount of unres\
tricted 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 de\
signed for smaller spaces has a “short throw,” while one designed for a farther proj\
ection 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 e\
xample, emits high frequencies in a limited angle of direction, so it has a “long th\
row,” while a subwoofer emits low frequencies in all directions and has a “short thr\
ow.” 3) Something a producer, engineer or musician might do with whatever is in his/\
her hand during a moment of intense frustration."),
quiz::Quiz("Tie Lines","Tie Lines – Cables with connectors at both ends, which are u\
sually 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 \
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track of signal flow."),
quiz::Quiz("Timbre","This word is often used to describe the unique tonal characteri\
stic of a sound you are creating, separate from its pitch or loudness. Different sou\
nds, by definition, have different timbres. When you change a parameter of a sound t\
hat changes its tonal characteristic – such as changing the filter cutoff, pulse wid\
th, amount of wavefolding, etc. – you are changing its timbre. The timbre often chan\
ges during life of a note. The sound quality that makes one instrument sound differe\
nt from other instruments, even while playing the same pitch. The timbre of a trumpe\
t, for example, is what makes it sound like a trumpet and not like a flute. Timbre i\
s largely shaped through the presence, absence and complexity of harmonics when the \
instrument is played."),
quiz::Quiz("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 MID\
I Time Code (MTC) and SMPTE time code."),
quiz::Quiz("Time Compression / Expansion","(Also called “Time Stretching” or “Time S\
hifting“) The process of speeding up or slowing down an audio recording without chan\
ging the pitch of the sounds."),
quiz::Quiz("Time Constant","A complex mathematical ides that basically describes the\
 time delay between when an electrical voltage is applied to a circuit and when the \
circuit responds to it."),
quiz::Quiz("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 o\
r might even damage a 1/8” jack."),
quiz::Quiz("Toms","The small drums (as little as 10 inch diameter) that mount on rac\
ks above the kick drum and the large drums in a drum set."),
quiz::Quiz("Tone Generator","1) A device that puts out test tones at various frequen\
cies to align a tape machine or for other testing purposes. 2) The circuits in a syn\
thesizer that create the audio signals put out by the unit, usually to emulate the s\
ound of another instrument."),
quiz::Quiz("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 frequenc\
ies."),
quiz::Quiz("Tonguing","The technique of controlling the start of a note in a brass o\
r woodwind instrument with the tongue."),
quiz::Quiz("Total Harmonic Distortion (THD)","The measure of the difference between \
the level of harmonic frequencies at the output stage of an amplifier as compared wi\
th the input stage, a ratio expressed as a percentage. It’s a fine-tuning specificat\
ion barely perceptible to many ears, but the lower the THD, the more accurately the \
amplifier/speaker is reproducing the sound."),
quiz::Quiz("Touch Sensitive","See “Velocity Sensitive.”"),
quiz::Quiz("Track & Hold","This is a variation of a Sample & Hold. Both have two inp\
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uts - a gate signal, and a voltage reference signal - and a voltage output. When a S\
ample & Hold receives a gate high signal, it freezes and outputs the voltage referen\
ce 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, t\
he reference input it passed along to the voltage output (this is the “tracking” pha\
se); when the gate goes low, the input voltage at that instant is frozen and maintai\
ned at the voltage output until a new gate high signal is received."),
quiz::Quiz("Track Log / Track Assignment Sheet","Track Log/Track Assignment Sheet - \
A sheet of paper kept with a multitrack tape which tells which instrument was record\
ed on each track."),
quiz::Quiz("Track","1) One audio recording made on a portion of the width of a multi\
track tape, or created as a digital representation using a DAW. 2) One set of contro\
l commands in a sequencer or DAW that is used to control one instrument over one MID\
I channel. 3) See “Band Track.”"),
quiz::Quiz("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 oct\
ave system where a rise of 1.00 volts on the pitch input should produce exactly a do\
ubling (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 volt\
age-controlled filters have a “tracking” switch for a CV input where the pitch of th\
e filter’s corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding chan\
ge 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 freq\
uency 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."),
quiz::Quiz("Transducer","A device that converts energy from one medium to another. T\
ransducers are prevalent throughout the equipment in a recording studio."),
quiz::Quiz("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, o\
r the strike of a drum."),
quiz::Quiz("Transistor Ladder Filter","This term is often used to describe the desig\
n of the much-loved Moog low-pass filter, which is still held up by many as being th\
e gold standard in low pass filter sound. Moog actually received a patent for this d\
esign (it has since expired); many of their competitors either outright copied it or\
 did their best to emulate it."),
quiz::Quiz("Transport","1) The portion of a tape machine that moves the tape from th\
e 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, emulatin\
g the functions of a tape machine transport."),
quiz::Quiz("Transpose","In the simplest terms, to transpose the pitch of a musical l\
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ine is to shift it up or down by a fixed number of semitones or octaves. This is som\
etimes referred to as “chromatic” transposition. A more sophisticated variation is “\
scalar” transposition where each note is shifted by a set number of scale steps; thi\
s differs from chromatic transposition because some scales may have differing number\
s 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 t\
hat 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 instrumen\
t, 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)."),
quiz::Quiz("Trap","1) A filter designed to reject audio signals at certain frequenci\
es. 2) An object designed with acoustically absorptive material, placed into walls t\
o reduce low frequency reflections in the room (also called “bass trap”). 3) Another\
 word for a drum set (as in “trap set”)."),
quiz::Quiz("Tremolo","This is the effect of varying the amplitude (loudness) of a no\
te. A way to create this effect on a modular synth is to patch a low frequency oscil\
lator (LFO) to one of the control voltage inputs on an amplifier. Tremolo is differe\
nt 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 i\
n a violin tremolo) or by rapid shifts in amplitude."),
quiz::Quiz("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 out\
put 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 s\
moother sine wave."),
quiz::Quiz("Triangular Wave","A harmonically rich waveform that appears triangular i\
n shape when depicted graphically, due to a combination of the presence of odd harmo\
nics and rapid rolloff."),
quiz::Quiz("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 fal\
ling back to 0 volts. It is often used to start or “trigger” the playback of a percu\
ssion 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 su\
bdivision of a note) starts. A trigger usually has a fixed duration, compared to a g\
ate 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\
."),
quiz::Quiz("Trim / Trim Control","A device that reduces or increases the signal stre\
ngth in an amplifier, often over a restricted range. Often used interchangeably with\
 gain, but usually referring to fine-tuning signal strength, rather than merely ampl\
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ifying it."),
quiz::Quiz("Truncation","1) The shortening of an audio signal, sample or song, typic\
ally 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."),
quiz::Quiz("Tune","The act of adjusting the pitch of a synthesizer’s oscillator (the\
 main pitch-generating element) to match another oscillator, instrument, or referenc\
e is known as tuning it."),
quiz::Quiz("Tuning Fork","A metal fork with two prongs that vibrate with a fairly pu\
re tone of one frequency when the fork is struck."),
quiz::Quiz("Turntable","A device to support and rotate a phonograph record during pl\
ayback."),
quiz::Quiz("Tweeter","A speaker designed to reproduce only the higher frequencies of\
 the sound."),
quiz::Quiz("Two Quadrant Multiplier","A two-quadrant multiplier performs a simple ve\
rsion of amplitude modulation (AM), where that varies the amplitude or loudness of o\
ne signal known as the carrier (typically an audio signal, swinging both above and b\
elow 0 volts) with a second signal called the modulator. In the typical amplitude mo\
dulation (AM) scenario, a low frequency oscillator with a positive voltage (say, bet\
ween 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into th\
e 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 p\
atching 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 wave\
form. (The case where the modulator's negative as well as positive excursions are us\
ed is referred to as a four quadrant multiplier.) "),
quiz::Quiz("Two-Way Speaker","A speaker system with separate speakers to reproduce t\
he lower frequencies (woofer) and the higher frequencies (tweeter)."),
quiz::Quiz("TZFM","TZFM is the abbreviation for Through-Zero Frequency Modulation. T\
hink 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 wavefo\
rm output of the modulator rises above zero volts, it is added to the normal pitch c\
ontrol voltage for the carrier, and the pitch of the carrier goes up. As the wavefor\
m output of the modulator goes below zero, it is subtracted from the normal pitch co\
ntrol voltage, and the pitch goes down. But what happens if the result of subtractin\
g 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 st\
art playing backwards – in essence, a negative frequency. This generally produces a \
more pleasing result, and is a desirable characteristic for an oscillator."),
quiz::Quiz("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 equ\
aling 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 c\
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m) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometime\
s referred to as MU for Moog Unit)."),
quiz::Quiz("Unbalanced Audio","Most audio signals are passed around on cables with t\
wo wires: one for the voltage that represents the audio vibrations, and one for grou\
nd. This arrangement is often referred to as unbalanced audio."),
quiz::Quiz("Unbalanced Cable","A cable with two conductors (a signal wire and a grou\
nd wire) and connectors on each end. Unbalanced cables are often susceptible to elec\
tromagnetic interference and noise. Examples of unbalanced cables are guitar/instrum\
ent cables (also called tip-sleeve or TS cables) and RCA cables."),
quiz::Quiz("Unidirectional Pattern","A microphone pick-up pattern which is more sens\
itive to sound arriving from one direction than from any other."),
quiz::Quiz("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 t\
he output of an envelope generator – only vary between 0 volts and some maximum posi\
tive voltage; this is referred to as unipolar."),
quiz::Quiz("Unison","Several performers, instruments or sound sources that are sound\
ing at the same time and with the same pitch."),
quiz::Quiz("Unity Gain","The scenario in which there is no increase or decrease in s\
ignal strength at the output of an amplifier or device compared to the signal streng\
th at the input (typically described as 0 dB)."),
quiz::Quiz("Unity","Usually used in the phrase “unity gain” this mean a signal keeps\
 the exact same level from input to output."),
quiz::Quiz("Vacuum Tube","A diode, a glass tube with the gases removed, through whic\
h electrical current can flow. In audio, vacuum tubes are used in amplifiers, oscill\
ators, and other analog devices."),
quiz::Quiz("Vamp and Fade","A method of ending the recording of a song where the mus\
ic has a repeating part and the engineer reduces volume until the music fades out.")\
,
quiz::Quiz("Vamp","A part of a song or chord progression that is repeated, usually a\
t the end of the song, and usually the chorus or part of the chorus."),
quiz::Quiz("Vari-Speed","A control on a tape machine that changes the play speed."),
quiz::Quiz("Variable-D","A trademarked, patented technology of ElectroVoice in its m\
icrophone 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 sensi\
tivity to higher frequencies the further they are placed from the microphone’s diaph\
ragm."),
quiz::Quiz("VCA Automation","A system of mix automation in some mixing consoles in w\
hich sound levels or other functions are altered through the use of voltage controll\
ed amplifiers."),
quiz::Quiz("VCA Group","Several VCA faders that are fed control voltages from a grou\
p master slide. A feature in higher-end mixing boards that enables the engineer to c\
ontrol groupings of independent signals by a single fader that uses VCA to adjust th\
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e voltage sent to each channel."),
quiz::Quiz("Velocity Message","In synthesizers and keyboard controllers, a MIDI mess\
age that transmits data on how hard the key was struck. Velocity messages can be use\
d to transmit volume information, as well as triggering different samples on a multi\
-sampled instrument patch."),
quiz::Quiz("Velocity Microphone","See “Pressure-Gradient Microphone.”"),
quiz::Quiz("Velocity Sensitive","(Also called “Touch Sensitive“) A feature on a MIDI\
 instrument such as a keyboard that transmits a MIDI velocity message depending on h\
ow hard the key is struck."),
quiz::Quiz("Vibrato","A smooth and repeated changing of the pitch up and down from t\
he regular musical pitch, often done by singers or performed by string and wind play\
ers."),
quiz::Quiz("Virtual Instrument","(Also called Software Instrument) One of a number o\
f software-based synthesizers, samplers or sound samples that are stored and accesse\
d via computer and performed by an external MIDI controller, rather than in a standa\
lone synthesizer or module. Because of the wide versatility available from these ins\
truments, a growing number of composers and electronic musicians are working with vi\
rtual instruments that can be stored in hard drives, rather than purchasing stacks o\
f keyboards and modules."),
quiz::Quiz("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 ins\
truments into the vocal microphone, and also reduces natural ambience and reverberat\
ion in the vocal recording."),
quiz::Quiz("Vocoder","An audio processing device effects device or plug-in that anal\
yzes the characteristics of an audio signal and uses them to affect another synthesi\
zed signal. Primarily developed for the purpose of producing synthesized voice effec\
ts from human speech, a vocoder creates the characteristic robotic vocal sound or th\
e “human synthesizer” effect that makes it sound like the synth is speaking or singi\
ng words."),
quiz::Quiz("Voice Over","The recording of vocal announcements or narration over a be\
d of music in video, film or commercials."),
quiz::Quiz("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 th\
at 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 s\
pecific sound patch available on the synth."),
quiz::Quiz("Volatile Memory","Computer memory whose data will will be lost when the \
computer is turned off. RAM (Random Access Memory) is the most common form of volati\
le memory."),
quiz::Quiz("Voltage Controlled Amplifier (VCA)","An amplifier whose gain level is af\
fected by an external voltage being sent to it. VCAs are commonly used in synthesize\
rs, signal processors, and as a means of automation for some mixing consoles."),
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quiz::Quiz("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."),
quiz::Quiz("Voltage Controlled Oscillator (VCO)","An oscillator whose frequencies ar\
e modified by voltage input. Most commonly found in synthesizers."),
quiz::Quiz("Voltage","The difference in electrical force or pressure (“potential”) b\
etween two objects, causing a flow of electric current between them."),
quiz::Quiz("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 Mete\
r. (See also “VU Meter.”)"),
quiz::Quiz("Volume","A common, non-technical term that either refers to sound pressu\
re level (which we hear as loudness), or to audio voltage level."),
quiz::Quiz("Vox","A Latin word meaning “voice,” often used as an abbreviation for tr\
ack logs in the studio."),
quiz::Quiz("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.\
"),
quiz::Quiz("Watt","Unit of electrical power."),
quiz::Quiz("Wave","This is the pattern of vibrations – up and down fluctuations in v\
oltage – output by an oscillator. Different patterns generate different sounds."),
quiz::Quiz("Wavefolder","A wavefolder is a very specific design of waveshaper that u\
ses a comparator and some other circuitry. What they do is look to see if the wave g\
oes 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 porti\
on of the wave back upon itself, creating more high harmonics and interesting spectr\
a in the process."),
quiz::Quiz("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 wa\
ve, showing the wave’s oscillations above and below the zero line."),
quiz::Quiz("Wavelength","The physical length of one cycle of a wave, measured in fee\
t, inches, etc. The longer the wavelength of a sound wave, the lower its frequency; \
the shorter the wavelength, the higher the frequency."),
quiz::Quiz("Waveshaper","It would be a bit obvious to say “a circuit that changes th\
e 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 ou\
tgoing sine wave, or to add tube-like soft clipping to the peaks and transients of w\
aves. Many waveshapers are simply intended to mangle (er, add higher harmonics to) w\
aveforms in interesting ways, creating noisier (er, more complex and bright) harmoni\
c spectra to create new sounds."),
quiz::Quiz("Wavetable","This term can have two related but slightly different meanin\
gs. 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 \
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describes one cycle of a wave, and therefore is often called a wavetable. Many digit\
al 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 crossf\
ading between them (what most digital wavetable oscillators today do). Some people r\
efer to each table as a “wave” and a set of individual waves as a wavetable."),
quiz::Quiz("Weighting","An equalization curve used in audio tests that compensates f\
or the Fletcher Munson Curve at various levels. (See also “Fletcher-Munson Curves.”)\
"),
quiz::Quiz("West Coast Synthesis","The so-called \"West Coast\" approach to synthesi\
s - traditionally associated with companies such as Buchla and Serge - is often base\
d 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 suc\
h as an AD or AR (often called slope generators) rather than four-stage ADSRs, as we\
ll 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 w\
hat's known as a Low Pass Gate (LPG). The West Coast approach also embraces non-trad\
itional controllers, such as touch plates and the such. Today it's common to mix bot\
h East Coast and West Coast approaches in the same system."),
quiz::Quiz("wet sound","Sometimes people will say a filter has a “wet” sound. This u\
sually refers to a fewer-than-4-pole filter sound - often low or bandpass - with res\
onance 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."),
quiz::Quiz("Wet","A sound with effects (such as reverb) mixed is referred to as \"we\
t\"; a sound with no effects is referred to as \"dry.\" Effects units or mixers ofte\
n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
d and the fully-effected sound. Refers to a signal that has the full amount of an ef\
fect (like reverb) applied to it, as opposed to “dry,” which refers to the un-effect\
ed sound. Many times, the preferred sound in mixing will be a blend of wet and dry s\
ignals. (See also “Dry.”)"),
quiz::Quiz("White Noise","Noise is a random signal that does not have a distinct pit\
ch, 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 differe\
nt 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 contain\
ing an equal spread of energy across all audible frequencies. Like pink noise, engin\
eers often send a white noise signal through audio equipment for tuning and calibrat\
ion purposes, or in EQ-ing a live audio space. (See also “Pink Noise.”)"),
quiz::Quiz("Whole Step","A change in pitch equivalent to two half steps, or the diff\
erence in pitch between two piano keys."),
quiz::Quiz("Wild Sound","In film and video, audio that is recorded separately from t\
```

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he visual that may be added to the audio track later, and does not need to be synchr\
onized with the picture."),
quiz::Quiz("Wind Controller","A device that is played like a wind instrument to cont\
rol a synthesizer, module or virtual instrument via MIDI signals, as opposed to a ke\
yboard controller."),
quiz::Quiz("Windscreen","A covering that fits over a microphone to reduce the excess\
ive noise resulting from wind blowing into the mic. Typically used for recording in \
outdoor locations."),
quiz::Quiz("Wireless Microphone","A microphone that transmits its signal over an FM \
frequency to a receiver offstage, rather than traveling over an audio cable."),
quiz::Quiz("Woofer","A speaker that is designed to reproduce bass frequencies only."\
),
quiz::Quiz("Write Mode","A mode of operation in an automated console where the engin\
eer is in control of channel gain and the computer is recording the gain changes ove\
r time."),
quiz::Quiz("XLR Cable","A balanced microphone cable utilizing XLR connectors. (See a\
lso “XLR Connector.”)"),
quiz::Quiz("XLR Connector","A balanced cable connector consisting of 3 or 7 pins, mo\
st commonly used in microphone cables."),
quiz::Quiz("XY Miking","A coincident stereo microphone placement technique in which \
two cardioid microphones are placed with their heads toward each other at a 90-degre\
e angle, and as close together as possible. (See also “Coincident Miking.”)"),
quiz::Quiz("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 compo\
nents."),
quiz::Quiz("Zenith","In analog tape recording, refers to the tilt of the tape head i\
n the direction perpendicular to the tape travel."),
quiz::Quiz("Zero-Order Hold (ZOH)","Refers to the mathematical expression of the sig\
nal processing done by a conventional digital-to-analog converter (DAC)."),
};


int main()
{
        std::random_device rd;
    std::mt19937 gen(rd());
    std::uniform_int_distribution<> distria(1, 4);
    std::uniform_int_distribution<> distrib(0, game.size()-1);
        std::shuffle(std::begin(game), std::end(game), std::default_random_engine());
        std::vector<std::string> answers;
        std::string question;
        uint32_t n;
        uint8_t correct;
```

```
        uint32_t score=0;
        uint32_t tqs=0;

        for (uint32_t ctr=0;ctr<game.size();++ctr) {
                answers.clear();
                correct=distria(gen);
                for (uint8_t i=1;i<=4;++i) {
                        if (i == correct) {
                                answers.push_back(game[ctr].getQ());
                                question=game[ctr].getA();
                        } else {
                                answers.push_back(game[distrib(gen)].getQ());
                        }
                }
                std::cout << "\33c\e[3J";
                if (tqs != 0) {
                        std::cout << "[QUESTIONS: " << tqs << " / " << game.size() << " SCORE: " <<
 << "]\n";
                }
                std::cout << "Question #" << tqs+1 << ": " << question << "\n\n";
                std::cout << "Answer #1.\n" << answers[0] << "\n\n";
                std::cout << "Answer #2.\n" << answers[1] << "\n\n";
                std::cout << "Answer #3.\n" << answers[2] << "\n\n";
                std::cout << "Answer #4.\n" << answers[3] << "\n\n";
                std::cout << "What answer is correct (q=quit)? ";
                std::cin >> n;
                if (n == 0) {
                        break;
                } else if (n == correct) {
                        score++;
                }
                tqs++;
                std::cout << n << " is the answer you gave. And the correct answer is: " << correc\
t << '\n';
        }

         std::cout << "\33c\e[3J";
         if (tqs != 0) {
                 std::cout << "[QUESTIONS: " << tqs << " / " << game.size() << " SCORE: " << score\
 << "]\n";
                 std::cout << "[" << ((double(score)/double(tqs))*100.0) << "% correct answers.]\n\
";
         }
```

```
        return 0;
}
```

quiz2.cpp - 338681 bytes.

```
// compile: clang++ -std=c++20 quiz2.cpp -o quiz2
#include <iostream>
#include <fstream>
#include <vector>
#include <algorithm>
#include <random>

namespace quiz
{
        class Quiz
        {
        public:
                Quiz(const std::string &q, const std::string &a) {_a=a;_q=q;}
                virtual ~Quiz() {}
                std::string getQ() {return _q;}
                std::string getA() {return _a;}
        private:
                std::string _q;
                std::string _a;
        };
}

std::vector<quiz::Quiz> game{
quiz::Quiz("0-5v","Denotes a range of 0 to 5 volts, which is common for gates, trigg\
ers, and modulation control voltages in modular synthesizers. Gates and triggers - w\
hich initiate events such as new notes - typically rise from 0v to 5v (0 to 10v is a\
lso common), with roughly the middle of that onset starting the event. Gates are con\
sidered high when held at 5v (or 10v), and then low when they return to 0v."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 6 decibels weaker for each octave further away you get from the cutoff fr\
equency. A 6dB/octave filter is often referred to as a “one pole” filter (as each po\
le of a filter’s design results in 6dB of attenuation), and has a relatively weak ef\
fect on the signal going through it. Low Pass Gates (LPGs) typically - but not alway\
s - use 1 pole low pass filters, reducing the strength of higher harmonics by 6 deci\
bels for every octave above its cutoff frequency."),
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quiz::Quiz("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 division\
s."),
quiz::Quiz("1 v/oct","The most common standard for controlling pitch in a modular sy\
nthesizer. Under the system, increasing the voltage going into a VCO (Voltage Contro\
lled Oscillator) 1 volt - say, from 0.5v to 1.5v - would raise its pitch by one octa\
ve."),
quiz::Quiz("1.2 v/oct","Buchla compatible synths have standardized on the 1.2 volt p\
er 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 f\
or a change in pitch of 1 semitone."),
quiz::Quiz("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."),
quiz::Quiz("1/8”","Often used to incorrectly describe the connector size commonly us\
ed in Eurorack format modules, as well as Buchla audio signals. In fact, Eurorack mo\
dules 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 phy\
sically 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.")\
,
quiz::Quiz("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 rang\
e may be varied to change the depth of their effect, so don’t get too hung up on spe\
cific voltage ranges. Pay more attention to whether they vary between 0v and some va\
lue, or swing in roughly equal amounts both above and below 0v (as 10vpp does)."),
quiz::Quiz("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 cutof\
f frequency. A 12dB/octave filter is often referred to as a “two pole” filter (as ea\
ch 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 soun\
ds when compared to those with 4-pole instruments."),
quiz::Quiz("16’","Sometimes seen on octave selector switches on oscillators. It refe\
rs to the length of an organ pipe. Longer pipes = lower pitches; 16’ is in the mid-b\
ass 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."),
quiz::Quiz("18 dB/oct","This format of numbers and abbreviations (dB/oct = decibels \
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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 cutof\
f frequency. It is often used a coded shorthand for when someone wants to refer to a\
cid-type bass lines from a TB-303 without mentioning the instrument by name."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 12 decibels weaker for each octave further away you get from the cutoff f\
requency. 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 Obe\
rheim instruments often featured 2-pole filters, often resulting in brighter sounds \
when compared to those with 4-pole instruments."),
quiz::Quiz("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 th\
at the modules are attached to."),
quiz::Quiz("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 cutof\
f frequency. This design is often used in vintage Moog and Roland synths. 4-pole fil\
ters are often associated with subjectively fatter, more “round” sounds than 2-pole \
filters – but generalizations are always dangerous."),
quiz::Quiz("24 ppqn","A common master clock division used in MIDI, DIN sync, and oth\
er systems common to electronic music and synthesizers. It means internally, 24 subd\
ivisions 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 (÷6), eighth notes\
 (÷12), eight note triplets (÷8), etc."),
quiz::Quiz("2’","Sometimes seen on octave selector switches for oscillators. It refe\
rs 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."),
quiz::Quiz("3 mm","A common screw thread size used to mount Eurorack modules. This s\
ize is most common when using module mounting rails that have been pre-drilled."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 18 decibels weaker for each octave further away you get from the cutoff f\
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requency. 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."),
quiz::Quiz("3.5 mm","The standard connector size used for jacks and cables in Eurora\
ck format modular synthesizers. Note that this is slightly larger that 1/8”."),
quiz::Quiz("303","The TB-303 Bass Line by Roland became a cult favorite in Acid Hous\
e 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;"),
quiz::Quiz("32’","Sometimes seen on octave selector switches on oscillators. It refe\
rs to the length of an organ pipe. Longer pipes = lower pitches; 32’ is the lowest s\
etting 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 h\
igher; etc."),
quiz::Quiz("3U","Refers to modules that are 3 rack units (U) high – the Eurorack sta\
ndard, which is by far the most common modular format today, even though it’s one of\
 the youngest formats."),
quiz::Quiz("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 fi\
lter typically has a cutoff or corner frequency it is tuned to. It then reduces (fil\
ters) the frequency spectrum of a signal going through it so that its loudness is mu\
ltiples of 24 decibels weaker for each octave further away you get from the cutoff f\
requency. This design is often used in vintage Moog and Roland synths. 4-pole filter\
s are often associated with subjectively fatter, more “round” sounds than 2-pole fil\
ters – but generalizations are always dangerous."),
quiz::Quiz("4-40","A screw thread size occasionally used to mount Eurorack modules. \
This size is used by Pittsburgh Modular for their cases, for example."),
quiz::Quiz("4U","Refers to modules that are 4U (rack units) high – namely, Buchla an\
d 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.)\
"),
quiz::Quiz("4’","Sometimes seen on octave selector switches on oscillators. It refer\
s 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.")\
,
quiz::Quiz("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 fo\
llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
d is both historical and physically large, some users “5U” as a badge of honor that \
they’re traditional and cool. (And the are.) There was also a briefly popular 5U for\
mat from MOTM that used a different width and power connection. It has since been di\
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scontinued, but there are still diehard MOTM format users today."),
quiz::Quiz("6 dB/oct","This format of numbers and abbreviations (dB/oct = decibels p\
er 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 (f\
ilters) 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 alw\
ays – use 1 pole low pass filters, reducing the strength of higher harmonics by 6 de\
cibels for every octave above its cutoff frequency."),
quiz::Quiz("808","The TR-808 Rhythm Composer by Roland created all of its sounds usi\
ng analog circuitry. When it first came out, it was not well loved, as the analog so\
unds weren’t realistic enough. But later, music styles such as House and Hip-Hop ado\
pted 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 s\
nappy attack."),
quiz::Quiz("8’","Sometimes seen on octave selector switches on oscillators. It refer\
s to the length of an organ pipe. Shorter pipes = higher pitches; 8’ is typically as\
sociated 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."),
quiz::Quiz("909","The TR-909 Rhythm Composer was the follow-up to Roland’s now-rever\
ed 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-lik\
e sounds, this is the instrument it is referencing."),
quiz::Quiz("A-440","This is the frequency in hertz (cycles per second) of the A abov\
e Middle C. It is often used as a tuning reference."),
quiz::Quiz("A/B Technique","A stereo microphone placement technique in which two car\
dioid or omnidirectional microphones are spaced somewhere between 3-10 feet apart fr\
om each other (depending on the size of the sound source) to create a left/right ste\
reo image. Also known as Spaced Pair."),
quiz::Quiz("A/D","Abbreviation of Analog-to-Digital Conversion, the conversion of a \
quantity that has continuous changes (like electrical signals) into numbers that app\
roximate those changes (i.e., computer data)."),
quiz::Quiz("Absolute Phase","This term describes a perfect polarity between an origi\
nal signal (into the microphone) and the reproduced signal (through the speaker). Wh\
en positive pressure exerted upon the microphone is translated as positive pressure \
to the loudspeaker, the two are in “absolute phase.”."),
quiz::Quiz("Absorption","In acoustics, absorption is what happens when sound waves a\
re absorbed by a surface, as opposed to bouncing off the surface (reflection). Absor\
ptive materials in a control room, for example, tend to “deaden” the sound of the ro\
om because the sound energy is absorbed rather than reflected. (See also “Reflection\
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.")"),
quiz::Quiz("AC Coupled","An AC coupled input attempts to remove any constant DC volt\
age going through it. This is useful if have an audio signal (such as the output of \
an oscillator) which is AC in nature, and you want to remove any accidental DC offse\
t 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 A\
C coupling is a high pass filter that is attempting to remove very low frequency com\
ponents."),
quiz::Quiz("AC","Alternating Current - The type of electrical current found in stand\
ard electrical outlets and studio signals running through audio lines. In AC, the cu\
rrent "alternates" directions, flowing back and forth through the circuit. In modula\
r terms, AC refers to a voltage that alternates between positive and negative values\
 – such as the output of an oscillator."),
quiz::Quiz("Accelerometer","A device that measures the acceleration to which it is s\
ubjected and creates an electric signal to match it. In music and audio, acceleromet\
ers are found in such things as microphones and guitar pickups."),
quiz::Quiz("Acorn Tube","Named for its acorn-like shape, an acorn tube is a small va\
cuum tube used in ultra high frequency (UHF) electronics such as tube amplifiers."),
quiz::Quiz("Acoustic Amplifier","The part of a musical instrument that vibrates in r\
esponse to the initial vibration of the instrument, causing the surrounding air to m\
ove more efficiently and making the sound louder. For example: the body of an acoust\
ic guitar, the bell of a horn, a drum's shell, and the wooden soundboard of a piano.\
"),
quiz::Quiz("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."),
quiz::Quiz("Acoustics","The science of the sound—more specifically, the science of t\
he properties and behavior of sound waves. A good understanding of acoustics is esse\
ntial to audio engineering and studio design."),
quiz::Quiz("Active Device","A component that is designed with the ability to control\
 electrical current (as opposed to a "Passive Device"). In the recording studio, act\
ive devices are generally components that include an amplifier. (See also "Passive D\
evice.")"),
quiz::Quiz("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 desti\
nations. An active or buffered multiple is one that includes a buffer circuit betwee\
n the input and output, making sure the signal does not lose its strength or integri\
ty 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 buil\
t into their outputs, and can drive multiple modules without issue. But if you try t\
o use a passive mult to connect to, say, three oscillators, and you realize the trac\
```

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king 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."),
quiz::Quiz("Actuator","The part of a switch that causes change of the contact connec\
tions (e.g., toggle, pushbutton, or rocker)."),
quiz::Quiz("AD","Shorthand for a two-stage Attack/Decay envelope. This simple envelo\
pe shape raises from 0 volts to its maximum level (typically 5, 8, or perhaps 10 vol\
ts) 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 enve\
lope: After finishing the Attack stage, it holds at the maximum level for a specifie\
d amount of time (in contrast to an AR envelope, which holds at the maximum level fo\
r as long as the note on gate is high), and then decays back to zero. I have heard t\
here are some envelopes that a hybrid of AHD and AR in that they hold the maximum le\
vel for either the defined Hold time or the as long as the incoming gate is high;"),
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. A\
dditive synthesis is a technique that gives you direct control over each of those co\
mponent harmonics, allowing you to directly dial in the mix you want. As immediate a\
nd intuitive as that sounds on paper (or on screen), in reality it takes a lot of wo\
rk to craft the correct mixture to recreate another sound, especially since the stre\
ngth of each harmonic usually varies over time. Additive synthesis oscillators are r\
elatively rare in modular synths; two examples are the Verbos Harmonic Oscillator an\
d the Make Noise tELHARMONIC."),
quiz::Quiz("ADSR","An envelope generator with four stages: Attack, Decay, Sustain, a\
nd 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 Controlle\
d Amplifier, or the lowest frequency when connected to a voltage controlled filter o\
r oscillator) and raises to the maximum voltage it can output (typically 5 to 10 vol\
ts depending on system; it can often be set with an output level control) over a tim\
e set by the Attack control. Once it reaches that level, the output voltage immediat\
ely starts dropping to speed set by the Decay control it until it reaches the voltag\
e set by the Sustain control. If the input gate is still active, this level is maint\
ained 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."),
quiz::Quiz("AES","Audio Engineering Society."),
quiz::Quiz("AES3","(sometimes called AES/EBU) A digital audio transfer standard deve\
loped by the Audio Engineering Society and the European Broadcasting Union for carry\
ing dual-channel digital audio data between devices. AES3 is the protocol behind XLR\
 cables, as well as RCA and S/PDIF cables."),
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 re\
furbished B-stock units come up for sale every now and then. The expansion modules w\
ere, to the best of my knowledge, never released (at least not widely)."),
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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 contro\
l 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 wi\
der. Monophonic aftertouch measures one pressure value for the entire keyboard, rega\
rdless of which key(s) you are pressing; polyphonic aftertouch produces a signal for\
 each individual key. Important trivia: Touch plate keyboards actually measure the s\
urface area of the skin touching them rather than pressure or force - so you can inc\
rease or decrease the aftertouch amount by rolling between the tip and length of you\
r finger."),
quiz::Quiz("AHDSR","Attack, Hold, Decay, Sustain, and Release. This is a slightly fa\
ncier ADSR envelope that holds the voltage typically at its maximum value for a spec\
ified time after the attack is done rising and before the decay starts falling."),
quiz::Quiz("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 s\
ample rate is an audible pitch, you will also hear a mirror image of the sound’s har\
monic content reproduced started at that half-sample-rate pivot (unless some excelle\
nt filtering has taken place). (See also “Nyquist Frequency.”)."),
quiz::Quiz("Alternating Current (or AC)","The type of electrical current found in st\
andard electrical outlets and studio signals running through audio lines. In AC, the\
 current “alternates” directions, flowing back and forth through the circuit."),
quiz::Quiz("AM","Amplitude Modulation (AM) is the name given the 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 oscil\
lator with a positive voltage (say, between 0v and 5v, or maybe something smaller su\
ch as between 1v and 2v) is fed into the control input of a voltage controlled ampli\
fier to add vibrato to an audio signal passing through it. Technically, this is know\
n as a two-quadrant multiplier or modulator, as any negative swings in the modulatio\
n signal are ignored; when patching tremolo, you may need to make sure an offset vol\
tage is being added to your LFO to make sure the sound doesn’t cut out on the lower \
excursions of the LFO’s waveform."),
quiz::Quiz("Ambience","In most cases, this refers to the “atmosphere” of a certain p\
lace, like a restaurant. But in recording, it refers to the part of the sound that c\
omes from the surrounding environment rather than directly from the sound source. Fo\
r example, the sound waves coming into your ears from a cello being played are comin\
g directly from the source, but the sound of the same cello coming to you after boun\
cing off the back wall is ambient sound."),
quiz::Quiz("Ambient Field","The area away from the sound source where the reverberat\
ion is louder than the direct sound."),
quiz::Quiz("Ambient Miking","This refers to placing a microphone in the ambient fiel\
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d of a room to record the ambient reverberations of the sound. The recording enginee\
r often does this in addition to direct micing of the instrument(s) to create a blen\
d or mix of direct and reverberant sound in the recording."),
quiz::Quiz("Amp","An abbreviation for “Amplifier,” “Amplitude” or “Ampere,” dependin\
g on context."),
quiz::Quiz("Ampere","The unit of measure for electrical current, abbreviated Amp."),
quiz::Quiz("Amplifier","A device that increases the level or amplitude of an electri\
cal signal, making the resulting sound louder."),
quiz::Quiz("Amplitude Modulation","Amplitude Modulation (AM) is the name given the t\
o the technique of varying the amplitude or loudness of one signal known as the carr\
ier (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 l\
ow frequency oscillator with a positive voltage (say, between 0v and 5v, or maybe so\
mething smaller such as between 1v and 2v) is fed into the control input of a voltag\
e controlled amplifier to add vibrato to an audio signal passing through it. Technic\
ally, this is known as a two-quadrant multiplier or modulator, as any negative swing\
s 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."),
quiz::Quiz("Amplitude","The height of a waveform above or below the zero line. In au\
dio, this usually translates to the signal strength or the volume of the sound."),
quiz::Quiz("Analog Recording","A recording of the continuous changes of an audio wav\
eform. The most common example of analog recording in a recording studio is recordin\
g on reel-to-reel magnetic tape."),
quiz::Quiz("Analog Shift Register","An Analog Shift Register (ASR) is a cross betwee\
n a Sample & Hold module and a Bucket Brigade Delay (assuming you already know how t\
hose 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 aga\
in with this new voltage presented at the first output, while the original voltage i\
s now moved to a second output. This game of \"telephone\" is passed along for as ma\
ny stages as the ASR has – traditionally three or four."),
quiz::Quiz("Analog To Digital Converter (A/D; or ADC)","A device that translates a c\
ontinuously changing signal (analog) into numeric values that approximate those chan\
ges (digital). In audio recording, this refers to converting recorded sound from ele\
ctrical voltages to computerized data."),
quiz::Quiz("Analog","The term analog implies a signal is continuously variable, comp\
ared 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."),
quiz::Quiz("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 o\
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utput: The output would be low if both inputs were high, but otherwise would be high\
."),
quiz::Quiz("AR","The two-stage Attack/Release envelope raises from 0 volts to its ma\
ximum level (usually 5, 8, or maybe even 10 volts) at a rate set by its Attack param\
eter, and then stays at that value for as long as the gate signal fed into the envel\
ope 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 g\
ate signal.)"),
quiz::Quiz("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 betwee\
n your keyboard and sound module – makes it easier for you to play arpeggios: You ju\
st hold down the notes of the chord, and it automatically plays the notes one at a t\
ime, over and over again, like a step sequencer you can program on the fly just by h\
olding 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 t\
his even after you’ve released the keys."),
quiz::Quiz("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 i\
nitially 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 v\
oltage presented at the first output, while the original voltage is now moved to a s\
econd output. This game of \"telephone\" is passed along for as many stages as the A\
SR has – traditionally three or four."),
quiz::Quiz("Attack/Decay/Sustain/Release","An envelope generator with four stages: A\
ttack, Decay, Sustain, and Release. When this envelope generator receives a gate inp\
ut, it typically starts at 0 volts (which is the equivalent of silence when connecte\
d to a Voltage Controlled Amplifier, or the lowest frequency when connected to a vol\
tage controlled filter or oscillator) and raises to the maximum voltage it can outpu\
t (typically 5 to 10 volts depending on system; it can often be set with an output l\
evel 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 it unt\
il it reaches the voltage set by the Sustain control. If the input gate is still act\
ive, this level is maintained until the gate goes back to 0 volts (usually because y\
ou released the key on a controlling keyboard, etc.). At that time, the output volta\
ge then starts dropping back to 0 volts at the rate set by the Release control."),
quiz::Quiz("Attack/Decay","Shorthand for a two-stage Attack/Decay envelope. This sim\
ple envelope shape raises from 0 volts to its maximum level (typically 5, 8, or perh\
aps 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 th\
e AHD envelope: After finishing the Attack stage, it holds at the maximum level for \
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a specified amount of time (in contrast to an AR envelope, which holds at the maximu\
m level for as long as the note on gate is high), and then decays back to zero. I ha\
ve 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 i\
s high;"),
quiz::Quiz("Attack/Hold/Decay/Sustain/Release","This is a slightly fancier ADSR enve\
lope that holds the voltage typically at its maximum value for a specified time afte\
r the attack is done rising and before the decay starts falling."),
quiz::Quiz("Attack/Release","The two-stage Attack/Release envelope raises from 0 vol\
ts 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 in\
to the envelope generator stays high. Then when the gate signal goes back to zero, t\
he 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 y\
ou specify a fixed time for the level to stay at its maximum, rather than pay attent\
ion to the gate signal.)"),
quiz::Quiz("Attack","This usually refers to the first stage of an envelope that occu\
rs 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."),
quiz::Quiz("Attenuation","The reduction of electrical or acoustic signal strength. I\
n audio, attenuation is measured in decibels (dB) and is typically heard as a reduct\
ion in volume. Sound waves traveling through the air naturally attenuate as they tra\
vel away from the source of the sound. Engineers also purposefully attenuate signals\
 in the studio through gain controls or pads to prevent overload."),
quiz::Quiz("Attenuator","A control that can reduce the strength of a signal or volta\
ge going through it."),
quiz::Quiz("Attenuverter","A special version of an attenuator that can also invert t\
he polarity of the signal or voltage going through it. Most attenuverters use pass t\
hrough no signal at their center position; as you turn them clockwise, you turn up t\
he 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 po\
larity switch added on."),
quiz::Quiz("Audio Frequency Generator","The AFG (Audio Frequency Generator) is a ver\
y full-featured analog oscillator released by Livewire Electronics. It has since bee\
n discontinued, but refurbished B-stock units come up for sale every now and then. T\
he expansion modules were, to the best of my knowledge, never released (at least not\
 widely)."),
quiz::Quiz("Audio","In its broadest sense, audio is the range of frequencies we huma\
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\
."),
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quiz::Quiz("Automatic Dialogue Replacement (ADR)","The process of re-recording dialo\
gue for film in a controlled environment after the film is shot, for the purpose of \
replacing poorly recorded dialogue."),
quiz::Quiz("Automatic Gain Control","A compressor with a long release time, which is\
 used to keep the volume of the audio at a consistent level."),
quiz::Quiz("Automation","Programming certain changes to occur automatically during r\
ecording and/or playback. In the studio, engineers use automation on their consoles \
or computers so various parameters will change automatically at different times duri\
ng multitrack recording and playback. This pre-programming feature makes it easier t\
o create those changes than attempting to perform them all manually in real time."),
quiz::Quiz("Auxiliary Equipment","External signal processing devices that work along\
side the mixing console to modify the signal."),
quiz::Quiz("Auxiliary Return","(Abbreviated Aux Return or Return) The input on a con\
sole or DAW that returns the effected signal sent through the auxiliary send back in\
to the channel mix."),
quiz::Quiz("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 equi\
pment 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 th\
rough the auxiliary return."),
quiz::Quiz("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 micr\
ophone 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."),
quiz::Quiz("Background Noise","Refers to either 1) The ambient noise in a room unrel\
ated to the instrument(s) or vocal(s) being recorded; or 2) The system noise unrelat\
ed to the recorded signal. (All electronics emit a level of noise.)"),
quiz::Quiz("Baffles","Sound absorbing panels that are used to prevent sound waves fr\
om entering or leaving a space."),
quiz::Quiz("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 eve\
n out the relative levels of audio signals in the channels of stereo recording."),
quiz::Quiz("Balanced Audio","This refers to a system where three wires are used to c\
arry an audio signal: one is the ground (the 0 volt reference), the second carries t\
he audio signal as it varies above and below 0v, and the third carries an inverted c\
opy of the audio signal that goes negative while the original is going positive. Bal\
anced audio usually implies a reference signal level of +4dB (higher than line level\
; still lower than most modular synths), although microphone signals - much weaker b\
y comparison, and therefore more susceptible to outside noise - are almost always ba\
lanced as well. Modular synths tend to use unbalanced audio for their internal signa\
ls. If you require a balanced output (or input), you need a special module that conv\
erts between balanced and unbalanced audio, plus does any necessary level matching."\
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),
quiz::Quiz("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 op\
posite 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."),
quiz::Quiz("Balanced Mixer","A circuit or device that generates the sum and differen\
ce frequencies of two input signals."),
quiz::Quiz("Balanced Modulator","Balanced or ring modulation is a special type of am\
plitude 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 a\
s the carrier. The modulator’s frequency is both added to and subtracted from the ca\
rrier’s frequency; the resulting harmonics replace the original carrier and modulato\
r."),
quiz::Quiz("Banana","An alternate type of connector (https://en.wikipedia.org/wiki/B\
anana_connector) used by 4U systems such as Buchla (control voltages) and Serge (bot\
h 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 refer\
ence. Banana connectors have an advantage in that they are usually “stackable” meani\
ng you can plug a one jack into the back of another, providing a passive multiple.")\
,
quiz::Quiz("Band Pass Filter","A device, circuit or plug-in that allows a narrow ban\
d of frequencies to pass through the circuit, rejecting or attenuating frequencies t\
hat are either higher or lower than the specified range."),
quiz::Quiz("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. T\
he exact opposite of a band pass filter."),
quiz::Quiz("Band Track","(Sometimes abbreviated “Track“) A mixdown of a song minus t\
he lead vocal and/or background vocals. In other words, a mixed track containing onl\
y the instrumental parts of the song."),
quiz::Quiz("Band","1) A range of frequencies, often identified by the center frequen\
cy of the range. 2) A group of musicians playing together."),
quiz::Quiz("Bandpass Filter","A bandpass filter (BPF) leaves the harmonics around th\
e 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 a\
ttenuated, based on the number of poles in the filter, with each pole equalling 6 de\
cibels of attenuation for each octave you get away from that center."),
quiz::Quiz("Bandwidth","In signal processing, bandwidth refers to the usable frequen\
cy range of a communication channel, measured by the difference between the device’s\
 highest and lowest usable frequencies."),
quiz::Quiz("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 sou\
nd modules as a unit."),
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quiz::Quiz("Bar","In music notation, bar is another term for measure a specified per\
iod of time containing a certain number of beats, and marked by bar lines on each si\
de of the written measure."),
quiz::Quiz("Bark Scale","The human auditory (hearing) system can be thought of as co\
nsisting of a series of bandpass filters. Interestingly, the spacing of these filter\
s do not strictly follow either a linear frequency scale or a logarithmic musical sc\
ale. The Bark Scale is an attempt to determine what the center frequency and bandwid\
th of those \"hearing filters\" are (known as critical bands)."),
quiz::Quiz("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, prev\
enting phase cancellation between the two."),
quiz::Quiz("Basic Session","The first audio recording session for recording the basi\
c tracks that serve as the song’s foundation (for example, the drums and bass)."),
quiz::Quiz("Bass Reflex","A type of loudspeaker cabinet design in which a port (open\
ing) in the speaker cabinet enhances bass frequencies. The principle is that the sou\
nd pressure generated by the back of the speaker cone inside the cabinet is routed o\
ut 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."),
quiz::Quiz("Bass","The lower range of audio frequencies up to approximately 250 Hz. \
A reference value."),
quiz::Quiz("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 samp\
le and hold (bucket) in the circuit, while a new level was sampled. Bucket brigade d\
elays (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 multi\
plied by the time interval between samples."),
quiz::Quiz("Beaming","A phenomenon found in loudspeakers in which higher frequencies\
 are projected straight out of the loudspeaker, rather than dispersing along with th\
e lower frequencies. When you stand on-axis in front of the speaker, it sounds as th\
ough it is only reproducing the high frequencies, rather than the mids or lows. This\
 phenomenon is alleviated by routing the high frequncies through horns in the loudsp\
eaker."),
quiz::Quiz("Beat Mapping","The process of adjusting the tempo variations in a record\
ed 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 temp\
o markers. This technique is often used, for example, to reconcile a drum or bass pe\
rformance that was recorded without a click track."),
quiz::Quiz("Beat","1) The steady, even pulse in music. 2) The action of two sounds o\
r audio signals of slightly different frequency interfering with one another and cau\
sing periodic increases and decreases in volume, heard to the ear as “beats.”"),
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quiz::Quiz("Beating","When two oscillators are tuned to very nearly - but not quite \
- the same frequency, the difference between them causes an interference pattern kno\
wn as beating. When the difference in frequency is below the audio rate, this can so\
und like a tremolo applied to the loudness of the combined sound."),
quiz::Quiz("Beatmatching","A technique predominantly used by DJs to synchronize the \
tempos of two recorded tracks, generally through the use of time stretching and pitc\
h shifting tools, to create a seamless transition from one song into another."),
quiz::Quiz("Beats Per Minute (B.P.M.)","BPM (beats per minute) is the most common wa\
y of stating tempo: How many beats (typically, quarter notes) should be counted ever\
y minute. A tempo of 120 beats per minute means there would be two beats every secon\
d (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."),
quiz::Quiz("Berlin School","A particular style of electronic music popularized by th\
e 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 re\
cent versions of Berlin School music can be heard from Node and Red Shift."),
quiz::Quiz("Bi-amplification","A technique in which high and low frequencies in a sp\
eaker or speaker system are driven by two separate amplifiers."),
quiz::Quiz("Bi-Directional Pattern","A microphone pickup pattern which is most sensi\
tive to picking up sounds directly in front and back of the mic, effectively rejecti\
ng sounds coming from the sides. Also called a “figure-8 pattern.”"),
quiz::Quiz("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 trigge\
r signal in a modular synth - even if generated by analog circuitry - could be refer\
red to as a binary type signal. See the entry for Boolean for things you can do with\
 binary signals like gates and divided clocks."),
quiz::Quiz("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 (va\
rying the pitch of an oscillator both above and below the note it is supposed to be \
playing) - are bipolar in nature."),
quiz::Quiz("Bit","The smallest unit of digital information representing a single “0”\
 or “1.”"),
quiz::Quiz("Bitrate (or Bit Depth)","In digital recording, the number of computer bi\
ts used to describe each sample. The greater the bitrate, the greater the dynamic ra\
nge of the sampled sound. The quality and resolution of an audio sample are describe\
d as a combination of sample rate and bitrate. (See also “Sample Rate.”)"),
quiz::Quiz("Blending","The mixing of multiple sounds or channels together to form on\
e sound, or mixing the left and right signals together."),
quiz::Quiz("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."),
quiz::Quiz("Boolean","Boolean logic only can have two states: high or low; 1 or 0; o\
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n or off."),
quiz::Quiz("Boom Stand","A microphone stand equipped with a telescoping support arm \
to hold the microphone."),
quiz::Quiz("Boom","A telescoping support arm attached to a microphone stand holding \
the microphone."),
quiz::Quiz("Boost","To increase gain at specific frequencies with an equalizer."),
quiz::Quiz("Bouncing","(also called “Ping-Ponging” or “Ponging“) The technique of co\
mbining 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 record\
ing them onto separate tracks, or digitally through a digital audio workstation. Bou\
ncing was once used frequently by engineers to free up additional tracks for recordi\
ng, but in digital workstations where tracks are virtually unlimited, this practice \
is basically obsolete. Today, engineers typically bounce tracks for the purpose of c\
reating a preliminary or final mix of a song."),
quiz::Quiz("Boundary Microphone","An omnidirectional microphone designed to be place\
d flush against a flat surface (or boundary), effectively creating a “half-Omni” pic\
kup pattern while eliminating the danger of phase issues from reflected sounds. A po\
pular type of boundary microphone is Crown Audio’s trademark Pressure Zone Microphon\
e (PZM)."),
quiz::Quiz("BPF","A bandpass filter (BPF) leaves the harmonics around the center, co\
rner 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, b\
ased on the number of poles in the filter, with each pole equalling 6 decibels of at\
tenuation for each octave you get away from that center."),
quiz::Quiz("BPM","BPM (beats per minute) is the most common way of stating tempo: Ho\
w many beats (typically, quarter notes) should be counted every minute. A tempo of 1\
20 beats per minute means there would be two beats every second (120 beats/minute x \
1 minute/60 seconds = 2)."),
quiz::Quiz("Breathing","Pumping and Breathing - In studio jargon, an effect created \
when a compressor is rapidly compressing and releasing the sound, creating audible c\
hanges in the signal level. “Pumping” generally refers to the audible increase of so\
und 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 an\
d breathing is a sign of cheap compression or over-compression, and is usually undes\
irable, although some engineers and musicians use it on purpose occasionally to crea\
te a particular effect."),
quiz::Quiz("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-e\
ffects to the audio signal, such as phase shift."),
quiz::Quiz("Bridging","A technique of feeding a single input to both channels of an \
amplifier, then summing them into one, thereby effectively doubling the amplifier po\
wer supplied to the signal."),
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quiz::Quiz("Brownian Noise","Also referred to as brown noise, technically it’s a typ\
e of noise whose power density (spectral loudness) decreases 6 dB per octave with in\
creasing frequency. It has a bass-heavy sound, akin to the sound of the surf at a di\
stance. It can also be used a slowly changing random control voltage or modulation s\
ignal, instead of as an audio source."),
quiz::Quiz("Buchla Bongos","This is a classic patch where a complex sound source - s\
uch 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 a\
ttack/fast decay envelope to create a nice percussive sound. The fact that the low p\
ass gate reduces the higher harmonics as its volume dies away helps tame the harmoni\
cs coming from the complex source, and give it a decay similar to a struck percussiv\
e instrument."),
quiz::Quiz("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 pow\
ers of two (256, 512, 1024, 2048, etc.); the delay length is determined by the numbe\
r of stages multiplied by the time interval between samples."),
quiz::Quiz("Bucking","A type of phase cancellation in which two identical signals or\
 frequencies, having the same amplitude but opposite polarity, cancel one another ou\
t. 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 si\
gnal using this principle."),
quiz::Quiz("Buffered Multiple","Quite often you need to split or copy a signal to se\
nd to more than one destination. This is commonly done with a multiple, where you pl\
ug one source in, and then plug in additional patch cables to go off to multiple des\
tinations. An active or buffered multiple is one that includes a buffer circuit betw\
een the input and output, making sure the signal does not lose its strength or integ\
rity by being split too many times, and that no funny business happening on one of t\
he outputs affects any of the other connections. Some modules have good buffering bu\
ilt 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 tr\
acking 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."),
quiz::Quiz("Bulk Dump","Short for System Exclusive Bulk Dump, a method of transmitti\
ng data such as the internal parameters between MIDI devices."),
quiz::Quiz("Burst Generator","When you send this module a trigger, it outputs a stre\
am or “burst” of triggers in response. You usually have control over the number of t\
riggers, 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 u\
se to create “double pluck” triggers in response to a normal note on; and its most e\
xtreme, it is used to trigger a high-energy, chaotic stream of drum hits that may or\
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 may not be in time with the music."),
quiz::Quiz("Bus Board","This simple circuit board takes the output of your modular s\
ystem's power supply and creates multiple copies of it, routed to connectors that go\
 to your individual modules."),
quiz::Quiz("Bus","An audio pathway by which one or more signals, usually from differ\
ent sources, are routed to a designated place. Because busses are highly connected t\
o signal flow, they serve a broad range of purposes in audio applications. 2) A shor\
thand term for the signals themselves that are routed through the bus (see also “Sub\
group”)."),
quiz::Quiz("Byte","Information (data) bits in a grouping of eight. One byte = eight \
bits."),
quiz::Quiz("Cable Assembly","Cable that is ready for installation in specific applic\
ations and usually terminated with connectors."),
quiz::Quiz("Cable Harness","A grouping of cables or wires used to interconnect elect\
ronic systems."),
quiz::Quiz("Cable Sheath","Conductive protective cover that is applied to cables."),
quiz::Quiz("Cable","A group of one or more insulated conductors, optical fibers, or \
a combination of both within an enveloping jacket, typically for transmitting electr\
ical signals of different types."),
quiz::Quiz("Capacitor","An electronic device made of two plates separated by an insu\
lator, designed to store electrostatic energy. The capacitor is a key component in c\
ondenser microphones, for example."),
quiz::Quiz("Capstan","A mechanical part of a magnetic tape recorder that controls th\
e speed of the tape as it passes across the tape heads."),
quiz::Quiz("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."),
quiz::Quiz("Carbon Microphone","A microphone that uses carbon granules to convert so\
und waves to electrical impulses. The carbon element sits between two plates; as sou\
nd waves hit the carbon granules, it generates changes in resistance between the pla\
tes, affecting the electrical signal."),
quiz::Quiz("Cardioid Pattern","A microphone pickup pattern which is most sensitive t\
o sound coming from the front, less from the sides, and least from the back of the d\
iaphragm. So named because the pickup pattern is in the shape of a heart (cardio).")\
,
quiz::Quiz("Carrier","There are a few different synthesis techniques where one usual\
ly audio-rate signal varies another audio signal. For example, in frequency modulati\
on, 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."),
quiz::Quiz("Cascade","To connect or “daisy chain” two mixers so that the stereo mixi\
ng busses of the first mixer feed into the stereo busses of the second."),
quiz::Quiz("CCW","Counter-clockwise, usually in the context of rotating a control th\
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e left (in the opposite direction of how a clock's hands move)."),
quiz::Quiz("CD","An abbreviation for Compact Disc, or a small optical disk with digi\
tal audio recorded on it."),
quiz::Quiz("Cent","When tuning instruments, a semitone is divided into 100 units cal\
led 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 me\
asured in cents."),
quiz::Quiz("Center Frequency","The frequency of an audio signal that is most affecte\
d 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."),
quiz::Quiz("Channel Path","The complete signal path from the sound source to the mul\
titrack recorder (or DAW). For example, an audio signal that travels from the microp\
hone to the preamplifier, then into a channel strip on the mixing console, then is s\
ent 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 t\
he recorded signals. (See also “Monitor Path.”)"),
quiz::Quiz("Channel","1) An audio recording made on a portion of the width of a mult\
itrack 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."),
quiz::Quiz("Chaotic","Believe it or not, chaotic does not mean completely random to \
mathematicians. Chaos theory deals with systems that are random within certain bound\
aries - 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 sy\
nthesis, a chaotic system usually refers to a modulation generator that is similar t\
o 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."),
quiz::Quiz("Chase","The automatic adjusting of the speed of a recorder (or sequencer\
) to keep time with another recorder."),
quiz::Quiz("Chord Chart","A shorthand form of musical notation that provides the bas\
ic 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 struc\
ture 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.”)"),
quiz::Quiz("Chord","Three or more musical pitches sung or played together."),
quiz::Quiz("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.")\
,
quiz::Quiz("Circuit","1) One complete path of electric current. 2) Similar to defini\
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tion 1, but including all audio signal paths and components to accomplish a particul\
ar audio function."),
quiz::Quiz("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 wo\
rld with MIDI to CV/Gate interfaces that use USB: If your converter is a USB Host, a\
nd you plug a class compliant USB Device such as a controller keyboard or fader pane\
l into it, the converter will recognize it."),
quiz::Quiz("Click Track","A metronome “click” fed into headphone monitors for the pu\
rpose of helping the musicians play in time with the song."),
quiz::Quiz("Clip","All active electronic circuits have a limit on how strong of a si\
gnal 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 hig\
her 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; t\
his 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.”."),
quiz::Quiz("Clock Signal","A signal sent by a device within the circuit that generat\
es 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 con\
nected devices will play in time."),
quiz::Quiz("Clock","Usually refers to the main rhythmic pulse in a system. Often, th\
e clock pulse is much faster than anything it might drive, such as a sequencer or LF\
O. The most common clock rate is 24 ppqn (pulses per quarter note), as is the case w\
ith 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 module\
s that create divisions and multiplications of the main clock to generate new clock \
signals with a relationship to the main clock."),
quiz::Quiz("Clockwise","Clockwise, as in rotating a control the the right - in the s\
ame direction as a clock's hands move."),
quiz::Quiz("Close Miking","A microphone placement technique that places the mic clos\
e to the sound source to pick up the direct sound and reject ambient sound."),
quiz::Quiz("Coaxial Cable","(abbreviated “Coax”) A two-conductor cable that consists\
 of one conductor surrounded by a shield."),
quiz::Quiz("Coincident Miking","A stereo miking technique in which two microphones a\
re placed with their heads as close to each other as possible. This prevents phase c\
ancellation problems in the mix because the distance from the sound to either microp\
hone is the same."),
quiz::Quiz("Compander","A signal processor serving as a combination compressor and e\
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xpander, primarily used for noise reduction purposes in analog systems.  The audio s\
ignal is compressed prior to recording, then expanded at the reproduction stage. Com\
panding is the principle behind Dolby noise reduction systems."),
quiz::Quiz("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."),
quiz::Quiz("Comping","1) In digital audio workstations (DAWs), the process of blendi\
ng portions of multiple recorded takes to create a “compliation” track. (See also “T\
ake,” “Playlist.)  2) In jazz music performance, an abbreviation for “accompanying.”\
"),
quiz::Quiz("Complex Oscillator","This module typically has a pair of oscillators beh\
ind 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 swit\
ch them so that the first modulates the amplitude of the second, or some other varia\
tion. They may also have waveshapers built in. They are based on a popular module cr\
eated by Buchla, which is a standard of the “West Coast” approach to synthesis."),
quiz::Quiz("Compression Driver","A diaphragm that feeds a sound pressure wave into a\
 horn loudspeaker."),
quiz::Quiz("Compression Ratio","The rate by which a compressor attenuates an incomin\
g signal, measured in decibels. For example, a compression ratio of 4:1 means the co\
mpressor will only allow a 1 dB increase in the signal for every 4 dB increase in th\
e signal above the threshold."),
quiz::Quiz("Compression","1) In signal processing, the action performed by a compres\
sor (see also “Compressor”). 2) In acoustics, the increased air pressure caused by t\
he peak of a sound pressure wave, used in the context of “compression and rarefactio\
n” (see also “Rarefaction”)."),
quiz::Quiz("Compressor","A signal processor or plug-in that reduces the dynamic rang\
e of an audio signal by amplifying its quieter sections and attenuating its louder o\
nes."),
quiz::Quiz("Condenser Microphone","A microphone in which sound is converted into ele\
ctrical current through changes in a capacitor. The sound pressure waves move the di\
aphragm, producing changes in capacitance which are then changed into electrical vol\
tage."),
quiz::Quiz("Contact Microphone","A microphone designed to pick up vibrations from so\
lid objects (as opposed to vibrations in the air). Also known as a “pickup” or “piez\
o,” this microphone is often used as an acoustic guitar pickup to pick up the vibrat\
ions from the soundboard, or by experimental musicians creating “noise music” from a\
 variety of objects."),
quiz::Quiz("Control Voltage Processor","CVP is the abbreviation for a module that al\
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lows 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 fr\
om an incoming voltage by running it through a comparator. Make Noise’s Maths is per\
haps the most well known control voltage processor out there; you will also find som\
e 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 d\
o what you want (or to teach them new tricks)."),
quiz::Quiz("Control Voltage","The concept of control voltage (CV) is at the very roo\
t of modular synthesizer. The general idea is that analog voltage levels are used co\
ntrol functions and parameters of a module. For example, one control voltage may det\
ermine 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 synthes\
izer 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."),
quiz::Quiz("Controller","In the broadest sense, a controller is any device that is u\
sed 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 device\
s. Other examples of controllers in the recording studio can include monitor control\
lers, DAW controllers and DJ controllers."),
quiz::Quiz("Corner Frequency","The cutoff or corner frequency of a filter is the poi\
nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
tered” – reduced in loudness – the further above 500Hz you go."),
quiz::Quiz("Counter Clockwise","Counter-clockwise, usually in the context of rotatin\
g a control the left (in the opposite direction of how a clock's hands move)."),
quiz::Quiz("CPU","Abbreviation for Central Processing Unit, the main “brain” chip in\
 a computer (also known simply as “Processor”)."),
quiz::Quiz("Critical Distance","The distance from the sound source at which the dire\
ct sound and the reverberant sound are at equal volume. Critical distance varies acc\
ording to the space; in a room with absorbent walls, the critical distance will be f\
urther from the source, and in a reverberant room, the distance will be closer to th\
e source."),
quiz::Quiz("Crossfade","An audio editing technique in which one sound is faded out a\
s 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 reco\
rded track into a composite take. Club DJs also use crossfading to transition from o\
ne song to the next with no stops."),
quiz::Quiz("Crossover Frequency","The frequency at which the crossover stops sending\
 the signal to one speaker and starts sending it to another."),
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quiz::Quiz("Crossover","An audio filter component that splits an audio signal into t\
wo or more bands or signals, usually to be fed into different components of a loudsp\
eaker system according to frequency range. (Also called a “crossover network.”)"),
quiz::Quiz("Crosstalk","The unwanted leakage of an audio signal between two audio ch\
annels—for example, overlapping signals between channels on a mixing console, or ove\
rlapping audio between two tracks of audiotape."),
quiz::Quiz("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) Th\
e 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 fo\
r video production."),
quiz::Quiz("Cutoff Frequency","The cutoff or corner frequency of a filter is the poi\
nt at which is starts filtering. For example, if a low-pass filter has a corner freq\
uency of 500 Hz (cycles per second), all harmonics or other sound components below 5\
00 Hz will be allowed through untouched, and all harmonics above 500 Hz will be “fil\
tered” – reduced in loudness – the further above 500Hz you go."),
quiz::Quiz("Cutoff Slope","The rate of reduction of the frequencies beyond the passb\
and of a filter. The slope is described as the number of dB the filter reduces the s\
ignal for each octave past the cutoff frequency."),
quiz::Quiz("CV/Gate","This is the shorthand to say a synthesizer may be controlled b\
y voltages – usually for pitch – and gate signals to indicate when a note is “on.” A\
n increasing number of controller keyboards are including CV/Gate output in addition\
 to the customary MIDI (Musical Instrument Digital Interface), making them much easi\
er to connect to a modular synthesizer, as no additional MIDI to CV interface is req\
uired."),
quiz::Quiz("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 functio\
ns and parameters of a module. For example, one control voltage may determine the pi\
tch played by an oscillator; a second control voltage may determine how loud that si\
gnal is after it’s passed through a voltage-controlled amplifier. CV is the most com\
mon shorthand to refer to control voltage – for example, when a synthesizer module s\
ays it features “CV over the filter’s resonance,” that means there is a control volt\
age input to control the amount of resonance (feedback) – not just the customary kno\
b on the front panel."),
quiz::Quiz("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 kno\
wn control voltage processor out there; you will also find some modules with CVP spe\
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cifically in their name. Regardless, it’s good to have one or more of this type of m\
odule in your system to help massage voltages to get them to do what you want (or to\
 teach them new tricks)."),
quiz::Quiz("CW","Clockwise, as in rotating a control the the right – in the same dir\
ection as a clock's hands move."),
quiz::Quiz("Cycle","One complete expression of a waveform beginning at a certain poi\
nt, progressing through the zero line to the wave’s highest and lowest points, and r\
eturning to the same value as the starting point. One complete vibration or sound wa\
ve."),
quiz::Quiz("D-Sub Connector","Abbreviation for “D-subminiature connector,” a D-sub i\
s a multipin connector that is most often used to connect a computer to a VGA monito\
r, but also used occasionally in digital audio applications in the recording studio.\
"),
quiz::Quiz("D/A","Abbreviation for Digital to Analog conversion, which changes digit\
al data numbers (digital audio signal) into discrete voltage level. The reverse proc\
ess of A/D. Also known as DAC."),
quiz::Quiz("DADSR","This is a slightly fancier take on the standard ADSR envelope ge\
nerator that introduces an initial timed delay before the initial attack stage (risi\
ng 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 n\
ote starts, and then after a pause it starts to swell into a brighter, fuller sound.\
"),
quiz::Quiz("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 secon\
d to reach the third, etc."),
quiz::Quiz("Damping Factor","Describes an amplifier’s ability to restrain the pushba\
ck motion (back-EMF) of the loudspeaker cone when the audio signal stops."),
quiz::Quiz("Damping","The reduction of energy in a vibrating system, through frictio\
n. Can refer to the reduced amplitude in an electrical signal, or the stifled vibrat\
ions of a musical instrument (for example, the damper pedal on an acoustic piano).")\
,
quiz::Quiz("DAW","An abbreviation for Digital Audio Workstation, a device or softwar\
e program designed for recording and mixing audio digitally."),
quiz::Quiz("dB","An abbreviation for decibel, a measurement ratio that compares sign\
al strengths (usually audio levels)."),
quiz::Quiz("DBX","A series of noise reduction systems, named for the company that de\
veloped them. DBX noise reduction has been less commercially successful than the mor\
e widely known Dolby systems, but is still found on occasion in recording studios.")\
,
quiz::Quiz("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, a\
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s an AC coupled input would try to remove the DC component of the signal (such as it\
s sustain level) and return it to 0v."),
quiz::Quiz("DC","Electrical current that flows in a single direction, as opposed to \
Alternating Current (AC), which flows in alternating directions. Many electronic dev\
ices run on DC, which is usually provided by battery power, USB power or an AC adapt\
er plugged into the wall. In modular terms, DC refers to a voltage that tends to sta\
y at one steady level for awhile, such as a gate output that switches between 0v whe\
n a note is off and 5 or 10v when a note is on. It can also refer to a slowly changi\
ng voltage, such as an envelope."),
quiz::Quiz("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 p\
itch that cause subtle detuning (and therefore the perception of “fatness”) when usi\
ng more than oscillator per voice."),
quiz::Quiz("De-esser","An audio compressor designed to reduce the volume of sibilant\
 sounds and frequencies, especially those produced by pronouncing the letter “s.”"),
quiz::Quiz("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."),
quiz::Quiz("Decca Tree","A stereo microphone placement technique involving three mic\
rophones (usually omnidirectional) placed in a “T” pattern. Commonly used in miking \
choirs, orchestras and other large ensembles, but variations of the Decca tree techn\
ique are also being used today in surround sound situations."),
quiz::Quiz("Decibel","(abbreviated “dB“) The ratio measurement of two levels accordi\
ng to a scale where a certain percentage change comprises one unit. Most often used \
to describe audio levels."),
quiz::Quiz("Degaussing","The process of demagnetizing an object. In the context of a\
udio, degaussing essentially erases the recording on magnetic tape."),
quiz::Quiz("Delay/Attack/Decay/Sustain/Release","This is a slightly fancier take on \
the standard ADSR envelope generator that introduces an initial timed delay before t\
he 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 mut\
ed, filtered sound when the note starts, and then after a pause it starts to swell i\
nto a brighter, fuller sound."),
quiz::Quiz("Delay","You all know what the word delay means in the normal world; it c\
an appear in different forms inside a modular synth. For example, it can refer to th\
e 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 signa\
l, such as a gate, trigger, or initial stage of an envelope so a note would start la\
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ter than it was actually played. Also, 1) An process by which an audio signal is rec\
orded to a medium or device, reproduced at a time delay, then mixed with the origina\
l, non-delayed signal to create a variety of effects such as a fuller sound, echo, c\
horusing, flanging, etc.  2) A signal processor that creates delay effects."),
quiz::Quiz("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 minima\
l tracking or production, almost like a “rough draft” of a recording."),
quiz::Quiz("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 th\
an 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 interes\
ting and “full” - and a bit more natural, as two singers or instruments can rarely h\
it 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 stu\
dio, but is often used in “double-tracking,” blending the detuned instrument/track w\
ith the original to create a fuller sound."),
quiz::Quiz("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."),
quiz::Quiz("Dialogue","The spoken word recorded in film/video sound, commercials and\
 instructional recordings."),
quiz::Quiz("Diaphragm","The part of a microphone that moves in response to sound wav\
es, converting them to electrical signals."),
quiz::Quiz("Difference","A fancy way of saying you subtracted on control voltage fro\
m another. It can also be applied to audio or harmonics."),
quiz::Quiz("Digital Audio Workstation","abbreviated DAW) A device or computer softwa\
re 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 co\
mmonly 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 in\
clude Reason, Ableton and Logic."),
quiz::Quiz("Digital Multimeter","A small device that tests electrical voltage, curre\
nt, and resistance. Multimeters are useful in recording studios for calibrating elec\
trical systems and troubleshooting problems."),
quiz::Quiz("Digital Recording","The process of converting audio signals into numbers\
 that represent the waveform, then storing these numbers as data."),
quiz::Quiz("Digital Signal Processing","(abbreviated “DSP”) Any signal processing do\
ne after an analog audio signal has been converted into digital audio."),
quiz::Quiz("Digital to Analog Converter","(abbreviated D/A) A device that converts t\
he digital data of digital audio into voltage levels that approximate the original a\
nalog audio."),
quiz::Quiz("Digital","There was a time when digital (referring to circuitry based ar\
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ound 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 wor\
d among synthesists. The assumption was digital techniques created sounds that were \
more sterile, brittle, and abrasive – and just not as “authentic.” Today, digital ci\
rcuitry is embraced in synthesizers, including modular systems. Although analog will\
 always hold a special place in our hearts, a well-implemented digital circuit can s\
ound just as good as an analog one, while digital signal processing and programming \
can create a wider range of sounds than most analog circuitry."),
quiz::Quiz("Digitally Controlled Oscillator","A DCO (Digitally Controlled Oscillator\
) is a hybrid design for an analog oscillator that – instead of using a voltage leve\
l 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, the\
re are no imperfections in pitch that cause subtle detuning (and therefore the perce\
ption of “fatness”) when using more than oscillator per voice."),
quiz::Quiz("DIN Stereo","A stereo microphone placement technique that places two car\
dioid microphones about 20cm apart and set outward from each other at a 90-degree an\
gle to create a stereo image.  Particularly for stereo miking at close ranges. (See \
also “Near-Coincident Miking.”)"),
quiz::Quiz("DIN Sync","A clock signal for controlling the tempo of sequencers, arpeg\
giators, and drum machines, distributed using cables with DIN-style connectors (yes,\
 just like old-fashioned MIDI connectors, but DIN Sync is even older). Roland pionee\
red 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 a\
t 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."),
quiz::Quiz("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 resona\
nce."),
quiz::Quiz("Diode","An electrical component that enables easy electrical current flo\
w in one direction but not the other. In the recording studio, these are commonly fo\
und in the vacuum tubes of tube amplifiers."),
quiz::Quiz("Direct Box","A small device that to converts an unbalanced, high-impedan\
ce speaker or instrument-level output to a balanced, low-impedance mic-level output.\
 Frequently used in the signal path connecting electric instruments “directly” to th\
e mixing console, as opposed to miking them acoustically. Also called “direct inject\
ion box” or “DI box.”"),
quiz::Quiz("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 cha\
nging voltage, such as an envelope. (abbreviated “DC“) Electrical current that flows\
 in a single direction, as opposed to Alternating Current (AC), which flows in alter\
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nating directions. Many electronic devices run on DC, which is usually provided by b\
attery power, USB power or an AC adapter plugged into the wall."),
quiz::Quiz("Direct Injection","(abbreviated “DI”) The process of sending an electric\
al 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."),
quiz::Quiz("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. T\
his feature is often used to send a “dry” signal to a monitor mix or a recording dev\
ice."),
quiz::Quiz("Direct Sound","The sound that reaches a microphone or a listener’s ear w\
ithout hitting or bouncing off any obstacles (as opposed to reflected or ambient sou\
nd)."),
quiz::Quiz("Directional Pattern","1) In microphones, a term meaning the same thing a\
s “Pick Up Pattern,” a description of the area in which a microphone is most sensiti\
ve 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."),
quiz::Quiz("Dispersion (also Dispersion Angle)","The area that is effectively covere\
d 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 s\
tanding directly in front of the speaker. Each speaker has both a horizontal and ver\
tical dispersion angle."),
quiz::Quiz("Distant Miking","The technique of placing a microphone far from the soun\
d source in order to pick up a combination of the direct and reflected sounds."),
quiz::Quiz("Distortion","Refers to the deforming of a waveform at the output of a de\
vice as compared with the input, usually due to overload, creating a distorted or “d\
irty” signal. While electrical or audio distortion is typically unwanted and avoided\
, it is frequently used in controlled situations in audio to create certain desirabl\
e effects, particularly with electric guitars and amplifiers."),
quiz::Quiz("Diversity","1) In audio settings: the use of two or more antennas in a w\
ireless receiver system to prevent dropouts in the audio from a wireless microphone.\
 2) In other settings: the embracing of the uniqueness of all individuals."),
quiz::Quiz("Dolby","The brand name of a manufacturer of noise reduction systems and \
other audio systems, to improve performance and fidelity of audio recording, playbac\
k, and transmission."),
quiz::Quiz("Doppler Effect","The phenomenon in which the human ear perceives a chang\
e in the frequency (pitch) of a sound while the sound source is in motion. As the so\
und source approaches, the sound waves travel a shorter distance to the ear, increas\
ing 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 frequenci\
es. 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 bet\
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ween the sound source and the listener (or microphone) and creating its signature wa\
rbling vibrato effect."),
quiz::Quiz("Double","1) To record a second performance closely matching the first pe\
rformance, for the purpose of blending the two tracks. 2) To use a delay line with m\
edium delay to simulate double tracking."),
quiz::Quiz("Driver","1) A transducer in a loudspeaker that converts electrical signa\
ls into sound pressure waves. 2) A computer program that controls an attached device\
 or piece of hardware."),
quiz::Quiz("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 unwan\
ted dip in audio, a crackle or a pop."),
quiz::Quiz("Drum Machine","An electronic device containing synthesized and/or sample\
d drum sounds in its memory, along with an internal sequencer that can be programmed\
 to play drum patterns or loops."),
quiz::Quiz("Drum Pattern","A specific sequence of drum sounds played by a drummer or\
 sequenced into a drum machine for use in a song."),
quiz::Quiz("Dry","A sound with no effects is referred to as \"dry\"; a sound with ef\
fects (such as reverb) mixed is referred to as \"wet.\" Effects units or mixers ofte\
n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
d and the fully-effected sound."),
quiz::Quiz("DSP","Any signal processing done after an analog audio signal has been c\
onverted into digital audio."),
quiz::Quiz("Dub (or Dubbing)","1) To copy a recording. 2) To record in real time wit\
h 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 ele\
ctronic music."),
quiz::Quiz("Ducking","A compression-based audio effect in which an audio signal is r\
educed proportionately by the presence of another audio signal, sometimes accomplish\
ed 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 e\
ngineer “ducks” specific sounds to make room for others in the track; for example, w\
hen a bass guitar signal triggers a slight reduction in the level of drums or guitar\
s. (See also “Sidechain.”)"),
quiz::Quiz("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 t\
hey could play two separate notes as once. Some MIDI interfaces for modular synths i\
nclude 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;"),
quiz::Quiz("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\
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 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."),
quiz::Quiz("Dynamic Microphone","(Also called Moving Coil Microphone) A microphone i\
n which sound pressure waves are converted to an electrical audio signal by an induc\
tion coil moving within a magnetic field—a process often compared to a loudspeaker w\
orking in reverse. Dynamic microphones are less sensitive than condenser microphones\
, but can be effective for miking louder sound sources or for close-miking applicati\
ons."),
quiz::Quiz("Dynamic Processing/Dynamic Signal Processing","The process of automatica\
lly changing the level (or gain) to alter the level relationship of the loudest audi\
o to the softest audio. Dynamic processors include compressors, limiters, expanders \
and gates."),
quiz::Quiz("Dynamic Range","1) The ratio (in dB) between the loudest peak and the so\
ftest level of a song or recording. 2) The ratio (in dB) between the softest and lou\
dest possible levels a device or system can provide without distortion."),
quiz::Quiz("Early Reflections","The first sound waves that reach a listener’s ear af\
ter bouncing off a surface in the room, usually heard almost immediately after the i\
nitial sound. The first stage of reverberation."),
quiz::Quiz("East Coast Synthesis","This blanket term is applied to most common synth\
esizer 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 osci\
llators 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 subt\
ractive synthesis, as the filter reduces (subtracts) harmonics that came from the os\
cillators. East Coast synthesizers also regularly have organ-style black & white key\
boards, and four stage ADSR type envelopes. Today it's common to mix both East Coast\
 and West Coast approaches in the same system."),
quiz::Quiz("Echo Chamber","An enclosed room designed with reflective, non-parallel s\
urfaces for the purpose of creating acoustic echoes (reverberation)."),
quiz::Quiz("Echo","The distinct repetition of an initial sound, caused by the reflec\
tion of the sound waves upon a surface. We recognize a sound as an echo when the dis\
tance between the source and the reflection is far enough apart that we can detect t\
he time delay between one and the other. Essentially, reverberation is the combinati\
on of many echoes occurring too rapidly to hear each individually. In the studio, ec\
hoes can be reproduced acoustically or simulated by a digital signal processor."),
quiz::Quiz("Edit","To change one or more parameters of a recorded sound after the fa\
ct. 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 g\
raphically; changing the sequence of playback; and many others. Analog editing would\
 typically involve splicing the magnetic tape on which the audio signals were record\
ed. These days, almost all editing in the studio is done via computer using a digita\
l audio workstation (DAW)."),
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quiz::Quiz("Effect Loop","Sometimes you might want to send a signal outside your mod\
ular system, process it through an external effects device, and bring it back into y\
our 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."),
quiz::Quiz("Effects Processor","(Also called Guitar Processor) A device that adds au\
dio effects to a direct guitar signal, such as reverb, chorusing, flanging, delay, o\
verdrive, amplifier simulation, etc. Effects processors can occur as individual effe\
cts boxes or multi-sound pedal boards (see also “Foot Pedals,” “Foot Switches”) adde\
d 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."),
quiz::Quiz("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 m\
ultitrack recording."),
quiz::Quiz("Effects","1) Various ways an audio signal can be modified by adding some\
thing 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 fi\
lm or video."),
quiz::Quiz("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 (Vo\
ltage Controlled Filter). To do this, and envelope generator creates a voltage that \
typically rises from zero volts to some maximum level, and back down again. You cont\
rol how long this takes, usually in various stages: an attack stage as it goes from \
zero to max, a decay stage as if 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 sust\
ain, and then (usually after a key has been released and the corresponding gate sign\
al has gone back to zero) from the sustain level back to zero over a duration known \
as its release."),
quiz::Quiz("Electret Microphone","A variation of condenser microphone that uses an e\
lectret instead of a capacitor. (Also called “Electret Condenser Microphone.”) Becau\
se the electret is permanently polarized, an electret microphone does not require an\
 external power source as a standard condenser microphone does."),
quiz::Quiz("Electret","A dielectric plate that is designed with permanent polarity, \
allowing it to function similarly to a magnet. (“Electret” comes from the words “ele\
ctricity” and “magnet.”) Used in some microphone types in place of a capacitor (cond\
enser)."),
quiz::Quiz("Electromagnetic Field","(Abbreviated EMF) A field of magnetic energy put\
 out because of current traveling through a conductor."),
quiz::Quiz("Electromagnetic Interference (EMI)","The bane of audio professionals eve\
rywhere, EMI is a type of interference caused by nearby electromagnetic activity, wh\
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ich can be picked up by audio cables and equipment, causing unwanted noise, hum or b\
uzz in audio systems. Common causes of EMI in audio systems may include high-current\
 power lines, fluorescent lighting, dimmer switches, computers, video monitors and r\
adio transmitters."),
quiz::Quiz("Electrons","Negatively charged particles revolving around the nucleus of\
 an atom. Electrical current is generated by electrons moving along a conductor, lik\
e a metallic wire."),
quiz::Quiz("Emphasis","This word can have two meanings. In a normal audio context, i\
t usually means some form of high frequency boost, as emphasizing the higher harmoni\
cs 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 se\
tting boosts (emphasizes) the harmonics at the cutoff or corner frequency."),
quiz::Quiz("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 p\
erform 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."),
quiz::Quiz("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 Ampl\
ifier), as well as how its frequency content or timbre changes over time when connec\
ted to a VCF (Voltage Controlled Filter). To do this, and 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 if falls back down from maximum to eit\
her zero (in the case of an AD, or Attack/Decay envelope) or an intermediate level k\
nown as the sustain, and then (usually after a key has been released and the corresp\
onding gate signal has gone back to zero) from the sustain level back to zero over a\
 duration known as its release."),
quiz::Quiz("Envelope Tracking","This describes the main action of an envelope follow\
er: a module or section of a module that follows the loudness of a signal and output\
s a voltage that corresponds to – tracks – that input."),
quiz::Quiz("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 describe\
s how a sound or audio signal varies in intensity over a period of time."),
quiz::Quiz("Equal Loudness Contours","A drawing of several curves showing how loud t\
he tones of different frequencies would have to be played for a person to say they w\
ere of equal loudness. (See also “Fletcher-Munson Curves.”)"),
quiz::Quiz("Equalizer","An audio signal processor that uses one or more filters to b\
oost or cut the amplitude (volume) of certain frequencies within the sound. The unde\
rlying principle is to balance or “equalize” the frequency response of the audio sys\
tem, 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.")\
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,
quiz::Quiz("Eurorack","Eurorack is arguably the most popular format of modular synth\
esizer today, with over 100 manufacturers and over 1000 modules available. It was cr\
eated by Doepfer Musikelektronik in 1995, basing its size off the Eurorack format fo\
r lab equipment. Some users will try to tell you that Eurorack doesn’t “sound” as go\
od as other formats, but that’s just based on a few substandard manufacturers or mod\
ules; there’s nothing inherent to the standard that makes a huge difference in the f\
inal sound (no; the difference between 12 and 15 volt power supplies is not enough t\
o most ears)."),
quiz::Quiz("Expander","A signal processor (or plug-in) that performs the opposite fu\
nction of a compressor, expanding the dynamic range of an audio signal rather than c\
ompressing it. It accomplishes this by further reducing the amplitude of signals tha\
t drop below a set threshold."),
quiz::Quiz("Expansion Ratio","The rate by which an expander attenuates an incoming s\
ignal, measured in decibels. For example, an expansion ratio of 2:1 means the expand\
er 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.\
"),
quiz::Quiz("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 a\
n 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 i\
n the same increase of cycles per second."),
quiz::Quiz("Fade","A gradual reduction of the level of the audio signal, or a gradua\
l change of level from one pre-set level to another."),
quiz::Quiz("Fader","A control which adjusts the level (gain or attenuation) of an in\
coming signal to a channel or grouping of channels on a console."),
quiz::Quiz("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 fa\
r field begins around 3 feet from the sound source. Audio engineers often use both n\
ear field and far field monitoring when fine-tuning a mix. (See also “Critical Dista\
nce,” “Near Field.”)"),
quiz::Quiz("Feed","To send an audio or control signal to."),
quiz::Quiz("Feedback Control","The control on a delay line or delay effects device t\
hat controls the amount of feedback into the system."),
quiz::Quiz("Feedback","The return of a portion of the output signal back into the in\
put 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 i\
s picking up the sound from a nearby speaker, generating a loud, oscillating frequen\
cy that increases in intensity until the feedback loop is broken by turning off the \
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mic or speaker, or by use of an equalizer to attenuate the frequency."),
quiz::Quiz("Fidelity","A term describing how accurately a sound is reproduced from i\
ts original source."),
quiz::Quiz("Figure-8 Pattern","A microphone pickup pattern which is most sensitive t\
o picking up sounds directly in front and back of the mic, effectively rejecting sou\
nds coming from the sides."),
quiz::Quiz("Filter","A module that reduced or removes certain frequencies and harmon\
ics from the sound that is passed through it. In a synthesizer, the most typical fil\
ter types are low pass (passes all of the harmonics below its cutoff or corner frequ\
ency untouched, and then reduces the level of higher harmonics the further you go ab\
ove that cutoff frequency), high pass (passes all harmonics above its cutoff frequen\
cy untouched, and reduces the level of progressively lower harmonics below the cutof\
f), 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 no\
tch (harmonics right around the cutoff frequency are reduced or cut out entirely; ot\
hers above or below are allowed to live)."),
quiz::Quiz("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 inpu\
t signal is delayed by a very small amount (longer than a chorus effect; shorter tha\
n an echo effect) and mixed in with the original. When the delay is constant, the re\
sult 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 th\
e flange of one of the tape reels to delay it."),
quiz::Quiz("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 “swi\
rling” sound. This was originally accomplished in analog tape recording by playing t\
he original tape and the copy on two tape machines simultaneously, then physically p\
ressing on the flange of one of the machines to alter the timing of the duplicate tr\
ack. These days, most flanging is done through delay boxes or digital plug-ins."),
quiz::Quiz("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 nota\
tion, an “accidental” mark that instructs the player to play/sing the note one-half \
step lower."),
quiz::Quiz("Fletcher-Munson Curves","Also known as “Equal Loudness Contours,” a set \
of graphical curves plotted to illustrate how the human ear responds to different fr\
equencies at different volume levels. Named after the two researchers who first plot\
ted the curves. (See also “Equal Loudness Contours.”)"),
quiz::Quiz("Flip-Flop","In binary logic terms, a flip-flop toggles between high and \
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low every time it receives an input trigger (i.e. the first trigger would set the ou\
tput 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."),
quiz::Quiz("Floating Unbalanced Line","A connection “workaround” in which an unbalan\
ced output is connected to a balanced input by modifying the connections in the line\
 to resemble a balanced line, alleviating unwanted hum or buzz."),
quiz::Quiz("Fly In","To add sounds into a mix or recording that have no synchronizat\
ion."),
quiz::Quiz("Flying Bus","This is a very simple type of power distribution or bus boa\
rd that typically uses a ribbon cable with multiple connectors along its length to t\
ake the output of your power supply and distribute it to your individual modules. Th\
ey're cheap and easy to install and use, but in a few cases might be a cause of nois\
e being shared between modules."),
quiz::Quiz("FM","Frequency modulation (FM for short) refers to a synthesis technique\
 where the pitch of an oscillator is varied (modulated) very quickly – at audio rate\
s – 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."),
quiz::Quiz("FOH","In live audio settings, the location in a venue opposite the stage\
, where live audio for the show is controlled and mixed."),
quiz::Quiz("Foldback","A stage monitoring system used in live audio. A set of on-sta\
ge 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 t\
o play. This mix is usually different from the FOH (front-of-house) mix that the aud\
ience 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 freq\
uently 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 goo\
d mix for the audience. For this reason, many live audio systems now use in-ear moni\
toring as an alternative to stage monitors to control the onstage noise and reduce t\
he risk of feedback."),
quiz::Quiz("Foot Pedal","An effects device controlled by a musician with his foot."),
quiz::Quiz("Foot Switch","A switch placed on the floor and pressed by a musician to \
do various functions."),
quiz::Quiz("Force-Sensing Resistor","In modular systems, an FSR (Force-Sensing or -S\
ensitive Resistor) usually takes the form of a circular pad that you press on to var\
y a parameter. It acts as a resistor that decreases in resistance the harder you pre\
ss."),
quiz::Quiz("Formant","Many instruments based on vibrating tubes – including our own \
vocal tract – have certain frequencies that they like to vibrate or “resonate” at. W\
hen you send a sound down these tubes, they will accentuate the frequency of that so\
und (or some of its harmonics) to match these resonate frequencies. Each of these re\
sonant frequencies is known as a formant of that instrument. A common way of synthes\
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izing 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 i\
s 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 reso\
nances that occur as a result of the physical structure of the sound source. These f\
requencies are what create timbre, that element of sound that creates the specific s\
ound of a guitar, a flute, a male or female voice, etc."),
quiz::Quiz("Format","1) One of many different media used to store and reproduce audi\
o, whether in the recording studio or for listening purposes. Examples include curre\
ntly 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 reco\
rding to analog tape, such as number of tracks, width, spacing and order. 3) To prep\
are a hard drive or memory card for use, usually erasing all existing data in the pr\
ocess."),
quiz::Quiz("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-qua\
drant 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 usua\
lly follow. Namely, the modulation frequency is both added to and subtracted from th\
e carrier’s frequency; the resulting harmonics replace the original carrier and modu\
lator. Say the carrier was a sine wave (only the fundamental harmonic present) at 60\
0Hz, and the modulator was a sine wave at 100Hz. The result would be a tone that had\
 frequency components at 500 and 700Hz."),
quiz::Quiz("FracRack","A less-common format of modular synthesizers put forward by P\
AiA and Blacet Research. It stands for Fractional Rack; one unit is 1.5” (3.8 cm) wi\
de by 3U, or 5.25” (13.3 cm) high."),
quiz::Quiz("Fractional Rack","A less-common format of modular synthesizers put forwa\
rd 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."),
quiz::Quiz("Frequency Modulation (FM) Synthesis","A method of sound synthesis in whi\
ch the frequencies generated by one oscillator (the carrier) are altered by the outp\
ut of one or more additional oscillators (operators) to create a diversity of harmon\
ically rich sounds."),
quiz::Quiz("Frequency Range","1) The range of frequencies over which an electronic d\
evice puts out a useful signal (see also “Bandwidth”). 2) The range of frequencies t\
hat 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 al\
so hear within a certain frequency range."),
quiz::Quiz("Frequency Response","The range between high and low frequencies that a c\
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omponent of an audio system can adequately handle, transmit or receive."),
quiz::Quiz("Frequency-Agile","In wireless microphone systems, frequency-agile descri\
bes the ability of the system to operate on a choice of different RF frequencies wit\
hin 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 R\
F frequency of the device can be changed to avoid interference."),
quiz::Quiz("Frequency-Shift Key (FSK)","A now out-of-date protocol in which a sync t\
one is recorded onto a spare track of a multi-track tape recorder to enable electron\
ic 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.”)"),
quiz::Quiz("Frequency","The number of occurrences of a particular event within a cer\
tain amount of time. In audio and acoustics, frequency specifically refers to the nu\
mber of complete cycles a vibration or waveform makes in a second, measured in cycle\
s per second, or Hertz (Hz). In sound, frequency determines what we hear as pitch. T\
he longer the wavelength, the fewer the cycles per second, and the lower the pitch."\
),
quiz::Quiz("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\
."),
quiz::Quiz("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 a\
cts as a resistor that decreases in resistance the harder you press."),
quiz::Quiz("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 pat\
h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\
rd into either the input or output side; when it is “half-normalled,” the path chang\
es 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 correspond\
ing output."),
quiz::Quiz("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."),
quiz::Quiz("Function Generator","The term function generator can have two meanings i\
n 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 sin\
e or creating a rise & fall in response to a trigger."),
quiz::Quiz("Fundamental","(Also called fundamental frequency or first harmonic) The \
lowest frequency present in the sounding of a note by musical instrument or voice.")\
,
quiz::Quiz("Gain Control","A device that changes the gain of an amplifier or circuit\
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, 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."),
quiz::Quiz("Gain Reduction","The action of a compressor or limiter in regulating the\
 amplitude of the audio signal."),
quiz::Quiz("Gain Structure","A term that describes the interconnection of multiple c\
omponents in an audio system, and the amount of gain increase or reduction that occu\
rs at each point. A configuration with a good gain structure means that the componen\
ts are working properly together to provide optimal gain with minimal distortion or \
noise."),
quiz::Quiz("Gain","The amount of increase in audio signal strength, often expressed \
in dB."),
quiz::Quiz("Gate Detector","This is one of the main signal types that are passed aro\
und 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 controll\
er), or when a sequencer jumps to the next “stage” or note. A gate typically stays a\
t that level for the duration of the note (i.e. while the key is being held down), a\
nd suddenly drops or “goes low” to its resting level - typically 0 volts, but someti\
mes -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 De\
cay stages; while the gate remains high, the envelope stays at its Sustain level, an\
d when the gate goes low again, the envelope moves onto its Release stage."),
quiz::Quiz("Generation Loss","The amount of clarity lost when recorded audio is copi\
ed, due to added noise and distortion."),
quiz::Quiz("Generation","A term used to describe the number of times that the record\
ed audio signal has been copied."),
quiz::Quiz("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 volta\
ge 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 sl\
ew generator, slew limiter, slope generator, or lag. Some use the terms glide, gliss\
ando, and portamento interchangeably, but if you want to split musical hairs, a glis\
sando (gliss) is a different effect where the intermediate notes are more distinct -\
 such as played rapidly in order - rather than slurred through."),
quiz::Quiz("Golden Section","(also called Golden Ratio) A ratio of height to width t\
o length, where the width is approximately 1.6 times the height, and the length appr\
oximately 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, inc\
luding room dimensions and studio design (to achieve “optimal acoustics” in the room\
), and even in the design of certain acoustic instruments."),
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quiz::Quiz("Granular Synthesis","Granular synthesis can be thought of as particle th\
eory applied to sound. The concept is that a sound can be broken down into very smal\
l “grains” – typically 1-50 or 100 msec in duration. These tiny snippets are then pl\
ayed back to reproduce the original sound, or to create new sounds by changing the s\
peed, 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 ca\
n range from audio processing tricks such as changing speed without changing pitch a\
nd vice versa, to creating psychedelic “clouds” of sound (and indeed, there is a pop\
ular module called Clouds)."),
quiz::Quiz("Graphic Equalizer","A type of equalizer that can adjust various frequenc\
ies of the incoming signal using sliders that are assigned to specific frequency ban\
ds. (See also “Equalizer.”)"),
quiz::Quiz("Ground Lift Plug","An adapter that enables a three-prong power cord to p\
lug into two-prong outlet. Some engineers wrongly use this plug to interrupt the gro\
und connection and prevent buzz, but it is a VERY unsafe practice to break the groun\
d connection using this plug without grounding the unit by another means."),
quiz::Quiz("Ground Lift Switch","A switch that breaks the connection between the gro\
und point in one circuit and the ground point in another circuit, for the purpose of\
 eliminating hum or buzz caused by ground loops."),
quiz::Quiz("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-freque\
ncy hum that can be annoying at best and cause damage to gear at worst. The best res\
olution 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 man\
agement of electricity."),
quiz::Quiz("Group (or Grouping)","A number of input channels on a console that can b\
e controlled and adjusted as a single set before sending the combined signal to the \
master output. Sometimes also called “Submix,” “Bus” or just “Group.”"),
quiz::Quiz("Group Delay","In audio, group delay is a phenomenon within all electroni\
c audio devices (e.g., speakers, amplifiers) in which different frequencies in the s\
ignal are output at slight delays from one another. In simpler terms, lower frequenc\
ies 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 a\
ny configuration is to keep the group delay as small as possible; in cases of extrem\
ely poor configurations, the delays between highs and lows can be audible."),
quiz::Quiz("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.\
"),
quiz::Quiz("Guitar Processor","A device that adds audio effects to a direct guitar s\
ignal, such as reverb, chorusing, flanging, delay, overdrive, amplifier simulation, \
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etc. Effects processors can occur as individual effects boxes or multi-sound pedal b\
oards (see also “Foot Pedals,” “Foot Switches”) added into the signal path between t\
he guitar and the console. They can also be found as presets in guitar amplifiers, o\
r even as digital plug-ins within a DAW."),
quiz::Quiz("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 t\
han 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 direc\
tion of the sound. This is what helps us distinguish the original sound source witho\
ut being confused by reflections and reverberations off of nearby surfaces. Understa\
nding the Haas effect is particularly useful in live audio settings, especially in l\
arge venues where loudspeakers are time-delayed to match the initial sound waves com\
ing from the source."),
quiz::Quiz("Half Step","A change in pitch equivalent to adjacent keys on a piano. Al\
so known as a “semitone.”"),
quiz::Quiz("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 pat\
h. When a patch bay is “full-normalled,” the connection is altered by inserting a co\
rd into either the input or output side; when it is “half-normalled,” the path chang\
es 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 correspond\
ing output."),
quiz::Quiz("Half-Wave Rectifier","A half-wave rectifier passes only positive voltage\
s, and replaces anything negative with 0v. In other words, anything “below zero” is \
clipped off."),
quiz::Quiz("Hall Program","A setting of a digital delay/reverb effects unit that app\
roximates concert halls. Hall programs are characterized by pre-delay of up to 25 ms\
."),
quiz::Quiz("Hard Knee","In compression, refers to a more abrupt introduction of comp\
ression of the signal once the sound level crosses the threshold. (See also “Knee.”)\
"),
quiz::Quiz("Hard Sync","This is the most common type of oscillator sync where the sl\
ave oscillator will reset its waveform whenever it receives a sync pulse. If the typ\
e of sync is not specified, then it’s probably hard sync."),
quiz::Quiz("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 chan\
ging the instrument’s timbre."),
quiz::Quiz("Harmonic","A single harmonic is the purest sound possible: It contains n\
o 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 \
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causing the electrical vibrations of a voltage going up and down – is a sine wave. M\
ost of the time, overtones have a very specific pitch relationship to each other. Th\
e 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, a\
nd spaced out as integer multiples of the fundamental: two times its frequency, thre\
e times, four times, and so forth. The first few harmonics happen to have a nice mus\
ical spacing: an octave; an octave and a fifth; two octaves. But the higher they get\
, the less musical they may seem."),
quiz::Quiz("Harmonics","Whole number multiples of the fundamental frequency that occ\
ur naturally within the playing of a tone. Mathematically, if the fundamental freque\
ncy is x, the harmonics would be 2x, 3x, 4x, etc. For example, if the fundamental fr\
equency 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 timbr\
e of an instrument or voice."),
quiz::Quiz("Head","In tape recording, an electromagnetic transducer that magneticall\
y 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 purp\
oses. An erase head creates a powerful electromagnetic field to the tape to erase pr\
evious signals from the tape."),
quiz::Quiz("Headroom","The difference in dB between normal operating level and clipp\
ing level in an amplifier or audio device.  Also describes the difference in dB betw\
een the peak levels of a recording and the point at which the signal distorts. (Also\
 called “Margin.”)"),
quiz::Quiz("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."),
quiz::Quiz("Hertz","(Abbreviated Hz) 1) The unit of measurement for frequency, speci\
fically, the number of complete wave cycles that occur in a second (cycles per secon\
d). 1 Hz = 1 complete wave per second. 2) A popular rental car company (not typicall\
y used in recording except for transport to the studio)."),
quiz::Quiz("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."),
quiz::Quiz("Hi-Z","(abbreviated Hi-Z) Described as an impedance or resistance of sev\
eral thousand ohms. In microphones, Hi-Z is typically designated as 10,000 or more o\
hms. (See also “Impedance.”)"),
quiz::Quiz("High (gate)","When a gate signal is at the voltage level (typically 5 vo\
lts, although it can be more) that indicates it is “on” – such as when a note is bei\
ng held down on a keyboard controller – it is said that the gate is high."),
quiz::Quiz("High Impedance","(abbreviated Hi-Z) Described as an impedance or resista\
nce of several thousand ohms. In microphones, Hi-Z is typically designated as 10,000\
 or more ohms. (See also “Impedance.”)"),
quiz::Quiz("High Pass Filter","An audio filter that attenuates signals below a certa\
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in frequency (the cut-off frequency) and passes signals with frequencies that are hi\
gher."),
quiz::Quiz("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” i\
n an audio signal."),
quiz::Quiz("High-Pass Filter","The high pass filter (HPF) design passes harmonics ab\
ove its cutoff or corner frequency untouched, and reduces the level of lower harmoni\
cs depending on how far below the cutoff they are. In a 12dB/oct (decibel/octave) hi\
gh 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 bel\
ow the cutoff (one quarter the frequency) are reduced by 24dB, and so forth. High pa\
ss 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."),
quiz::Quiz("Horizontal Pitch","HP = Horizontal Pitch. In the Eurorack format for syn\
thesizer modules, the width of a module is defined as the number of hp (horizontal p\
itch) units. Each hp is 0.2” (0.5 cm). Most modules are even numbers of hp wide, alt\
hough some are odd numbers. Also, modules tend to be ever so slightly less than exac\
tly some multiple of 0.2” wide, just to make sure you don’t run into problems with e\
ver so slightly too wide modules overlapping."),
quiz::Quiz("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."),
quiz::Quiz("House Sync","A reference signal such as SMPTE time code that is used to \
keep all devices in the room in sync."),
quiz::Quiz("HP","HP = Horizontal Pitch. In the Eurorack format for synthesizer modul\
es, the width of a module is defined as the number of hp (horizontal pitch) units. E\
ach 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 multi\
ple of 0.2” wide, just to make sure you don’t run into problems with ever so slightl\
y too wide modules overlapping."),
quiz::Quiz("HPF","The high pass filter (HPF) design passes harmonics above its cutof\
f 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 filte\
r, harmonics one octave below the cutoff frequency (in other words, one half the cut\
off 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 ar\
e typically used to create bright sounds where the higher harmonics are much stronge\
r than the fundamental and lower harmonics - for example, the sound of a harpsichord\
."),
quiz::Quiz("Hum","1) The low-frequency pitch that occurs when power line current is \
accidently induced or fed into electronic equipment. The hum reflects the fundamenta\
l frequency of the current (60 Hz in the U.S., and 50 Hz in many European countries)\
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. 2) To vocalize a pitch without opening one’s mouth."),
quiz::Quiz("Hybrid Power Supply","A hybrid power supply uses a lower weight, more ef\
ficient switching power supply to perform most of the drop in voltage - say, from 12\
0v 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 modu\
lar synthesizer enclosures. Shortcomings with the power supply - too noisy, or not e\
nough - tend to be at the cause of many unexpected problems in modular synthesizers.\
"),
quiz::Quiz("Hypercardioid","A variation of the cardioid microphone pick up sensitivi\
ty pattern in which the shape of the optimal pickup area is tighter and more directi\
onal than cardioid. Hypercardioid microphones are most sensitive directly on-axis in\
 front of the microphone, and begins rejecting sounds between 90-150 degrees off-axi\
s, depending on the tightness of the pattern."),
quiz::Quiz("Hz/V","A system where a change of 1 volt at the input results in a chang\
e in pitch of a fixed number of hertz (cycles per second), rather than a fixed music\
al interval."),
quiz::Quiz("Hz","An abbreviation for the term Hertz, or the unit of frequency."),
quiz::Quiz("IADSR","This is an Attack/Decay/Sustain/Release (ADSR) envelope generato\
r that allows you to start the attack phase at an initial level - the “I” - rather t\
han the customary 0 volts. The envelopes in the Prophet VS, as well as a module from\
 Ladik, have this capability."),
quiz::Quiz("IC","Integrated Circuit - A miniature circuit of many components set on \
semiconductor material, used in electronics. A fancy term for “chip” or “microchip.”\
"),
quiz::Quiz("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 f\
ools our ears into thinking the sound source is in emanating from a specific point i\
nstead 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 com\
ing from performers on the stage, rather than from the speakers."),
quiz::Quiz("Impedance","Refers to the resistance of a circuit or device to alternati\
ng 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 to\
gether. Impedance is abbreviated by the letter Z, and measured in ohms (W)."),
quiz::Quiz("In Line Console","An audio mixing console that is designed and configure\
d so each channel strip can be used for both recording and monitoring functions duri\
ng multitrack recording. This configuration is in contrast to split mixing consoles,\
 which requires separate channels on the board for recording and monitoring function\
s."),
quiz::Quiz("In Phase","The desirable situation in which two or more devices (and the\
ir respective audio signals) are on the same side of the polarity spectrum, producin\
g waveforms that do not conflict or cancel each other out."),
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quiz::Quiz("In Port","A jack on a MIDI device or computer that will accept an incomi\
ng data signal."),
quiz::Quiz("Inductance","A characteristic of electrical conductors in which electric\
al charge (voltage) is produced or stored magnetically due to the natural resistance\
 to change in the electrical current. Inductance is an electromagnetic principle tha\
t 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)."),
quiz::Quiz("Inductor","A device (usually a coil of wire) that converts electrical en\
ergy into stored magnetic energy as electrical current passes through it. Commonly f\
ound in a variety of audio applications such as guitar pickups and loudspeakers."),
quiz::Quiz("Infinite Baffle","A loudspeaker mount or enclosure designed so that soun\
d waves coming from the front theoretically do not reach the back, preventing the so\
und waves from cancelling each other out. The term “infinite” comes from the idea th\
at 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 mo\
unting the speaker on a wall of an enclosed room, or building it inside a sealed cab\
inet large enough to prevent rear sounds from affecting the cone from the back."),
quiz::Quiz("Initial/Attack/Decay/Sustain/Release","This is an Attack/Decay/Sustain/R\
elease (ADSR) envelope generator that allows you to start the attack phase at an ini\
tial level – the “I” – rather than the customary 0 volts. The envelopes in the Proph\
et VS, as well as a module from Ladik, have this capability."),
quiz::Quiz("Input / Output (I/O)","I/O – An abbreviation for “Input/Output.” In audi\
o, it refers to any device, program or system involving the transferring of electric\
al/audio signals or data."),
quiz::Quiz("Input Impedance","The opposition to current flow by the first circuits o\
f a device."),
quiz::Quiz("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)."),
quiz::Quiz("Input","The jack or physical location where a device receives a signal. \
Also refers to the incoming signal itself."),
quiz::Quiz("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 outp\
ut. Commonly used to add processing such as reverb, compression or EQ to a channel o\
r group of channels."),
quiz::Quiz("Instrument Amplifier","A device that has a power amplifier and speaker t\
o reproduce the signal put out by an electric instrument."),
quiz::Quiz("Instrument Out Direct","Feeding the output of an electric instrument (li\
ke an electric guitar) directly to the recording console or tape recorder, as oppose\
d to miking the amplifier."),
quiz::Quiz("Insulator","A substance such as glass, air, plastic, etc., that will (fo\
r all practical purposes) not conduct electricity."),
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quiz::Quiz("Integrated Circuit","Integrated Circuit (Abbreviated “IC”) - A miniature\
 circuit of many components set on semiconductor material, used in electronics. A fa\
ncy term for “chip” or “microchip.”"),
quiz::Quiz("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 d\
etail on its uses) or less often as a lag generator or processor."),
quiz::Quiz("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 c\
omputer or other electronics. There are many examples of interfaces in professional \
audio situations, including MIDI (Musical Instrument Digital Interface); audio inter\
faces 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, m\
ix and playback. Even the mixing console is an interface of sorts, connecting the ma\
ny elements of the control room."),
quiz::Quiz("Intermodulation (IM) Distortion","Distortion caused by two or more audio\
 signals of different frequencies interacting with one another. The sum and differen\
ce of the frequencies produce new (usually unwanted frequencies) that didn’t exist i\
n any of the original frequencies."),
quiz::Quiz("Inverse Square Law","A mathematical rule that describes an inverse relat\
ionship between one quantity and the square of another quantity. In plain English, o\
ne 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 obst\
ructions, the sound pressure level will drop by half (6dB) each time the distance fr\
om the sound source is doubled. (This equation is quite useful to audio engineers tr\
ying to provide sound in open-air settings, for example.)"),
quiz::Quiz("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 (f\
or example) 0v becomes 5v and 5v becomes 0v. This is sometimes referred to as a pola\
rizer, as it changes the polarity (+ versus -) of a signal. A control voltage invert\
er is often combined with an offset voltage to adjust the output voltage into the de\
sired 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 vol\
tages, you would then need to add 8 volts to that result to get an upside-down (inve\
rted) envelope that still had an overall range of 0 to +8v."),
quiz::Quiz("Inverting Mixer","Most signal mixers make an effort to keep the same pol\
arity of a signal as it passes through the mixer. However, some mixers may invert th\
e 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 addin\
g one waveform or filter mode output out of phase with another coming from the same \
oscillator or filter."),
quiz::Quiz("Isolation","The process of containing sound within a certain area so tha\
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t 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 o\
ther sounds away from instrument mics during live recording sessions."),
quiz::Quiz("IV Cable","You often need to send one signal to multiple destinations. O\
ptions 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 intravenou\
s or “IV” drip), in addition to having 3.5mm plugs at either end."),
quiz::Quiz("Jack","That hole you plug your patch cables into on the face of your syn\
thesizer modules? That’s called a jack. The size and type of jack - 3.5mm, banana, o\
r 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, i\
t’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.)"),
quiz::Quiz("Jam Sync","A process available on some clock or syncing devices which re\
ads 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."),
quiz::Quiz("Karplus Strong","This is a physical modeling synthesis algorithm designe\
d 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, altho\
ugh it can be a high frequency chirp or other sound - is sent to both the output, an\
d 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 o\
f 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 cha\
llenge to patch yourself and play with the results."),
quiz::Quiz("Key","1) In music, the note scale in which a piece of music is written o\
r played, identified by the first note (tonic) of the scale, as in, “Key of C.” 2) T\
he control of a dynamics processing device by an external audio signal through the u\
se of a side chain. 3) A digital or data code that unlocks the use of a device or so\
ftware. Example: Pro Tools is licensed through an iLok ID via the use of a physical \
USB key."),
quiz::Quiz("Keyboard Controller","A piano-styled keyboard that sends out MIDI signal\
s to control other MIDI devices. Most keyboard instruments are equipped with MIDI co\
ntrol capabilities, but dedicated MIDI keyboard controllers emit no audio signals, o\
nly MIDI data."),
quiz::Quiz("Keyboard Tracking","Most modular synths follow a strict relationship bet\
ween 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” (con\
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trol voltage) to make it clear they are not attenuated or otherwise modified when co\
ntrolling a function on a module."),
quiz::Quiz("Keyboard","Any musical instrument or computer controlled by pressing a k\
ey."),
quiz::Quiz("Keytar","A strap-on, lightweight, portable keyboard meant to allow keybo\
ardists the same freedom (not to mention posturing opportunities) as guitarists."),
quiz::Quiz("Kick Drum","The bass drum on a trap drum set, so called because it is pl\
ayed with a kick pedal."),
quiz::Quiz("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 d\
escribing audio frequency ranges or digital sampling rates."),
quiz::Quiz("Knee","A function on a compressor that determines how abruptly or gradua\
lly 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” refe\
rs to an abrupt activation of the compressor, while “soft knee” refers to a more gra\
dual change."),
quiz::Quiz("Krell Patch","Recreating this patch is a challenge many modular musician\
s 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 no\
te has a random pitch, envelope, and duration."),
quiz::Quiz("Lag Generator","This function smoothes out an incoming signal so that th\
e change in voltage level cannot exceed a certain number of volts per second. This c\
auses the result to “lag behind” changes in the input. It is sometimes called a slew\
 limiter or technically as an integrator."),
quiz::Quiz("Layering","Refers to almost any blending of similar multiple musical par\
ts or sounds at once, often combined on one channel or assigned to one controller. I\
n audio recording, layering usually involves recording similar takes of the same ins\
trument or vocal (or duplicating parts with slight delays or chorusing effects) to c\
reate a fuller, richer sound than the vocal/instrument by itself. In sound design, i\
t also refers to blending multiple samples (example: two or more drum sounds) to cre\
ate a fuller sound."),
quiz::Quiz("Lead Sheet","A shorthand form of music notation (similar to a chord char\
t) that displays the basic essential elements of a song so musicians can follow alon\
g without the full notation of every note or expression. Lead sheets most commonly i\
nclude a melody line written in music notation with chord changes above the staff, a\
nd lyrics below it. (See also “Chord Chart.”)"),
quiz::Quiz("Leakage","Sounds from other instruments and sound sources that were not \
intended to be picked up by the microphone."),
quiz::Quiz("Level","The amount of signal strength; the amplitude, especially the ave\
rage amplitude."),
quiz::Quiz("LFO","This module produces repetitive, cycling waves ranging in frequenc\
y 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 lo\
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udness of a signal), vibrato (when controlling the pitch of a signal), repetitive fi\
lter wah-wah effects, pulse width modulation to vary the waveshape of a pulse in an \
oscillator, and more."),
quiz::Quiz("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 prev\
ent overload and signal peaking. A compressor effectively becomes a limiter when its\
 ratio is 10:1 or higher. (See also “Compressor.”)"),
quiz::Quiz("Line Input","Line Input (“Line In”) – An input designed to take a line l\
evel signal."),
quiz::Quiz("Line Level","Most consumer and lower-cost professional audio equipment u\
se a signal level reference known as line level or -10dBV (decibel volts). The most \
common connectors are RCA (phono) or 3.5mm, although 1/4” is also used; the signal i\
s “unbalanced” (it uses two wires: signal and ground). In the line level standard, a\
 sine wave that varies between +/-0.447 volts is considered to be at -10dBV. By cont\
rast, a typical oscillator signal in a modular synthesizer is +/-5 to +/-8 volts. As\
 a result, you will need either an output module in your modular synth or one heckuv\
a 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 sig\
nal to get it up to modular standards to process in your modular synth."),
quiz::Quiz("Line Output","Line Output (“Line Out”) – Any output that sends out a lin\
e level signal, such as the output of a console that feeds a recorder."),
quiz::Quiz("Linear FM","This is often the preferred input response for frequency mod\
ulating (FM’ing) an oscillator, as the result stays in tune while you change the mod\
ulator."),
quiz::Quiz("Linear Power Supply","A linear power supply design takes a higher incomi\
ng 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 signa\
l, at the cost of increased heat and weight (they're not very efficient). Many are m\
oving to a hybrid power supply that combines a switcher with a small linear supply o\
r regulator to get the best of both worlds."),
quiz::Quiz("Linear VCA","A linear voltage-controlled amplifier (VCA) uses a simple m\
athematical 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 lou\
d. 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 exponenti\
al output to a linear VCA, then you will get the desired aural result. Confusing? Th\
at’s why it’s great when an envelope generator or VCA has a switch or control to var\
y 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 swi\
ng as the input. Again, this is fine for altering control voltages, but not for mixi\
ng audio signals; in that case you want a mixer with exponential controls."),
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quiz::Quiz("Linear VCO","A linear voltage-controlled oscillator (VCO) follows the vo\
lts/hertz (v/Hz) standard; more common is the exponential volts/octave (v/oct) stand\
ard. The exception is frequency modulation (FM), where a linear control voltage inpu\
t is often preferred to recreate classic style FM as it does not change the fundamen\
tal pitch of the carrier oscillator."),
quiz::Quiz("Live Recording","A recording session where all the musicians are playing\
 at once with no overdubbing."),
quiz::Quiz("Live Room","The large, main room of the recording studio where most of t\
he instruments and/or vocalists perform. So called, not just because there is room f\
or live performances, but because the room has been acoustically treated to produce \
a pleasing amount of live reverberation."),
quiz::Quiz("Live","1) A term describing a space with a reverberant or reflected soun\
d. In a “live” space, the sound waves are active or “live.”  2) Occurring in real ti\
me, as opposed to previously recorded."),
quiz::Quiz("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 m\
odule when the keyboard is played; “Local Off” disconnects it. “Local Off” is freque\
ntly used to prevent unwanted looping of MIDI messages in some configurations, or wh\
en controlling the internal module via another controller."),
quiz::Quiz("Logic Functions","In a modular synth, control voltages tend to be contin\
uous in nature, while gate and trigger signals are binary: on or off; high or low. T\
his 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). Anoth\
er is AND: If and only if signal A and signal B are both, then output a high gate (o\
n); otherwise, output a low gate (off). These are great functions for combining beat\
 triggers from different timing sources."),
quiz::Quiz("Logic","Binary or Boolean logic is a way of combining gate signals (on o\
r off voltages) to create new outputs. Each section of a logic module typically incl\
udes 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 signa\
l if all of the inputs see “high” gate signals (i.e. input 1 and input 2 etc. all ha\
ve gate ons). Adding an “N” to the front of a function’s name says “not” this functi\
on - in other words, a NOR function would only output a high signal if all inputs we\
re low (not input 1 nor input 2 are high)."),
quiz::Quiz("Loop","1) Effectively, any piece of music or data that repeats endlessly\
. Before digital audio and sampling, loops were created by looping tape. Today, loop\
s 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) A\
nother term for antinode, or the points of maximum displacement of motion in a vibra\
ting stretched string or a sound wave.  (See also “Standing Wave.”)"),
quiz::Quiz("Looping","Sometimes it’s useful to have a module loop or repeat its func\
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tions. For example, an envelope generator that can be set to loop becomes a low freq\
uency oscillator: as it attacks to a maximum value and decays back to zero, it start\
s that attack phase again. Quite often you want a note sequencer to loop: When it re\
aches 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 ar\
e also popular for live performance."),
quiz::Quiz("Loudness","A term referring to how the human ear perceives incoming soun\
d 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 relationshi\
p. In acoustic terms, volume is more about the amplitude of the sound waves, while l\
oudness describes how our ears hear the intensity of those waves."),
quiz::Quiz("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” o\
r “on” signal of +5 volts)."),
quiz::Quiz("Low Frequency Oscillator","This module produces repetitive, cycling wave\
s ranging in frequency from the low end of the audio spectrum to as slow as many sec\
onds or even minutes per cycle. They are used to produce effects such as tremolo (wh\
en controlling the loudness of a signal), vibrato (when controlling the pitch of a s\
ignal), repetitive filter wah-wah effects, pulse width modulation to vary the wavesh\
ape of a pulse in an oscillator, and more."),
quiz::Quiz("Low Impedance","(abbreviated Lo-Z) Described as impedance of 500 ohms or\
 less. (See also “Impedance.”)"),
quiz::Quiz("Low Pass Filter","The low pass filter (LPF) design passes harmonics belo\
w 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 cu\
toff (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 w\
eaker high harmonics – especially as a note fades to silence."),
quiz::Quiz("Low Pass Gate","By strict definition, a low pass gate (LPG) is a low pas\
s filter whose cutoff frequency goes down into the subsonic range as its control vol\
tage 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 ei\
ther 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."),
quiz::Quiz("Low-Frequency Oscillator (LFO)","A circuit that emits low-frequency elec\
tronic 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 paramet\
ers in the audio signal, such as pitch or volume. LFOs are frequently used in sample\
rs, synthesizers and signal processors to create such effects as vibrato, tremolo, a\
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nd phasing."),
quiz::Quiz("low-pass-filter","An audio filter or device that attenuates signals abov\
e a certain frequency (the cut-off frequency) and passes signals with frequencies th\
at are lower than the cut-off."),
quiz::Quiz("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 lo\
ws” in an audio signal."),
quiz::Quiz("LPF","The low pass filter (LPF) design passes harmonics below its cutoff\
 or corner frequency untouched, and reduces the level of lower harmonics depending o\
n 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 freq\
uency) are reduced in level by 12 dB; harmonics two octaves above the cutoff (four t\
imes 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 h\
armonics - especially as a note fades to silence."),
quiz::Quiz("LPG","By strict definition, a low pass gate (LPG) is a low pass filter w\
hose 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. S\
ome 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 (hi\
gher harmonics are filtered more) as it falls to silence. This mimics the way many n\
atural sounds work."),
quiz::Quiz("M2.5","A common screw thread size used to mount Eurorack modules. This s\
ize is most common when using a system of loose nuts that slide along the rails that\
 the modules are attached to."),
quiz::Quiz("M3","A common screw thread size used to mount Eurorack modules. This siz\
e is most common when using module mounting rails that have been pre-drilled."),
quiz::Quiz("Magnetic Tape","Recording tape consisting of a plastic strip coated by m\
agnetic materials, finely ground iron oxide (rust) particles. Commonly used for anal\
og recording."),
quiz::Quiz("Magnetism","A natural attractive energy of iron based-materials toward o\
ther iron-based materials."),
quiz::Quiz("MArF","The rare Buchla Model 248 MArF (Multiple Arbitrary Function Gener\
ator) is a cross between a sequencer and an envelope generator (both described elsew\
here in this glossary) in that it typically contains 16 or 32 stages (sometimes refe\
rred 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 creat\
ed. Later on, Buchla used the term MARF to describe the multi-step envelopes in inst\
ruments such as the Buchla 400."),
quiz::Quiz("Margin","See “Headroom.”"),
quiz::Quiz("Masking","The characteristic of hearing by which loud sounds prevent the\
 ear from hearing softer sounds of similar frequency. Also refers to the obscuring o\
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f softer sounds by louder ones."),
quiz::Quiz("Master","1) The main output control of a console or DAW, setting the lev\
el of the mixed signal as it leaves the console. (Also called “master fader.”) 2) Th\
e final-mixed original recording from which copies are made."),
quiz::Quiz("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."),
quiz::Quiz("Measure","The grouping of a number of beats in music. (See also “Bar.”)"\
),
quiz::Quiz("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 (1,000,0\
00 bytes, MB)."),
quiz::Quiz("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 t\
he number of beats in a measure."),
quiz::Quiz("Mic / Line Switch","Mic, Mike – Abbreviations for “microphone.”"),
quiz::Quiz("Microphone (Mic) Input","The input of a console or other device designat\
ed for a microphone signal."),
quiz::Quiz("Microphone (Mic) Level","The very low audio voltage level emitted by a s\
tudio microphone. The signal must go through a preamplifier to be increased to line \
level before entering the console. (See also “Line Level,” “Preamplifier.”)"),
quiz::Quiz("Microphone (Mic) Pad","A setting on a microphone or preamp, or a separat\
e adapter/connector, that reduces the level of the microphone signal before it enter\
s the preamplifier to prevent overload."),
quiz::Quiz("Microphone","A transducer which converts sound pressure waves into elect\
rical signals."),
quiz::Quiz("Mid-Side Miking (M/S)","(Abbreviated M/S) A stereo coincident microphone\
 placement technique in which one cardioid pattern microphone is aimed directly at t\
he sound source, and a bi-directional microphone placed sideways and as close as pos\
sible to the first mic."),
quiz::Quiz("MIDI Clock","A clock signal conveyed by MIDI that is used by the connect\
ed sequencers and musical devices to stay in sync with one another. Not to be confus\
ed with MIDI time code (MTC), MIDI clock is tied to the Beats-Per-Minute (BPM) tempo\
, advancing 24 steps per quarter note."),
quiz::Quiz("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."),
quiz::Quiz("MIDI Interface","A device that converts a MIDI signal into the digital f\
ormat of a computer so it can store and use the MIDI signal."),
quiz::Quiz("MIDI over Bluetooth","Bluetooth Low Energy (BLE) is a wireless connectio\
n specification supported by the majority of mobile computing devices. BLE (also cal\
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led Bluetooth SMART) can extend battery life for mobile devices using connected acce\
ssories (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 f\
rom different manufacturers to interoperate. The Specification for MIDI over Bluetoo\
th Low Energy (BLE-MIDI) is based on Apple's implementation which appeared in iOS8 a\
nd OSX 10.10, so that products from early adopters would remain compatible with the \
industry standard."),
quiz::Quiz("MIDI Sample Dump Standard (SDS)","A sub-protocol that was added into MID\
I to enable the transfer of digitally recorded samples between instruments, storage \
units or sound modules without converting them to analog."),
quiz::Quiz("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 serie\
s of timed steps. MIDI sequencers can exist on board MIDI controllers, keyboards or \
workstations, as standalone devices, or as computer software."),
quiz::Quiz("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 al\
ternative to “daisy chaining” devices, which can cause slight delays in the MIDI sig\
nal."),
quiz::Quiz("MIDI Thru","A port that puts out a MIDI signal that is the same as the i\
ncoming MIDI signal, effectively relaying the signal to another device without alter\
ing or changing it.  (Many MIDI devices have three MIDI ports: In, Out and Thru.)"),
quiz::Quiz("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 SMT\
PE code (usually in relation to video). (See also “SMPTE Time Code.”)"),
quiz::Quiz("MIDI","Short for Musical Instrument Digital Interface. MIDI is a common \
language to connect one synthesizer to another, and synthesizers to a computer. Alth\
ough 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 connect\
or is required: the traditional 5-pin DIN, or a USB computer-style connection."),
quiz::Quiz("Mids","Abbreviation for “mid-range frequencies,” the audio frequencies f\
rom about 250 Hz through 6000 Hz. Meant in the context of “highs, mids and lows” in \
an audio signal."),
quiz::Quiz("Mini Keys","A number of keyboard controllers and even keyboard synths us\
e a key size that is much smaller than a typical piano key. Mini keys is the term co\
mmonly used (sometimes derisively, although the space and cost savings can be quite \
significant) to refer to this hardware choice."),
quiz::Quiz("Mix Down","Mixdown or Mix Down – The processes of creating a final mix b\
y 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”\
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).") ,
quiz::Quiz("Mix","1) The blending of audio signals together into one composite signa\
l. 2) Can also refer to the blending of a portion of an effected audio signal back i\
nto the direct signal."),
quiz::Quiz("Mixer","This module combines signals together. You may use a mixer to co\
mbine 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 wa\
nt linear level controls plus additional functions to invert and offset the voltages\
 going through it."),
quiz::Quiz("Modular","A modular synth breaks down the main components of a synthesiz\
er - the tone-generating oscillators, the tone-modifying filters, the amplitude-shap\
ing VCAs, and the modulation sources that create envelopes, tremolos, and more - int\
o individual modules you can purchase and install. At the most basic level, this all\
ows you to play mix-and-match in building your own custom synth."),
quiz::Quiz("Modulation Noise","Noise that is present only when the audio signal is p\
resent."),
quiz::Quiz("Modulation","When you vary a parameter of a synthesizer module using vol\
tage 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 w\
ah-wah effect, it is said that the LFO is modulating the cutoff. When an envelope ge\
nerator causes a voltage controlled amplifier (VCA) to open up to allow a sound to b\
ecome suddenly loud, and then fades it back down to silence, you can also say the en\
velope 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."),
quiz::Quiz("Modulator","We touched on the general subject of modulation and modulato\
rs in the definition above. However, quite often when someone uses the term modulato\
r, they're usually discussing a synthesis techniques where one usually audio-rate si\
gnal “modulates” (varies) another audio signal. For example, in frequency modulation\
 (FM) synthesis, the modulator (or modulating oscillator) varies the frequency (pitc\
h) of the main signal generator (oscillator), called the carrier. In ring, balanced,\
 or amplitude modulation, the modulator is varying the loudness of the carrier signa\
l. So the term modulator is a way to make it clear which component you're talking ab\
out in one of these patches: not the main tone generator, but the module that is dri\
ving that generator crazy."),
quiz::Quiz("Module","A self-contained group of circuits and controls. In the recordi\
ng studio, modules are often contained in interchangeable housing for installation o\
n 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 modul\
es."),
quiz::Quiz("Monaural (Mono)","(Abbreviated “Mono”) Describing an audio signal coming\
 through a single, as opposed to stereo, which is two channels. (See also “Monophoni\
c.”)"),
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quiz::Quiz("Monitor Mix","A mix of the live and/or recorded audio signals that is fe\
d to the musicians so the 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 (oft\
en 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 se\
ttings, the monitor mix is often controlled by a separate audio engineer running a s\
eparate sound board."),
quiz::Quiz("Monitor Mixer Section","Monitor Section/Monitor Mixer Section - The sect\
ion of the console that is used to create a rough mix so the engineer can hear what \
is being recorded without effecting the levels being fed to the multitrack recorder \
or DAW."),
quiz::Quiz("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.”)"),
quiz::Quiz("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 t\
o 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 performe\
rs to hear a mix of the tracks while they are performing."),
quiz::Quiz("Monophonic","(Abbreviated “Mono”) 1) A single sound source or single-cha\
nnel 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 play\
s one pitch (or “voice”) at a time. (See also “Voice.”)"),
quiz::Quiz("Morphing","In the context of a modular synth, morphing refers to an osci\
llator 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 digit\
al oscillators which internally crossfade (or in some cases, switch) from one wavesh\
ape to another, although it is sometimes applied to analog oscillators that give you\
 real time control over waveshapes."),
quiz::Quiz("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-sty\
le transistor ladder filter that can be low pass or high pass, and one AD or AR enve\
lope generator. It also has a very capable step sequencer plus a miniature one-octav\
e 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 exp\
and a basic modular system. I have an online introductory course to the Mother-32 co\
ming out this spring, and will have a course plus ongoing weekly series on adding di\
fferent modules to this starter system."),
quiz::Quiz("Moving Coil Microphone","A microphone in which sound pressure waves are \
converted to an electrical audio signal by an induction coil moving within a magneti\
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c field—a process often compared to a loudspeaker working in reverse. Dynamic microp\
hones are less sensitive than condenser microphones, but can be effective for miking\
 louder sound sources or for close-miking applications."),
quiz::Quiz("Moving Fader Automation","A feature in some consoles in which fader chan\
ges can be pre-programmed to occur automatically during playback of a multitrack rec\
ording."),
quiz::Quiz("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 fo\
llowed in their footsteps, including Synthesizers.com (Dotcom) and Moon Modular. You\
 will sometimes hear this used interchangeably with MU for Moog Units, which also re\
fers to a standardized width of 2.125” (5.4 cm) wide per MU. Given that this standar\
d is both historical and physically large, some users “5U” as a badge of honor that \
they’re traditional and cool. (And the are.) There was also a briefly popular 5U for\
mat from MOTM that used a different width and power connection. It has since been di\
scontinued, but there are still diehard MOTM format users today."),
quiz::Quiz("Multi-Tap Delay","A delay works by in essence putting audio in one end o\
f 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 wh\
en 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."),
quiz::Quiz("Multimeter","A small device that tests electrical voltage, current, and \
resistance. Multimeters are useful in recording studios for calibrating electrical s\
ystems and troubleshooting problems."),
quiz::Quiz("Multiple Arbitrary Function Generator","The rare Buchla Model 248 MArF (\
Multiple Arbitrary Function Generator) is a cross between a sequencer and an envelop\
e generator (both described elsewhere in this glossary) in that it typically contain\
s 16 or 32 stages (sometimes referred to as “segments”), and a rate control to inter\
polate between these stages. This means very complex envelope shapes and other contr\
ol 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."),
quiz::Quiz("Multiple","Quite often you need to split or copy a signal to send to mor\
e than one destination. This is commonly done with a multiple (“mult” for short) whe\
re you plug one source in, and then plug in additional patch cables to go off to mul\
tiple destinations."),
quiz::Quiz("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 possiblity of one input being switched between mult\
iple outputs. A sequential switch is a type of multiplexor, as it chooses among mult\
iple inputs to decide which one to send to the output (or the other way around). The\
re are some modules that do this at audio rate, using an oscillator's output to swit\
ch between variations of another waveshape to create complex, chopped mixtures of th\
ose waveforms."),
quiz::Quiz("Multitimbral","Refers to the ability of a synthesizer or module to play \
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several different sounds, patches or “timbres” at once."),
quiz::Quiz("Multitrack Recording","Also called tracking or multitracking) The heartb\
eat of the recording studio, multitrack recording is process of recording a collecti\
ve of sound sources onto separate tracks, each with its own audio channel, then comb\
ining 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 o\
nto multitrack analog tape, but today they can also be recorded digitally as separat\
e audio files into a digital audio workstation (DAW)."),
quiz::Quiz("Multitrack Tape","A piece/reel of magnetic tape which can be used to sto\
re two or more discrete signals in sync with each other."),
quiz::Quiz("Musical Instrument Digital Interface (MIDI)","Short for Musical Instrume\
nt Digital Interface. MIDI is a common language to connect one synthesizer to anothe\
r, 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 t\
o watch out for is what type of connector is required: the traditional 5-pin DIN, or\
 a USB computer-style connection."),
quiz::Quiz("Mute Switch","A switch on a console or other piece of audio equipment th\
at 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 ca\
n cut the signal for that channel."),
quiz::Quiz("Mute","Sometimes you need to silence or disconnect a signal. A circuit t\
hat allows you to do so is called a mute."),
quiz::Quiz("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 us\
ed instead. Nanowebers per meter of tape effectively describes the signal strength t\
hat is being recorded to tape."),
quiz::Quiz("Narrowband Noise","Noise (random energy) that occurs over a limited freq\
uency range."),
quiz::Quiz("Near Field","The area between 1-5 feet from the sound source. Studio mon\
itors are generally considered “near-field” speakers because they are meant to be li\
stened to at close range. (See also “Far Field.”)"),
quiz::Quiz("Near-Coincident Miking","A stereo miking technique in which two micropho\
nes are placed near each other at an outward angle to create a stereo image (as oppo\
sed 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).\
"),
quiz::Quiz("Negative Feedback","A portion of the output signal that is fed back to t\
he input of an amplifier with its phase inverted from the original output signal. Th\
is has a dampening effect on the output, effectively cancelling out a portion of the\
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 volume."),
quiz::Quiz("Noise Floor","The level of the noise present below the audio signal, mea\
sured in dB. Every electronic device emits a minimum level of noise, even when no au\
dio is traveling through it; this is described as its noise floor. Generally speakin\
g, 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 record\
ing is the sum of all the noise generated by connected devices. The objective is alw\
ays to keep the noise floor as low as possible."),
quiz::Quiz("Noise Gate","A gate that is used reduce audible noise by automatically t\
urning off an audio channel when the signal is not present."),
quiz::Quiz("Noise Reduction","Any of a number of processes to remove noise from a si\
gnal, device or system."),
quiz::Quiz("Noise","Describes any unpleasant, objectionable or unintended sound freq\
uencies 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 ot\
herwise silent passages. (See also “Noise Floor.”) Bad connections, improper groundi\
ng, radio interference and other issues can also cause introduce noise into the sign\
al. Engineers may also deliberately run a noise signal through a sound system for te\
sting purposes. (See also “White Noise, “Pink Noise.”)"),
quiz::Quiz("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 ve\
rsion at any time. (Sometimes also called “Nonlinear editing.”)"),
quiz::Quiz("Nondirectional","In microphones, picking up evenly from all directions."\
),
quiz::Quiz("Normalize","To apply a fixed amount of gain to audio so that the highest\
 peak is set at the highest acceptable recording level."),
quiz::Quiz("Normalled","The power of modular synthesizers is that you can patch a si\
gnal to flow the way you prefer through your system. This can also be a time-consumi\
ng 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 conn\
ections 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 con\
nections are often referred to as being normalled. For example: An internal noise so\
urce may normally be connected to one channel of a mixer that appears before the fil\
ter, but if you insert a patch cable into a jack usually labeled external input, thi\
s “normalled” connection is broken and replaced by your external connection."),
quiz::Quiz("Notch Filter","This is a particular type of filter mode where audio freq\
uencies 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, whic\
h only passes harmonics around the cutoff frequency. Notch filters tend to have a su\
btle effect on the sound; moving (modulating) the cutoff frequency can result in a w\
eak phasing sort of sound. Notch filters are often used in sound systems to weaken o\
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r 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 fil\
ter in a patch to hollow out a sound, leaving room in the harmonic spectrum for othe\
r sounds to exist with less competition, or just to create a sound more likely to ca\
tch the ear because something that is expected is instead missing."),
quiz::Quiz("Notch","A narrow band of audio frequencies."),
quiz::Quiz("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 t\
o tape."),
quiz::Quiz("Nybble","Nybble (or Nibble) - One half byte of computer data, or 4 bits.\
"),
quiz::Quiz("Nyquist Frequency","In digital recording, the highest frequency that can\
 be recorded and reproduced properly, equivalent to a one-half the sampling rate. (F\
or example, with the common sampling rate of 44,100 kHz per second, the Nyquist freq\
uency would be 22,050 kHz.) Aliasing begins to occur with frequencies that exceed th\
is threshold. (See also “Aliasing.”)"),
quiz::Quiz("Nyquist Rate","he lowest sampling rate that can be used to record and re\
produce a given audio signal, equivalent to twice the highest frequency. If the high\
est frequency found in an analog signal or sound is 18,000 kHz, theoretically the si\
gnal 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 inver\
se principle of the Nyquist Frequency. (NOTE: the sample rate of 44,100 kHz/second i\
s considered the standard sample rate because it easily covers the upper range of hu\
man hearing, which is about 20,000 kHz.)"),
quiz::Quiz("Octave Divider","A module that creates a new tone one or two octaves bel\
ow the fundamental harmonic - the “pitch” - of the sound coming into it, to emphasiz\
e the bass. Sometimes also known as a suboctave or sub bass function."),
quiz::Quiz("Octave","An octave is a typical musical internal. For example, all of th\
e “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 su\
btracting 1 volt to get a one octave change in pitch; some oscillators also have oct\
ave 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 oc\
taves). Suboctave or subharmonic generators divide the input pitch by 2 or 4 to crea\
te new waveforms that are one or two octaves lower in pitch, which adds bass."),
quiz::Quiz("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-a\
xis.)"),
quiz::Quiz("Offset Time","1) The SMPTE time that will trigger a MIDI sequencer to be\
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gin. 2) The amount of position difference needed to get two reels to play the music \
in time."),
quiz::Quiz("Offset","In simple terms, Offset modules usually add or subtract a volta\
ge from a signal passing through - such as shifting a 0 to +10v signal to instead va\
ry between -5 and +5 volts."),
quiz::Quiz("Ohm's Law","The mathematical relationship between voltage, current and r\
esistance."),
quiz::Quiz("Ohm","The unit used to measure the amount of opposition (impedance) to e\
lectrical current flow in a signal or device. (See also “Impedance.”)"),
quiz::Quiz("Omni Mode","A setting that enables a MIDI device to recognize and respon\
d to all MIDI channels at once."),
quiz::Quiz("Omni","A prefix meaning “all.”"),
quiz::Quiz("Omnidirectional Pattern","In microphones, picking up evenly from all dir\
ections (sometimes also called “Nondirectional”). 2) In speakers, sending out the si\
gnal evenly in all directions."),
quiz::Quiz("On Axis","The position directly in front of the diaphragm of a microphon\
e, in line with its movement."),
quiz::Quiz("Open Circuit","An electrical circuit that is disconnected, interrupted o\
r incomplete, preventing the flow of electricity."),
quiz::Quiz("Operating Level","(Sometimes called “Reference Level“) The maximum level\
 that should not be exceeded in normal operation."),
quiz::Quiz("Operational Amplifier","(Abbreviated “Op Amp“) An amplifying circuit use\
d in most audio and electronic devices."),
quiz::Quiz("Operational Transconductance Amplifier","An OTA (operational transconduc\
tance 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, yo\
u 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 voltag\
e to resistance device that makes it possible to add voltage control to circuits suc\
h as filters. In general, when someone touts they have an OTA based filter, they usu\
ally mean it has a “warm” sound...unless it’s an MS-20 filter clone, in which case i\
t’s thinner and more edgy. In reality, using an OTA is more about convenience of des\
ign than creating a specific sound."),
quiz::Quiz("Operator","There are a few different synthesis techniques where one usua\
lly audio-rate signal does something to another audio signal. For example, in freque\
ncy modulation (FM), a second signal (called the modulator) varies the frequency (pi\
tch) of the main signal, called the carrier. These two signals or oscillators are of\
ten 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."),
quiz::Quiz("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 \
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or input 2 or input 3, etc.), to output a gate on signal. A NOR function has an inve\
rted output: it would only be on (high) if all inputs were low (not input 1 nor inpu\
t 2 are high). An XOR (Exclusive OR) would only output a high signal if one of the i\
nputs was high, but not if both inputs were high (or low). Finally, an XNOR is the i\
nvert of an XOR function."),
quiz::Quiz("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 oscill\
ator because its output varies up and down (oscillates) in voltage in a repeating pa\
ttern. This pattern is referred to as its waveshape (such as a square wave, that alt\
ernates 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 t\
hat vibrates back and forth on a guitar, a drum head that vibrates up and down, or t\
he vibrations in the reed of a woodwind instrument. The vibrations of a modular synt\
h’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.)"),
quiz::Quiz("Oscilloscope","This is a piece of test equipment that displays voltage f\
luctuations as graphical waveforms. A ’scope can run at a wide range of frequencies,\
 displaying slowly changing voltages like LFOs or envelopes, or quickly changing vol\
tages 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 jo\
b to your computer, or scopes as modules."),
quiz::Quiz("OTA","An OTA (operational transconductance amplifier) circuit is one tha\
t 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, an\
d is also known for a “warm” sound. Therefore, you may find it in VCAs (voltage cont\
rolled 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 i\
t possible to add voltage control to circuits such as filters. In general, when some\
one touts they have an OTA based filter, they usually mean it has a “warm” sound...u\
nless it’s an MS-20 filter clone, in which case it’s thinner and more edgy. In reali\
ty, using an OTA is more about convenience of design than creating a specific sound.\
"),
quiz::Quiz("Out of Phase","1) Being similar to another signal in amplitude, frequenc\
y and wave shape but being offset in time by part of a cycle. 2) Having the opposite\
 polarity."),
quiz::Quiz("Outboard Equipment","Equipment that is used with, but is not a part of, \
a console."),
quiz::Quiz("Output Impedance","The opposition to the flow of electrical current by t\
he output circuits of an amplifier (or other device)."),
quiz::Quiz("Output Level","The signal level at the output of a device."),
quiz::Quiz("Output","1) The jack or physical location of where a device sends out a \
signal. 2) The signal put out by a device."),
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quiz::Quiz("Overdubbing","The process of recording an additional musical performance\
 over an existing recording, usually on its own track. Overdubbing has become a comm\
on recording technique with the advent of multitrack recording, first on multitrack \
analog tape, and more recently via computers and Digital Audio Workstations (DAWs)."\
),
quiz::Quiz("Overtone","Any harmonic in a tone except the fundamental frequency. (See\
 also “Partial.”)"),
quiz::Quiz("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 somet\
imes 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."),
quiz::Quiz("Pan (Panning)","The process of “placing” a particular sound within the s\
tereo field. This is accomplished by controlling the balance of the signal between t\
he left and right speakers so the ear hears the sound as coming from a particular po\
int 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."),
quiz::Quiz("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 ster\
eo (or surround) space by controlling how much of the signal is sent to each speaker\
 or channel."),
quiz::Quiz("Parallel Jacks","Several jacks that are wired so that each connection is\
 wired to the corresponding connection of other jacks."),
quiz::Quiz("Parallel Port","A connector that is able to transmit and receive digital\
 data at the same time though different pins."),
quiz::Quiz("Parameter","Parameter is the fancy name given to any value or property o\
r control of a synthesizer module that you’re trying to change. For example, an osci\
llator’s parameters typically include its pitch and the width of its pulse wave. A f\
ilter’s parameter will include its cutoff frequency (pitch), the amount of resonance\
 (feedback), and possibly other controls such as a blend between its different outpu\
ts. Parameter was a popular term to describe a value you could change in software, a\
nd it’s been carried over by some to hardware modular synths."),
quiz::Quiz("Parametric Equalization","An equalizer in which all parameters of equali\
zation can be adjusted to any amount, including the center frequency, the amount of \
boost or cut, and the bandwidth."),
quiz::Quiz("Paraphonic","A paraphonic synth is one where all of the notes being play\
ed go through a single filter (VCF) and amplifier (VCA). This was a popular scheme i\
n the early days of polyphonic synths in that a separate oscillator (or organ-like f\
requency divider, in the case of “string synths” and the such) was used for each not\
e played, but they were mixed before all going to the filter and amp to articulate t\
he note(s). It was not uncommon for some monophonic synths to allow two to four inde\
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pendent notes to independently control the pitch of its oscillators, while still goi\
ng 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 n\
otes will be re-articulated together."),
quiz::Quiz("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."),
quiz::Quiz("Pass Band","The frequency range of signals that will be “passed” by a fi\
lter, rather than reduced."),
quiz::Quiz("Passive Device","A component that does not generate or control electrica\
l current (as opposed to an “Active Device”). In audio applications, this usually re\
fers to a piece of gear that does not include an amplifier as part of its design. Fo\
r example, active speakers are self-powered, while passive speakers require an exter\
nal amplifier in order to reproduce sound. (See also “Active Device.”)"),
quiz::Quiz("Passive","Means no active (i.e. connected to a power supply) electronics\
 are involved – such as sending a signal straight through a potentiometer control, i\
nstead of using op amps and other electronics to create a mixer circuit around it. P\
assive is cheap and easy, and does not add noise to a signal. But passive electronic\
s cannot buffer one signal from another (meaning they might interact in undesirable \
ways), and cannot boost, offset, or invert a signal."),
quiz::Quiz("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 an\
d outputs of the console and components in the studio, used for the purpose of organ\
izing, managing and regulating signal flow."),
quiz::Quiz("Patch Cable","The cables used to connect together the different inputs a\
nd 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 physical\
ly connect two callers together using electrical cables. As different modular format\
s often use different connector standards, you need to make sure the connectors at t\
he 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)."),
quiz::Quiz("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 us\
ed to make connections in the patch bay (hence the name); however, patch cords facil\
itate 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."),
quiz::Quiz("Patch Field","A panel or component containing a series of jacks with con\
nections for most of the inputs and outputs of the console and components in the stu\
dio, used for the purpose of organizing, managing and regulating signal flow."),
quiz::Quiz("Patch Librarian","A computer program allowing for the storing of sound p\
atches outside of a synthesizer via MIDI."),
quiz::Quiz("Patch Panel","A panel or component containing a series of jacks with con\
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nections for most of the inputs and outputs of the console and components in the stu\
dio, used for the purpose of organizing, managing and regulating signal flow."),
quiz::Quiz("Patch","The shorthand term used to refer how a series of modules are int\
erconnected to create a sound, derived from the fact that patch cords are used to co\
nnect the modules together. 1) To route or reroute the signal in an audio system (su\
ch as a console) by using short cables with plugs inserted into jacks. 2) A sound se\
tting or program on a synthesizer."),
quiz::Quiz("Path","Short for Signal Path, the way in which current does or may trave\
l in a circuit or through a device."),
quiz::Quiz("PCM","Pulse Code Modulation - A process by which analog signals are tran\
slated to digital code. This is done by taking samples of the amplitude of the analo\
g signal at regular rapid intervals, then translating it into binary numbers as a di\
gital 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 a\
udio."),
quiz::Quiz("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 di\
fferences to avoid problems with the patent used by Yamaha’s FM synths of the era. I\
ntriguingly, it did a good job at mimicking many “analog” synth effects including th\
e sound of a resonant filter."),
quiz::Quiz("Peak Filter","An EQ circuit/filter that boosts or cuts the middle (cente\
r frequencies in an audio signal, as opposed to high-pass or low-pass filters. (NOT \
to be confused with amplitude peaks.)"),
quiz::Quiz("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 Mete\
r.”)"),
quiz::Quiz("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 wav\
e. (See also “Peak Value.”)"),
quiz::Quiz("Peak Value","eak Value (also called Peak Level) – The measure of the max\
imum positive or negative value (amplitude) of a waveform at any moment. In audio, t\
his is visually depicted as the farthest point of the waveform above or below the ze\
ro axis."),
quiz::Quiz("Pedal Board","A board with several guitar pedals attached and inter-conn\
ected so that a guitar player can conveniently activate a number of different effect\
s."),
quiz::Quiz("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."),
quiz::Quiz("Phase Addition","The increased audio energy that happens when waveforms \
are in similar phase relationships, resulting in an increase in volume up to twice w\
hat it should be."),
quiz::Quiz("Phase Cancellation","The opposite of phase addition, this is the reducti\
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on of energy that occurs when two similar waveforms that are out of phase with one a\
nother and begin cancelling each other out, either greatly reducing or eliminating t\
he 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."),
quiz::Quiz("Phase Distortion Synthesis","Phase Distortion synthesis was used by Casi\
o originally in the 80s in the CZ line of synths. It is related to FM (frequency mod\
ulation), 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" sy\
nth effects including the sound of a resonant filter."),
quiz::Quiz("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 ad\
ding harmonics to the sound."),
quiz::Quiz("Phase Lock","Any of a number of processes used to help synchronize signa\
ls or devices by correcting phase differences. For example, in analog tape machines,\
 phase locking helps to keep multiple machines synced together by sensing phase diff\
erences in the playback of pilot tunes by the two machines and adjusting the speed t\
o eliminate the phase difference. In synthesizers, phase locking controls one tone g\
enerator so that it begins its waveform in phase with the signal from another tone g\
enerator. Phase-locked loops (PLL) are reference signals used in the clock functions\
 of electronic devices."),
quiz::Quiz("Phase Locked Loop","A phase locked loop is, in essence, an oscillator th\
at 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 tha\
t 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 incom\
ing pitch - perhaps to drive a special circuit such as a switched-capacitor filter."\
),
quiz::Quiz("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 ba\
ck 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 wavefor\
m. 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."),
quiz::Quiz("Phase Reversal","A change in a circuit to get the waveform to shift by 1\
80 degrees."),
quiz::Quiz("Phase Shift","A delay introduced into an audio signal measured in degree\
s delayed."),
quiz::Quiz("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 harmo\
nics 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, an\
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d different harmonic components of the original sound cancel each other out (see Pha\
se), 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."),
quiz::Quiz("Phase-Locked Loop","PLL -  Any of a number of processes used to help syn\
chronize signals or devices by correcting phase differences. For example, in analog \
tape machines, phase locking helps to keep multiple machines synced together by sens\
ing phase differences in the playback of pilot tunes by the two machines and adjusti\
ng the speed to eliminate the phase difference. In synthesizers, phase locking contr\
ols 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 c\
lock functions of electronic devices."),
quiz::Quiz("Phase","A measurement (expressed in degrees) of the time difference betw\
een 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) c\
an be defined by the phase. For example, if you are one-quarter of the way through a\
 waveform’s cycle, your phase is 90°."),
quiz::Quiz("Phasing","An effects sound created by varying the phase shift of an audi\
o signal, then mixing it with the direct signal."),
quiz::Quiz("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 i\
t is perceived to be as loud as a 1000-Hz tone playing at 60dB."),
quiz::Quiz("Phone Plug","A plug (or its mating jack) with a diameter of 1/4 inch and\
 a length of I 1/4 inches used for interconnecting audio."),
quiz::Quiz("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."),
quiz::Quiz("Physical Modeling","One approach to (often digital) synthesis is to recr\
eate 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 t\
o create sounds. There are a handful of modules available which perform this modelin\
g to create their sounds."),
quiz::Quiz("Pickup Pattern","The shape of the area in front of or around the microph\
one from where it evenly picks up sound. Many use this term interchangeably with “po\
lar pattern,” but a polar pattern gives more detail about microphone sensitivity. (S\
ee also “Polar Pattern.)"),
quiz::Quiz("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 “Conta\
ct Microphone.”"),
quiz::Quiz("Pinch Roller","A rubber (or plastic) wheel on a tape recorder that pinch\
es the tape between it and the capstan, allowing the capstan to pull the tape."),
quiz::Quiz("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 an\
alog by playing the tracks through the console and recording them onto separate trac\
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ks, or digitally through a digital audio workstation. Bouncing was once used frequen\
tly by engineers to free up additional tracks for recording, but in digital workstat\
ions where tracks are virtually unlimited, this practice is basically obsolete. Toda\
y, engineers typically bounce tracks for the purpose of creating a preliminary or fi\
nal mix of a song."),
quiz::Quiz("Pink Noise","A noise signal similar to white noise, containing all audib\
le 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 o\
ctave. As each higher octave has double the frequency of the octave below it which s\
preads out the energy over a wider range of frequencies, pink noise tends have a mor\
e 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) a\
nd therefore tends to sound very bright."),
quiz::Quiz("Pitch Bend","A mechanism on a synth, keyboard or controller that can cau\
se the pitch of the note to move up or down by a small amount."),
quiz::Quiz("Pitch to Voltage Converter","A device that detects the frequency of an a\
udio waveform and changes it into a control voltage, which is in turn fed to an osci\
llator that produces a pitch at the same frequency."),
quiz::Quiz("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-Convert\
er.”)"),
quiz::Quiz("Pitch-to-Voltage Converter","A device that detects the frequency of an a\
udio waveform and changes it into a control voltage, which is in turn fed to an osci\
llator that produces a pitch at the same frequency."),
quiz::Quiz("pitch","1) The perception of frequency by the ear (a higher or lower ton\
e 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."),
quiz::Quiz("Plate Reverb","A device that produces artificial reverberation by sendin\
g vibrations across a metal plate via a transducer similar to a speaker driver. Phys\
ical plate reverbs today are considered a vintage form of artificial reverb; nowaday\
s, most plate reverb effects are emulated digitally by plugins or reverb units."),
quiz::Quiz("Playback Head","A transducer that converts magnetic flux recorded on tap\
e into an audio signal for playback."),
quiz::Quiz("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."),
quiz::Quiz("Playback","1) The reproduction of recorded audio. 2) In motion picture o\
r video production, the reproduction of the music over loudspeakers so the performer\
s/musicians can perform in time to the music for the camera."),
quiz::Quiz("Playlist","1) See “Take.” 2) A user-defined selection of songs; a featur\
e available on most streaming and digital media players."),
quiz::Quiz("PLL","A phase locked loop is, in essence, an oscillator that tries to ma\
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tch 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 h\
igher than the incoming reference signal - such as a timing module that can create a\
n output clock that is 2, 4, 8, or more times the tempo of an incoming clock, or a v\
ery high frequency oscillator that is locked to a multiple of an incoming pitch - pe\
rhaps to drive a special circuit such as a switched-capacitor filter."),
quiz::Quiz("Plug","A connector, usually on a cable, that mates with a jack."),
quiz::Quiz("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 i\
n dB of sound arriving from different directions. Similar to “Pickup pattern,” but m\
ore specific. 2) In speakers, a graphic display of the speaker’s dispersion of sound\
."),
quiz::Quiz("Polarity","The direction of current flow or magnetizing force."),
quiz::Quiz("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 pol\
arizer, as it changes the polarity (+ versus -) of a signal. A control voltage inver\
ter is often combined with an offset voltage to adjust the output voltage into the d\
esired 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. Sinc\
e some modules such as voltage controlled amplifiers usually expect only positive vo\
ltages, you would then need to add 8 volts to that result to get an upside-down (inv\
erted) envelope that still had an overall range of 0 to +8v."),
quiz::Quiz("Polarizing Voltage","In condenser and electret microphones, the introduc\
tion of a small amount of electrical current to create the magnetism by which the ca\
pacitor converts audio signals to electrical current. In condenser microphones, pola\
rizing voltage is provided externally (see also “Phantom Power”); in electret microp\
hones, the polarizing voltage is permanently impressed on the condenser during manuf\
acturing."),
quiz::Quiz("Pole Pieces","Iron or other magnetic material that conducts magnetic for\
ce for use in transducers like record heads, playback heads, microphones, speakers, \
etc."),
quiz::Quiz("Pole","This is a technical term that helps describe the design of a filt\
er. Each pole of a filter attenuates frequencies beyond its cutoff or corner frequen\
cy 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 cuto\
ff frequency by 24 dB; frequencies two octaves above the cutoff are attenuated by 48\
 dB and so forth."),
quiz::Quiz("Polyphonic","The term \"polyphonic\" refers to a synthesizer that can pl\
ay 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 t\
he same time. A term commonly used to describe synths and keyboards. (See also “Voic\
e.”)"),
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quiz::Quiz("Ponging (Bouncing)","The technique of combining and mixing multiple trac\
ks 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, o\
r digitally through a digital audio workstation. Bouncing was once used frequently b\
y engineers to free up additional tracks for recording, but in digital workstations \
where tracks are virtually unlimited, this practice is basically obsolete. Today, en\
gineers typically bounce tracks for the purpose of creating a preliminary or final m\
ix of a song."),
quiz::Quiz("Pop Filter","A device that is placed over a microphone or between the mi\
crophone and vocalist to prevent loud “pop” sounds created by the vocalist’s breath \
directed toward the microphone."),
quiz::Quiz("Port","1) A connection point in computer or electronic device for transm\
itting and receiving digital data, similarly to how a jack receives and transmits au\
dio signals. 2) An opening or vent in a speaker case that resonates with air movemen\
t in the speaker, used in bass reflex speakers and woofers to enhance low frequencie\
s."),
quiz::Quiz("Portamento","A pitch change that smoothly glides from one pitch to anoth\
er. Also refers to the synthesizer mode or MIDI command that allows or causes this t\
o happen."),
quiz::Quiz("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 effec\
ts.  NOTE: We mention film in this context because film post-production includes a l\
ot of audio work (e.g., voiceovers, foley, audio mixing and editing) to the point th\
at many audio engineers are involved in film post-production as a full-time career."\
),
quiz::Quiz("Post Roll","A segment of blank tape (or track silence, on a DAW) that ru\
ns past the end of the recording. (See also “Pre-Roll.”)"),
quiz::Quiz("Post-Fader","Refers to an aux send position or setting that places the s\
end after the channel fader within the signal path. Sending a signal post-fader mean\
s the fader itself affects the level of the send signal, as opposed to pre-fader. (S\
ee also Pre-Fader.)"),
quiz::Quiz("Post","Refers to an aux send position or setting that places the send af\
ter 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 als\
o Pre-Fader.)"),
quiz::Quiz("Pot","Often thought of as a fancy word for “knob,” a potentiometer is ba\
sically 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 t\
o an aux send or bus. Potentiometers can be knobs or faders, meaning that almost eve\
ry control on a console that isn’t a button or switch is a potentiometer. However, m\
any engineers commonly refer to faders as “faders” and knobs as “pots.”"),
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quiz::Quiz("Potentiometer","(Abbreviated “Pot“) Often thought of as a fancy word for\
 “knob,” a potentiometer is basically any mechanism that controls input or output vo\
ltage 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 f\
aders, 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.”"),
quiz::Quiz("Power Amplifier","(abbreviated “Power Amp”) A device that amplifies a li\
ne level signal to drive a speaker or set of speakers.  (See also “Line Level.”)"),
quiz::Quiz("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 con\
nectors that go to your individual modules."),
quiz::Quiz("PPQN","When you send a clock signal (usually a gate signal or other elec\
trical 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 t\
erms of how many pulses there are per quarter note – PPQ or PPQN for short. If the c\
lock is just happening every quarter note, then the clock speed is 1 PPQN; in the ca\
se of DIN Sync (a popular standard among early Roland synths, with DIN being the typ\
e 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)."),
quiz::Quiz("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 (S\
ee also “Pre-Fader,” “Post-Fader.”)"),
quiz::Quiz("Pre Emphasis","A boosting of high frequencies during the recording proce\
ss to keep the audible signal above the noise floor."),
quiz::Quiz("Pre Fader","Refers to an aux send position or setting that places the se\
nd 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."),
quiz::Quiz("Pre-Delay","A parameter on a reverb unit or plugin that determines the a\
mount of time (delay) between the original dry sound and the early reflections of re\
verberation. This feature is often used to simulate the natural acoustic properties \
of a room, but can also be used to create interesting unnatural effects."),
quiz::Quiz("Pre-Echo","(Also called “Forward Echo”) A compression artifact that ofte\
n occurs in digital audio in which an “echo” of a sound (or part of a sound) is hear\
d ahead of the sound itself, often due to the data inconsistencies in certain compre\
ssed 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 g\
enerally an unwanted problem that requires additional signal processing to resolve—b\
ut in some cases it can also be used on purpose as a sound effect (not to be confuse\
d with “Reverse Echo”)."),
quiz::Quiz("Pre-Fade Listen (PFL)","A function on the channel strip of a mixer or DA\
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W that allows a channel signal to be heard and often metered before the channel fade\
r."),
quiz::Quiz("Preamplifier (Preamp)","A low-noise amplifier designed to take a low-lev\
el signal (for example, from a microphone) and bring it up to normal line level befo\
re sending it into the mixing console."),
quiz::Quiz("Precedence Effect (Haas Effect)","Simply stated, a factor in human heari\
ng in which we perceive the source of a sound by its timing rather than its sound le\
vel. 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 refl\
ection or reproduction from another source. The reflection of the sound must be at l\
east 10dB louder than the original source, or delayed by more than 30ms (where we ca\
n 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 con\
fused by reflections and reverberations off of nearby surfaces. Understanding the Ha\
as 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."),
quiz::Quiz("Precision Adder","Synthesizers are very sensitive to unintentional varia\
tions in pitch control voltage – any error can result in the oscillators under contr\
ol going out of tune. Therefore, whenever you add together pitch control voltages in\
side a modular synth, you really should be using a precision adder that precisely ad\
ds together the pitch voltages without introducing an error. Ordinary mixers might s\
lightly attenuate or amplify a voltage passed through them, which in most cases woul\
d create tuning errors."),
quiz::Quiz("Premix","1) The process of mixing a set of tracks as group, then managin\
g 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 n\
ot necessarily pre-recorded. (See also “Bouncing.”) 2) An important part of film pos\
t-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."),
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.")\
,
quiz::Quiz("Presence","1) In amplification and mixing, the boosting of upper-mid fre\
quencies 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.”\
"),
quiz::Quiz("Preset","A factory programmed set of parameters on a synth, signal proce\
ssor, plug-in or other electronic device."),
quiz::Quiz("Pressure Microphone","(Also called “pressure operative microphone”) – A \
microphone whose diaphragm responds to incoming sound wave pressure as it works agai\
nst the normal or controlled air pressure inside the microphone case. This design ma\
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kes the diaphragm sensitive to pressure regardless of direction, giving it an omnidi\
rectional pickup pattern. (See also “Omnidirectional Pattern.”)"),
quiz::Quiz("Pressure Sensitivity (Aftertouch)","A feature in some keyboard instrumen\
ts by which applying additional pressure to a key after it has been pressed can acti\
vate an additional MIDI control command. a synthesizer or Keyboard Controller of Aft\
er Touch (a control or operational function of a synthesizer where pressing a key af\
ter it has been pressed, and before it is released, will activate a control command \
that can be set by the player)."),
quiz::Quiz("Pressure Zone Microphone (Boundary Microphone)","An omnidirectional micr\
ophone 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 tradem\
ark Pressure Zone Microphone (PZM)."),
quiz::Quiz("Pressure-Gradient Microphone","(Also called “Velocity Microphone“) A mic\
rophone whose diaphragm is exposed front and back, with diaphragm movement being cau\
sed by the pressure difference between its front and back. This creates a bi-directi\
onal or “figure-8” pickup pattern (See also “Bi-Directional Pattern.”)"),
quiz::Quiz("Pressure","Some keyboards measure how hard you press down on the keys, a\
nd convert this to a voltage (or other control signal such as MIDI, which can then b\
e converted into a control voltage) that you can use to add expression to a note, su\
ch 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 r\
ather than pressure or force - so you can increase or decrease the aftertouch amount\
 by rolling between the tip and length of your finger."),
quiz::Quiz("Print Through","The unwanted transfer of magnetic flux from one layer of\
 analog tape to another."),
quiz::Quiz("Pro Tools","Avid’s trade name for its digital audio workstation (DAW) th\
at has become an industry standard in professional recording studios."),
quiz::Quiz("Producer","In music, the producer is the director of an audio recording \
project; the person responsible for getting a final product of desired quality withi\
n a budget."),
quiz::Quiz("Production Studio","Broadly speaking, any space dedicated to production \
within the arts, for example, film/video, animation or post production. In the conte\
xt 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."),
quiz::Quiz("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."),
quiz::Quiz("Program Change","A MIDI message that tells the receiving device to chang\
e presets."),
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quiz::Quiz("Programmable","Able to have the parameters changed by the user, especial\
ly in a computer controlled device."),
quiz::Quiz("Prompt","A set of instructions for the user to follow, which appears on \
a computer screen."),
quiz::Quiz("Protocol","In digital and information technology, a set of rules governi\
ng the structuring and transmitting of data in a standardized format so all related \
devices can properly interpret the data."),
quiz::Quiz("Proximity Effect","The natural boost in the microphone's output for bass\
 frequencies as the mic is placed closer to the sound source."),
quiz::Quiz("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 p\
sychological and emotional factors into account. This branch of science is helpful t\
o audio engineers in understanding how the brain interprets various sounds and frequ\
encies."),
quiz::Quiz("Puck","Any circular piece of metal, fiber, rubber, etc., which drives so\
mething from a rotating power source. A common example in the recording studio is th\
e puck in a rotating Leslie speaker."),
quiz::Quiz("Pulse Code Modulation (PCM)","A process by which analog signals are tran\
slated to digital code. This is done by taking samples of the amplitude of the analo\
g signal at regular rapid intervals, then translating it into binary numbers as a di\
gital 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 a\
udio."),
quiz::Quiz("Pulse Per Quarter Note","When you send a clock signal (usually a gate si\
gnal or other electrical pulse) around a modular synth to move sequencers through th\
eir steps and the such, it's good to know how fast that clock is pulsing. This is us\
ually defined in terms of how many pulses there are per quarter note - PPQ or PPQN f\
or 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)."\
),
quiz::Quiz("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 port\
ion narrower and vice versa). The act of varying the width of the resulting pulse wa\
ve 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 re\
ferred 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, mor\
e lush feel, or of giving a digital sound more analog properties."),
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quiz::Quiz("Pulse","Pulse has a couple of different meanings in a modular synth. Whe\
n you alter the shape of a square wave so that one portion is narrower than the othe\
r, 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 refer\
red to as a pulse. 1) The steady beat in music based on its tempo, whether audible o\
r perceived. 2) A type of sound wave commonly created and manipulated by synthesizer\
s whose waveform is characterized by sharp rises and drops in amplitude like a squar\
e wave, but whose peaks are shorter than its troughs, giving the wave a pulse-like f\
eel. Also called “Pulse Wave.”"),
quiz::Quiz("Pumping and Breathing","In studio jargon, an effect created when a compr\
essor is rapidly compressing and releasing the sound, creating audible changes in th\
e signal level. “Pumping” generally refers to the audible increase of sound levels a\
fter 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, alth\
ough some engineers and musicians use it on purpose occasionally to create a particu\
lar effect."),
quiz::Quiz("Punch In / Punch Out Recording","The process of activating and/or deacti\
vating the record function on tape or DAW during playback of a passage, usually as t\
he performer plays/sings along. This can be used either as a method of doing quick o\
verdubs, or as a way of getting a better take on a certain passage without having to\
 start the track from the beginning."),
quiz::Quiz("Pure Tone","A tone consisting of only the fundamental frequency with no \
overtones or harmonics, graphically represented as a simple sine wave."),
quiz::Quiz("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 (to\
uch 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 con\
vert that pitch to a corresponding voltage that can drive a VCO in unison with the o\
riginal sound."),
quiz::Quiz("PWM","Most oscillators that output a square waveform also have an additi\
onal control voltage input that sets the width of the top portion of the “square” wa\
ve (obviously, making the top portion wider makes the bottom portion narrower and vi\
ce versa). The act of varying the width of the resulting pulse wave creates a sort o\
f Doppler shift; varying the width back and forth - for example, by modulating the p\
ulse width with a low frequency oscillator - creates a chorusing effect that can sou\
nd like a detuned pair of oscillators. The resulting effect is referred to as pulse \
width modulation."),
quiz::Quiz("PZM","Abbreviation for Crown Audio’s Pressure Zone Microphone. (See also\
 “Boundary Microphone.”)"),
quiz::Quiz("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, th\
e broader the bandwidth curve of frequencies that will be boosted or cut. If you com\
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e from the pro audio world, you may be used to Q referring to the width or narrownes\
s of a peak or notch filter. In a synthesizer filter, when you increase the resonanc\
e (feedback), a peak forms around the cutoff frequency of the filter’s curve or shap\
e. The higher the resonance, the higher and narrower this peak. As a result, some us\
ed to use the audio term Q to refer to the resonance amount, although you don’t hear\
 that term used nearly as much today."),
quiz::Quiz("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."),
quiz::Quiz("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."),
quiz::Quiz("Quantization Distortion","Quantization Distortion/Quantization Error – T\
he effective “error in translation” between an analog signal and its sampled counter\
part due to the rounding of a large number of analog values to the nearest digital q\
uantity. This often results in additional random frequencies in the sound, often hea\
rd as noise."),
quiz::Quiz("Quantization Noise","The modulation noise in a signal resulting from qua\
ntization error. "),
quiz::Quiz("Quantization","1) In digital music, the process of adjusting the rhythmi\
c performance of music by moving the notes to precise locations on the time line, ef\
fectively “rounding” the note occurrences to the nearest defined increment. 2) In an\
alog-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)."),
quiz::Quiz("Quantizer","A quantizer auto-corrects the input voltage to the nearest d\
esired 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, bu\
t quite often they are standalone modules."),
quiz::Quiz("Rack Ears","Rack Ears/Rack Flanges – Mounting brackets that can are atta\
ched to equipment so it can be mounted in a standard equipment rack."),
quiz::Quiz("Rack Mounted","Describing outboard gear that can be housed in an equipme\
nt rack."),
quiz::Quiz("Rack Rash","When you mount a module into a case, the head of the screw o\
r bolt used to mount the module can scratch the faceplate of the module. These scrat\
ches are referred to as rack rash. You can almost never see it when you mount a modu\
le, as the scratches are behind the screw or bolt head, but nonetheless some will pa\
y more for a used module that is unscratched. So buy a bag of plastic washers and pu\
t them behind the screw or bolt head just to remove another reason for someone to no\
t buy your used module."),
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quiz::Quiz("Rack Unit","Rack-mounted equipment usually follows a standard set of dim\
ensions, including 19” (48.3 cm)for width, and a “rack unit” (or U for short) for he\
ight equaling 1.75” (4.4 cm) per U. Many common modular synthesizer formats follow t\
he rack unit system for standardizing module height – such as 3U (3 x 1.75 = 5.25” o\
r 13.3 cm) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (\
sometimes referred to as MU for Moog Unit)."),
quiz::Quiz("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 d\
escribes the area where sound arrives at the microphone, while a radiation pattern d\
escribes how sound is dispersed from the loudspeaker."),
quiz::Quiz("Radiation","The angle and pattern of coverage of a speaker."),
quiz::Quiz("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 rai\
ses 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."),
quiz::Quiz("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 h\
ard-drive storage memory where projects are saved and recalled). In the recording st\
udios, 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 MID\
I instruments)."),
quiz::Quiz("Random Note Generator","A device that generates random pitches at a set \
rate, used in synthesizers."),
quiz::Quiz("Random","Most voltages moving around inside a modular synth are very pur\
poseful in their variations: the repeating waveforms of an audio rate or low frequen\
cy oscillator; the rising then falling voltages of an envelope generator. However, i\
t can also be useful to have randomly wandering voltages to create everything from s\
ubtle variations in pitch to wildly varying volumes or filterings. Noise is an examp\
le of an audio-rate random signal."),
quiz::Quiz("Rap","To perform a spoken rhythmic part to a music or percussion perform\
ance."),
quiz::Quiz("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.”)"),
quiz::Quiz("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 bur\
st of four sixteenth notes appearing instead for one or more stages."),
quiz::Quiz("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."),
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quiz::Quiz("Rated Load Impedance","The input impedance, or opposition to current flo\
w by an input of a device, that a piece of equipment is designed to feed."),
quiz::Quiz("RCA Plug","(Also called Phono Plug) A common audio connector found on mo\
st stereo systems with a center pin as one connection and an outer shell as the seco\
nd connection."),
quiz::Quiz("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)."),
quiz::Quiz("Read","To retrieve information bits from a storage device; in digital au\
dio, the reproduction of digital signals."),
quiz::Quiz("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 virt\
ual musical instruments which can be played live or used with other sequencing softw\
are."),
quiz::Quiz("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 electrol\
ytic capacitor to be 25 years. Therefore, serious vintage synth owners “recap” (repl\
ace the age-sensitive capacitors in) their older equipment."),
quiz::Quiz("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."),
quiz::Quiz("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."),
quiz::Quiz("Record Monitor","On some tape machines, a switch position that allows th\
e VU meter and sound output of the tape machine electronics to monitor the input sig\
nal to the tape machine."),
quiz::Quiz("Record Ready","A control state of a multitrack tape recorder where the d\
esignated track will begin recording when the record function of the tape recorder i\
s activated."),
quiz::Quiz("Recording Bus","A bus that sends a mix signals from the console channels\
 to the multitrack recorder or DAW. (See also “Bus.”)"),
quiz::Quiz("Recording Session","A bloc of time in which music is being recorded in t\
he studio."),
quiz::Quiz("Rectifier","A circuit that makes sure a voltage stays only positive or n\
egative. In power supplies, it is used to remove the negative component of AC voltag\
e, or to protect you from plugging in module's power connector backwards. As a modul\
e, a half-wave rectifier passes only positive voltages and replaces anything negativ\
e with 0v; a full-wave rectifier takes any negative voltages and inverts them so the\
y become positive. This effectively doubles the frequency of many simple waveforms, \
like the triangle and sine."),
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quiz::Quiz("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 increas\
ing frequency. It has a bass-heavy sound, akin to the sound of the surf at a distanc\
e. It can also be used a slowly changing random control voltage or modulation signal\
, instead of as an audio source."),
quiz::Quiz("Reel","1) The hub and flanges onto which analog tape is spooled; recordi\
ng and playback involves unspooling the tape from one reel and onto another. 2) Some\
times 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 inten\
ded 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 delivere\
d on reels."),
quiz::Quiz("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 Leve\
l.”"),
quiz::Quiz("Reference Tone","A single-frequency tone (often at 1000 kHz) used to cal\
ibrate the levels of sound equipment; the tone used to set reference level. (See als\
o “Test Tones.”)"),
quiz::Quiz("Reflected Sound","Sound that reaches a microphone or listener after one \
or more reflections from surrounding surfaces."),
quiz::Quiz("Reflection","In acoustics, the bouncing of sound waves off of a flat sur\
face, as opposed to absorption. Reflection can have a great impact on how we perceiv\
e the collective sound; reflected sounds from a distance is perceived as echo, while\
 reverberation is created from thousands of reflections. (See also “Absorption,” “Ea\
rly Reflection,” “Echo,” “Reverberation.”)"),
quiz::Quiz("Regeneration","Regeneration can have a couple of different meanings insi\
de a synth, both meaning feedback. An echo unit can feed some of its output back int\
o its input, causing the delayed signal to be repeated again; this is sometimes refe\
rred to as regeneration. Also, very rarely you will hear resonance in a filter refer\
red to as regeneration."),
quiz::Quiz("Regulated Power Supply","A device to supply power to electronic equipmen\
t whose output voltage will not fluctuate when more equipment is turned on, or if th\
ere is a change in voltage of the power line. A regulated power supply is designed t\
o protect sensitive electronics from destructive power surges."),
quiz::Quiz("Relay","An electromagnetically activated switch that connects or disconn\
ects two terminals when a control voltage is applied."),
quiz::Quiz("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 designa\
ted threshold."),
quiz::Quiz("Release","This refers to the final stage of an envelope that typically f\
alls back to zero volts, usually resulting in silence. It is often used in the conte\
xt of talking about an Attack/Release (AR) or Attack/Decay/Sustain/Release (ADSR) en\
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velope generator, but can refer to any final stage of an envelope."),
quiz::Quiz("Remote","1) A device that controls the functions of another device wirel\
essly. 2) Describing on-site recording, as opposed to recording in the studio."),
quiz::Quiz("Reset","The Reset input on a module accepts a trigger or gate signal, an\
d 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 se\
quencer, it means go back to the first stage. In the case of an envelope, it means g\
o back to the start of the attack. In the case of a gate delay, it means to re-start\
 the timer for the delay."),
quiz::Quiz("Residual Magnetization","The amount of magnetism left in a magnetic mate\
rial after the magnetizing force is removed. Residual magnetism can accumulate in ta\
pe machines over time, either creating distortions and noise in the sound output or \
partially erasing the tape."),
quiz::Quiz("Residual Noise","The noise level left on recording tape after it has bee\
n erased."),
quiz::Quiz("Resistance","The opposition of a substance to the flow of electrical cur\
rent, measured in ohms."),
quiz::Quiz("Resistor","An electrical component with a specific amount of resistance \
to electrical current, used within the circuit to regulate the flow of current."),
quiz::Quiz("Resonance","The natural tendency of physical substances to vibrate with \
more energy at certain frequencies. The principle of resonance is a key element in t\
he design of acoustic instruments; for example, the hollow chamber of a guitar or vi\
olin 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 re\
sult is an increased boost in the harmonics right around the filter’s cutoff or corn\
er 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. The\
refore, the term resonance is often used to refer to a filter’s feedback amount."),
quiz::Quiz("Resonant Frequency","A frequency at which a physical item vibrates natur\
ally."),
quiz::Quiz("Resonate","To vibrate at the resonant frequency. Also refers to the ling\
ering reverberation that causes a sound to continue after the sound source has stopp\
ed. This continuing sound is due to the sympathetic resonance of nearby objects."),
quiz::Quiz("Resonator","Many acoustic instruments include a body or sound chamber th\
at “resonates” - sympathetically vibrates at, or reinforces - one or more frequencie\
s. To simulate this effect in modular synths, you can get a specialized filter or eq\
ualization module that boosts the sound at typically three or so user-definable freq\
uencies, each usually within a narrow band. This is one of the secrets of synthesizi\
ng real-world sounds or spaces."),
quiz::Quiz("Reverb (Reverberation)","1) Short for “Reverberation.” (See “Reverberati\
on.”) 2) A signal processor or plug-in that creates artificial reverb to a signal.")\
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,
quiz::Quiz("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 reve\
rb time is measured between the point at which the sound source stops and the point \
at which the reverberation levels fall by 60 dB."),
quiz::Quiz("Reverb","Short for reverberation. This is an effect device that mimics b\
eing 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 dimensio\
n to what might otherwise be a stark sound. There are relatively few modules that im\
plement a reverb effect, and even fewer that allow you to voltage control some of it\
s parameters (the ErbeVerb being the most famous); many just use an external reverb \
effect."),
quiz::Quiz("Reverberant Field","Describes the space that is far enough from the soun\
d source that the reverberations are louder than the direct sound."),
quiz::Quiz("Reverberation Chamber","A device built to simulate room reflections."),
quiz::Quiz("Reverberation Envelope","The attack, decay, sustain and release of the r\
everberation volume; or how fast the reverberation reaches peak level and its rate o\
f decay."),
quiz::Quiz("Reverberation","The persistence of a sound after the source stops emitti\
ng it, caused by many discrete echoes arriving at the ear so closely spaced in time \
that the ear cannot separate them."),
quiz::Quiz("RF Interference","The unwanted noise introduced into electronics, circui\
ts and/or audio systems by the presence of RF signals. RF interference in a system c\
an result in humming, buzzing, static or even the reproduction of radio transmission\
s."),
quiz::Quiz("RF Signals","RF Signals (or RF) - Short for Radio Frequency Signals, ele\
ctromagnetic waves that carry wireless radio and television signals. The vast majori\
ty of RF signals exist at frequencies higher than 100 kHz."),
quiz::Quiz("Rhythm Section","The musical instruments in a band or ensemble that are \
responsible for playing rhythmic parts rather than melody parts. In contemporary mus\
ic, rhythm sections typically consist of drums and bass, along with some combination\
 of percussion, piano/keyboard and/or guitars."),
quiz::Quiz("Ribbon Controller","This is a long strip that is capable of measuring th\
e position where you press it along its length, and the pressure used to press it. I\
t can be used as an alternate keyboard or as a pitch bend controller, with the posit\
ion determining pitch. Shorter versions also appeared sometimes as alternate control\
lers on synthesizers, such as the Yamaha CS-80."),
quiz::Quiz("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-di\
rectional or figure-8 pattern."),
quiz::Quiz("Riff","A short melody repeatedly played in a tune often with variation b\
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etween vocal lines."),
quiz::Quiz("Ring Modulator","Balanced or ring modulation is a special type of amplit\
ude modulation, where one bipolar (swinging both above and below 0 volts) signal – t\
he modulator – is used to vary the amplitude of a second bipolar signal, known as th\
e carrier. The modulator's frequency is both added to and subtracted from the carrie\
r's frequency; the resulting harmonics replace the original carrier and modulator.")\
,
quiz::Quiz("Ringing Out a Room","The process of identifying and compensating for pro\
blem 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 f\
requencies."),
quiz::Quiz("Rise Time","The rate at which an audio waveform makes a sudden increase \
to a higher amplitude."),
quiz::Quiz("RMS Meter","A meter that recognizes and responds to the effective averag\
e, the RMS level, or the effective average value of an AC waveform, rather than to t\
he peak level. (See also “Root-Mean-Square,” “Peak Meter.”)"),
quiz::Quiz("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."),
quiz::Quiz("Room Equalization","In live audio, an equalizer inserted in the monitor \
system that attempts to compensate for frequency response changes caused by room aco\
ustics."),
quiz::Quiz("Room Sound","The natural ambience of a room, including the reverberation\
 and background noise."),
quiz::Quiz("Room Tone","The natural background noise occurring in a room without mus\
ic playing or people speaking. In recording audio for film and TV, on-set sound mixe\
rs capture a take of room tone for the purpose of providing continuity between clips\
 of dialogue during post-production."),
quiz::Quiz("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 th\
e peaks. (See also “RMS Metering,” “Peak Metering.”)"),
quiz::Quiz("Rotating Head","A circular head with two (or more) gaps that rotates aga\
inst the direction of tape motion at a slight angle to the tape travel."),
quiz::Quiz("Rumble","A low-frequency noise, typically caused by earth/floor vibratio\
n or by uneven surfaces in the drive mechanism of a tape recorder or playback unit."\
),
quiz::Quiz("Rythm Tracks","The recording of the rhythm instruments in a music produc\
tion."),
quiz::Quiz("S-trig","Some systems – such as the original Moog modular – use an s-tri\
gger (switch or shorting trigger) instead of a normal gate, which was a wire that wa\
s shorted to 0 volts ground, like the closing of a switch wired to ground. You canno\
t interconnect these two systems without some form of conversion between the two, wh\
ich can be as simple as a special cable."),
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quiz::Quiz("S/H","A sample and hold (S/H) module has two inputs: a signal that is be\
ing sampled, and a trigger input that indicates when the first input should be sampl\
ed. When a trigger is received, the current voltage at the first input is sampled (m\
easured) 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 cr\
eating 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, whic\
h is then dutifully sent to the output."),
quiz::Quiz("S/PDIF","Abbreviation for “Sony/Phillips Digital Interface,” a protocol \
for sending and receiving digital audio signals using a common RCA connector."),
quiz::Quiz("Safety Take (ST)","An additional take of audio captured for good measure\
 after a take of acceptable quality has been recorded."),
quiz::Quiz("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.\
"),
quiz::Quiz("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 shoul\
d 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 volta\
ge is held until a new trigger is received. Sample and holds are most often associat\
ed 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 v\
alue, which is then dutifully sent to the output."),
quiz::Quiz("Sample Dump Standard (SDS)","See “MIDI Sample Dump Standard.”"),
quiz::Quiz("Sample Rate Conversion","The conversion of digital audio taken at one sa\
mple rate to a different sample rate without first converting the signal to analog."\
),
quiz::Quiz("Sample Rate","This is a specification of digital audio: How fast the ind\
ividual measurements (samples) that reconstruct a sound are recorded or played back.\
 The bandwidth of that audio file (which corresponds to the highest frequency that c\
an 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 sa\
mple 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 resolu\
tion of a digitally reproduced sound are described as a combination of sample rate a\
nd bitrate. (See also “Bitrate.”)"),
quiz::Quiz("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 pur\
pose of reproducing and manipulating the sound digitally."),
quiz::Quiz("Sampler","A device that records and plays samples, often with features f\
or editing, manipulating and storing the samples."),
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quiz::Quiz("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 inst\
ead held at that limit. However, saturation usually implies a more rounded, shaped a\
pproach to that clipping limit, resulting in a more pleasing (or at least less annoy\
ing) 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 proc\
essing. 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 exces\
s of sound level. This creates some distortion that some audiophiles describe as “an\
alog 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 co\
lor and warmth in the sound that engineers often find appealing. 3) A digital plugin\
 that emulates tape or tube saturation."),
quiz::Quiz("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 “R\
amp Wave.”)"),
quiz::Quiz("Sawtooth","One of the most common waveforms produced in a synthesizer. T\
his ramp-shaped wave contains both even and odd harmonics, strongest at the fundamen\
tal frequency (the note being played) and diminishing at the higher frequencies. The\
 result is very bright, loud, “brassy” sound."),
quiz::Quiz("Schmitt Trigger","This is a type of gate detector that looks at a varyin\
g input signal and outputs either a “high” (typically 0, 10, or even 15 volts) signa\
l or a “low” signal (typically 0 volts). When the input goes above one reference thr\
eshold – 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."),
quiz::Quiz("scope","This is a piece of test equipment that displays voltage fluctuat\
ions as graphical waveforms. A ’scope can run at a wide range of frequencies, displa\
ying slowly changing voltages like LFOs or envelopes, or quickly changing voltages l\
ike oscillators and noise. Oscilloscopes used to be bulky pieces of external equipme\
nt, but now you can get USB scopes that offload the display portion of the job to yo\
ur computer, or scopes as modules."),
quiz::Quiz("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 pa\
rts. At a later date the final vocal track is overdubbed. 3) The action of a musicia\
n or disc jockey quickly moving a record back and forth on a turntable reproducing t\
he stylus motion to create a rhythm pattern of sound."),
quiz::Quiz("Scrubbing","The action or function of shuttling a piece of recorded audi\
o back and forth while monitoring it, typically to locate a certain point in the rec\
ording. In earlier days, scrubbing was done with reel-to-reel analog tape by manuall\
y turning the reels to pull the tape across the playhead. Today, scrubbing is primar\
ily done digitally on a DAW by dragging the cursor back and forth across the wavefor\
m."),
quiz::Quiz("Second Engineer","An assistant recording engineer."),
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quiz::Quiz("SEM","The Oberheim SEM (Synthesizer Expander Module) was one of their ea\
rliest products. It was an entire synthesizer voice - two oscillators, two simple en\
velopes, VCA, and a very popular two-pole state variable filter design with a knob t\
hat crossfaded between low pass, notch, and high pass outputs plus a separate bandpa\
ss 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 t\
he original Oberheim synth."),
quiz::Quiz("Semi-modular","The components of a semi-modular synth - such as the osci\
llator, filter and amplifier - are pre-wired behind the front panel in what the manu\
facturer considers to be a typical, logical way. However, they also provide patch po\
ints 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 w\
ho are new to modular synthesis dip their toe in the water by getting a semi-modular\
 synth, and then expanding it with additional modules."),
quiz::Quiz("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 d\
ivision is 1/12 of an octave."),
quiz::Quiz("Send Level","A control determining the signal level sent to a send bus."\
),
quiz::Quiz("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 o\
f another microphone from the same sound level. 2) In music, describes the artistic \
persona in general."),
quiz::Quiz("Sequence","1) A pre-programmed set of musical events, such as pitches, s\
ounding 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 ref\
erring to a segment of music in general."),
quiz::Quiz("Sequencer","The most common type of sequencer you’re going to see in a m\
odular synth contains a row of knobs (also known as steps or stages) that may each b\
e set to output a different voltage. A sequencer then goes through steps one at a ti\
me. This is most often used to create repetitive musical lines where each note has t\
he same duration, which is popular in trance-like forms of music as well as the clas\
sic Berlin School style (70s-era Tangerine Dream and Klaus Schulze; current Red Shif\
t and Node). A computerized device or software that can be programmed to play a step\
ped order of musical events, including playing of pitches, sounding of samples, and \
rests."),
quiz::Quiz("Sequential Switch","This module comes in a few different forms; in the m\
ost common, a few different inputs are routed to one output (although they are usual\
ly 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 t\
o 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 mig\
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ht directly route a series of inputs to corresponding outputs (with usually a summed\
 output as well)."),
quiz::Quiz("Serial Data","A digital data stream where individual bits are transmitte\
d one after another over a single connection (as opposed to “parallel data,” in whic\
h multiple bits can be sent at once). Most data connections in the recording studio \
transmit serial data—for example, USB, Firewire and MIDI."),
quiz::Quiz("Series Connection","Connecting devices (especially circuit elements) so \
that the electrical signal flows from one thing to the next, to the next, etc."),
quiz::Quiz("Set Up","The positioning of microphones, instruments, connections and mo\
nitoring in the studio, as well as the controls and levels on consoles, DAWs, etc., \
in preparation for recording."),
quiz::Quiz("Shelf Filter","A name for the circuit in an equalizer used to obtain the\
 shelf."),
quiz::Quiz("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 i\
t; a low-frequency shelf does the same for signals at and below the set frequency.")\
,
quiz::Quiz("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 electromagn\
etic interference."),
quiz::Quiz("Shielded Cable","Cable that has a shield around an inner conductor or in\
ner conductors."),
quiz::Quiz("Shock Mount","An elastic mount on microphone stand that reduces the impa\
ct of unwanted vibrations that may affect the stand (for example, floor vibrations f\
rom footsteps)."),
quiz::Quiz("Short Circuit","A direct connection between two points in a circuit that\
 (usually) should not be connected."),
quiz::Quiz("Short Delay","Delay times under 20 milliseconds."),
quiz::Quiz("Shortest Path","A technique in recording that routes the signal through \
the least amount of active (amplified) devices during recording."),
quiz::Quiz("Shotgun Microphone","A microphone with a long line filter, a tube that a\
coustically cancels sound arriving from the side, to make the microphone pick up muc\
h 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."),
quiz::Quiz("Sibliance","Energy from a voice centered around 7 kHz, caused by pronoun\
cing “s”, “sh” or “ch” sounds."),
quiz::Quiz("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 sidechaini\
ng is in compressors, particularly in ducking effects where the presence of a partic\
ular audio signal triggers the compression of another audio signal. (See also “Ducki\
ng.”)"),
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quiz::Quiz("Signal Flow","1) In the general sense, the path that an audio signal tra\
vels from the sound source to the system output. (For example, from the vocalist’s v\
oice into the microphone, through the cables, into the preamp, out of the preamp int\
o 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 sig\
nal through the console, from input to output."),
quiz::Quiz("Signal Processing","The practice of altering the character or sound of a\
n audio signal through a variety of devices or plug-ins, such as equalizers, compres\
sors, reverb units, etc."),
quiz::Quiz("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."),
quiz::Quiz("Signal","1) In audio, an alternating current (or voltage) matching the w\
aveform of, or being originally obtained from, a sound pressure wave. 2) Also in aud\
io, an alternating current (or voltage) between 20 Hz and 20,000 Hz. 3) A digital au\
dio bit stream."),
quiz::Quiz("Sine Wave","1) In the general sense, the path that an audio signal trave\
ls from the sound source to the system output. (For example, from the vocalist’s voi\
ce 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 signa\
l through the console, from input to output."),
quiz::Quiz("Sine","This is the purest waveform: It contains only the fundamental har\
monic, and no higher harmonics. As a result, it’s a great wave to use to create a su\
b 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 modul\
ation (AM), or wavefolding which add or shift harmonic content."),
quiz::Quiz("Slap Echo (also called Slapback)","A single, distinct echo of a sound, w\
hich can result naturally from higher frequencies reflecting off a non-absorbent wal\
l, or artificially reproduced by a signal processing unit or plugin. Slap echo creat\
es a “live” sounding effect similar to what you would hear in an arena."),
quiz::Quiz("Slate","Slate (Slating) – 1) In video/film, the identification of a scen\
e and take at the beginning of the clip for the purpose of video editing. This is do\
ne by presenting the scene/take in written form in front of the camera on a clapboar\
d, 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 tr\
ack. 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 v\
isually on the screen."),
quiz::Quiz("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 instr\
ument that is being operated remotely by MIDI information sent from another device."\
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),
quiz::Quiz("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 re\
sult, it is sometimes called a lag generator or processor, or more technically as an\
 integrator."),
quiz::Quiz("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 nu\
mber of nuts are inserted into these channels, which can then be slid to any positio\
n to accommodate the mounting hole spacing of your modules. In a Eurorack case, thes\
e nuts tend to have a 2.5mm or 3mm hole and corresponding thread."),
quiz::Quiz("Slope Generator","A slope generator creates ramps: rising or falling vol\
tages. It is essentially a gate generator and a slew limiter (see above) wired toget\
her 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 gener\
alized control voltage functions - even creating a sawtooth or triangle wave oscilla\
tor - some companies such as Buchla and Serge referred to by its elemental function \
of generating sloping voltage changes."),
quiz::Quiz("Slope","Most filters typically have a cutoff or corner frequency they ar\
e tuned to. It then reduces (filters) the frequency spectrum of a signal going throu\
gh it so that it 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 h\
ave slopes that are defined multiples of 6 decibels (dB) weaker for each octave furt\
her away you get from the cutoff frequency. For example, a low-pass filter (LPF) wit\
h a slope of 24 dB/octave would attenuate harmonics one octave above its cutoff freq\
uency by 24 decibels."),
quiz::Quiz("Smart FSK (Frequency-Shift Key)","Smart FSK - An updated form of Frequen\
cy-Shift Key (FSK) sync that enables MIDI devices to sync to analog tape recorders a\
nd/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 be\
at for MIDI sequencers/devices at any point in the recording. This enables the devic\
e to start playing at exactly the right place and tempo no matter where you start th\
e tape. (See also “Frequency-Shift Key.”)"),
quiz::Quiz("SMPTE Time Code","(Abbreviated “SMPTE“) A standardized timing and sync s\
ignal 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 purp\
oses in audio recording environments. Many audio professionals simply refer to this \
time code as “SMPTE.”"),
quiz::Quiz("SMPTE","1) Abbreviation for Society of Motion Picture and Television Eng\
ineers. 2) See “SMPTE Time Code.”"),
quiz::Quiz("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."),
quiz::Quiz("Sock Cymbal","A rarely used alternate term for “hi-hat,” left over from \
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the days when hi-hat cymbals were placed at “sock level.” (See also “Hi-Hat.”)"),
quiz::Quiz("Soft Knee","In compression, refers to the gradual introduction of compre\
ssion of the signal once the sound level crosses the threshold. (See also “Knee.”)")\
,
quiz::Quiz("Software Instrument (Virtual Instrument)","One of a number of software-b\
ased synthesizers, samplers or sound samples that are stored and accessed via comput\
er and performed by an external MIDI controller, rather than in a standalone synthes\
izer or module. Because of the wide versatility available from these instruments, a \
growing number of composers and electronic musicians are working with virtual instru\
ments that can be stored in hard drives, rather than purchasing stacks of keyboards \
and modules."),
quiz::Quiz("Soldering","The action of making connections with solder, a soft metal a\
lloy that is used to bond two metal surfaces by melting. In audio settings, solderin\
g 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 b\
uying new ones or sending them off for repair."),
quiz::Quiz("Solid State","In electronics, refers to the use of transistors and semic\
onductors (solid materials) in the building of electronic devices, as opposed to tub\
es. In the recording studio, solid state amplifiers have different properties than t\
ube amps, and each has its own advantages and disadvantages. A more recent applicati\
on of solid state construction is in computer devices, particularly solid state hard\
 drives (SSD), which transfer data more quickly than conventional spinning disc driv\
es, and are less prone to breakage."),
quiz::Quiz("Solo Switch","A switch that activates the solo function on a console or \
DAW."),
quiz::Quiz("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 featur\
ed above other instruments."),
quiz::Quiz("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.\
"),
quiz::Quiz("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."),
quiz::Quiz("Sound Effects (SFX)","Sounds other than dialogue, narration or music tha\
t are added to audio, usually in the context of film/video."),
quiz::Quiz("Sound File","A digital audio recording that can be stored in a computer \
or on a digital storage medium (such as a hard disk)."),
quiz::Quiz("Sound Modeling","A technique that recreates a sound without directly mod\
eling the physical device. An example is additive synthesis, which uses a combinatio\
n of sine waves and noise to recreate sounds."),
quiz::Quiz("Sound Module","An electronic instrument (tone generator, synth or sample\
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r playback unit) that has no playable interface, but instead responds to incoming MI\
DI 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 configurat\
ion. Today, sound modules can also occur as software versions or plugins to be acces\
sed on a computer."),
quiz::Quiz("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 pe\
rceive SPL in terms of amplitude, volume or loudness of the sound."),
quiz::Quiz("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."),
quiz::Quiz("Sound Source","The origin of a sound, whose vibrations create sound wave\
s."),
quiz::Quiz("Sound Wave","(Also called “Sound Pressure Wave”) A wave caused by a vibr\
ation that results in slight variations in air pressure, which we hear as sound."),
quiz::Quiz("Soundtrack","1) Broadly speaking, refers to any/all audio that accompani\
es an instance of visual media, whether music, dialogue or SFX. 2) In more common te\
rms, refers to the musical score and/or licensed music synced to a film, video, TV p\
rogram or video game."),
quiz::Quiz("Source of Uncertainty","This was the name for the Buchla 265 and 266 mod\
ules that create random control voltages. Its name is often used for random source m\
odules that follow or are inspired by the original Buchla template."),
quiz::Quiz("Spaced Pair","(Also called “A/B Technique“) A stereo microphone placemen\
t technique in which two cardioid or omnidirectional microphones are spaced somewher\
e between 3-10 feet apart from each other (depending on the size of the sound source\
) to create a left/right stereo image."),
quiz::Quiz("Speaker","A device that converts electrical signals to sound; more techn\
ically, a transducer that changes an electrical audio signal into sound pressure wav\
es."),
quiz::Quiz("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, liquid\
s and gases, and though we usually think of sound as traveling through the air, diff\
erences in temperature, air pressure and humidity can also affect how fast sound tra\
vels.  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 ai\
r at sea level at a temperature of 15 degrees Celsius (59 degrees Fahrenheit). By co\
ntrast, at 70 degrees Fahrenheit under standard atmospheric conditions, the speed of\
 sound is about 344 m/s, or 770 mph."),
quiz::Quiz("Splicing","Historically, the act of attaching previously cut pieces of a\
udio tape or film in precise locations by applying a special kind of adhesive tape o\
n the back. This is/was done for the purpose of shortening sections of audio or edit\
ing film. Today, splicing has become a very simple process by editing sections of au\
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dio or video digitally with a DAW or film editing software."),
quiz::Quiz("Splitter","The short definition is something that can divide a signal in\
to two or more copies, such as a splitter cable where two outputs are wired to one i\
nput. For a deeper discussion, see the entry on multiple, as there are ways of going\
 about this beyond simple wiring."),
quiz::Quiz("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 o\
ften pleasing chorusing sound. Depending on the module, you might even be able to sp\
read these tones to form intervals, triads, and chords."),
quiz::Quiz("Spring Reverb","A device that simulates reverberation by creating vibrat\
ions within a metal spring by attaching it to a transducer and sending the audio sig\
nal through it. A pickup at the other end converts those vibrations into an electric\
al signal which is mixed with the original audio signal. While the physical spring r\
everbs still exist, most studios emulate spring reverb with the use of plug-ins or h\
ardware reverb units."),
quiz::Quiz("Square wave","This is a common waveform produced by a synthesizer’s osci\
llator. 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 positiv\
e voltage). Aside from being a really easy waveshape to generate with analog circuit\
ry, it has an interesting harmonic series: it has a strong fundamental, then gradual\
ly weaker odd harmonics: a component at three times the fundamental frequency, one a\
t fives time 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 harmo\
nics 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."),
quiz::Quiz("Stackable Cable","Many banana style cables are constructed that each plu\
g 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."),
quiz::Quiz("Stage Monitor","A speaker on the stage that enables performers to hear t\
hemselves and to hear what the other musicians are playing on stage."),
quiz::Quiz("Stage","1) The partially enclosed or raised area where live musicians pe\
rform. 2) In reverberation effects devices, an echo added before the reverberation t\
o 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 sequenc\
er, 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 attac\
k, where the envelope generally rises from 0 volts to the highest voltage it can out\
put – is a stage. You might also hear it used to describe the number of sample stage\
s in a BBD (Bucket Brigade Delay), described elsewhere."),
quiz::Quiz("Standard Operating Level","A reference voltage level or maximum average \
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level that should not be exceeded in normal operation."),
quiz::Quiz("Standing Wave","An unwanted sound wave pattern that often occurs when th\
e sound wave bounces between two reflective parallel surfaces in a room, and the ref\
lected 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 r\
oom. This creates the audible illusion that the wave is standing still, so the frequ\
ency is amplified to an unwanted level in certain parts of the room while nearly abs\
ent in others. Standing waves are most common in square or rectangular rooms with pa\
rallel surfaces, so acoustic designers try to prevent these waves by installing abso\
rptive materials or introducing other items to offset the parallel surfaces."),
quiz::Quiz("Step Mode","A setting in a sequencer or DAW in which notes are input man\
ually, one note or step at a time."),
quiz::Quiz("Step Sequencer","This usually refers to a type of sequencer where you st\
ep to and pause on a stage, enter the note (and possibly the duration) for that stag\
e, move on to the next step, and so forth."),
quiz::Quiz("Step","Step is often used interchangeably with stage (see above), especi\
ally when talking about sequencers."),
quiz::Quiz("Stereo Image","The audible perception of stereo, in which different soun\
ds sources appear to be coming from far left, far right or any place in between."),
quiz::Quiz("Stereo Micing","Placement of two or more mics so that their outputs comb\
ine to create a stereo image."),
quiz::Quiz("Stereo","A recording or reproduction of at least two channels where posi\
tioning of instrument sounds left to right can be perceived."),
quiz::Quiz("Strike","This term appears on several Make Noise modules, although it ha\
s 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 l\
ight source such as an LED, allowing a voltage to be turned into a resistance to con\
trol a parameter) may have a strike input. When you flash an LED at a light sensitiv\
e resistor, it does not change the resistance instantaneously and stay there - inste\
ad, there is some delay as it glides to the desired resistance. When you turn the LE\
D off, the resistance may not go instantaneously to full; instead it might take a br\
ief 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 equipmen\
t and clean up after a recording session."),
quiz::Quiz("Subcode","Additional information bits that are recorded alongside digita\
l audio, used for control and playback purposes."),
quiz::Quiz("Subframe","A unit smaller than one frame in SMPTE time code."),
quiz::Quiz("Subgroup","A number of input channels on a console that can be controlle\
d and adjusted as a single set before sending the combined signal to the master outp\
ut. Sometimes also called “Submix,” “Bus” or just “Group.”"),
quiz::Quiz("Subharmonic","A circuit that divides the fundamental harmonic of the inc\
oming sound to produce lower frequencies, and therefore subharmonics. The most commo\
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n is an octave divider or sub bass circuit that divides creates a subharmonic by div\
iding the fundamental by 2 (some can also create a subharmonic two octaves below the\
 fundamental by dividing it by 4)."),
quiz::Quiz("Submaster / Sub-Master","The fader which controls the combined level of \
sound from several channels during mixdown or recording."),
quiz::Quiz("Submix","See “Subgroup.”"),
quiz::Quiz("Suboctave","A module that creates a new tone one or two octaves below th\
e 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 sin\
e-like, or that more closely resembles the original waveform."),
quiz::Quiz("Subtractive Synthesis","The most common synthesis technique: You start w\
ith one or more oscillators outputting waveforms with a large number of harmonics, a\
nd 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 ad\
ds articulation to the note by varying its loudness. An old-school method of sound s\
ynthesis in which sounds are designed and created by generating harmonically rich wa\
veforms, then filtering out unwanted harmonics to arrive at the desired sound."),
quiz::Quiz("Sum","To sum is a fancy way of saying you added two (or more) things tog\
ether; the sum is the result. It usually is used in the context of adding together c\
ontrol 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."),
quiz::Quiz("Summing","The process of blending two or more signals into one mixed sig\
nal. 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."),
quiz::Quiz("Super-Cardioid Pattern","A very tight cardioid microphone pattern with m\
aximum sensitivity on axis and the least amount of sensitivity approximately 150 deg\
rees off-axis."),
quiz::Quiz("Surround Sound","A technique of recording and playback in which the list\
ener 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. Surroun\
d sound can occur in various formats with different numbers of speakers arrayed thro\
ugh the room. Surround sound today is most commonly used in film and TV production."\
),
quiz::Quiz("Sustain","This is a common stage of an envelope generator where a voltag\
e – usually being sent to a filter’s cutoff frequency or an amplifier’s level – is b\
eing 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 co\
ntrol for how long the sustain stage should last. Envelopes that contain sustain sta\
ges include the ADSR (Attack/Decay/Sustain/Release) and AR (Attack/Release, which us\
ually assumes a sustain stage)."),
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quiz::Quiz("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 a\
vailable simultaneously. Another side effect is that they are not prone to oscillati\
ng at high feedback (resonance) settings, although some have certainly figured out h\
ow to make this happen. The Oberheim SEM (Synthesizer Expander Module) filter is per\
haps the most famous state variable design."),
quiz::Quiz("Sweetening","A vague term referring to the fine-tuning of audio in the p\
ost-production stage of recording. Effectively, any small “tweaks” to to make the au\
dio sound better is considered sweetening."),
quiz::Quiz("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. Y\
ou cannot interconnect these two systems without some form of conversion between the\
 two, which can be as simple as a special cable."),
quiz::Quiz("Switch","A device that makes and/or breaks electrical connections."),
quiz::Quiz("Switchable Pattern Microphone","A microphone having the capability of tw\
o or more pickup patterns, which can be toggled by use of a switch on the microphone\
."),
quiz::Quiz("Switching Power Supply","A switching power supply starts by directly con\
verting 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 switc\
hed voltage is then smoothed out to create a constant DC supply at the desired volta\
ge. 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 (t\
his is why they are often surrounded by a shielding cage). Many are moving to a hybr\
id power supply that combines a switcher with a small linear supply or regulator to \
get the best of both worlds."),
quiz::Quiz("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 tr\
acks stay in sync with each other through all stages of post-production."),
quiz::Quiz("Sync24","Sync24 is an alternate name used for the Roland-created standar\
d 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."),
quiz::Quiz("Sync","Sync can have two different meanings, depending on whether we're \
talking about oscillators or about clock signals. Some oscillators support a mode wh\
ere they reset their waveshapes to the beginning when they receive a signal from ano\
ther oscillator. If there is not a precise octave relationship between the two oscil\
lators, 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” sou\
nds 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 patter\
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ns, it is common to send a master timing or sync signal around the modular for all t\
he 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, synchro\
nizing 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.”"\
),
quiz::Quiz("Synthesizer Expander Module","The Oberheim SEM (Synthesizer Expander Mod\
ule) was one of their earliest products. It was an entire synthesizer voice - two os\
cillators, two simple envelopes, VCA, and a very popular two-pole state variable fil\
ter design with a knob that crossfaded between low pass, notch, and high pass output\
s plus a separate bandpass setting - in a cube-like case. Most often today, when a m\
odular manufacturer uses the magic letters \"SEM\", they're referring to a filter me\
ant to emulate that in the original Oberheim synth."),
quiz::Quiz("Synthesizer","A musical instrument that uses electrical oscillators to g\
enerate tones artificially, either to simulate the sounds of other instruments or to\
 create other sounds not possible with other instruments."),
quiz::Quiz("System Exclusive","System Exclusive (SysEx) - A MIDI message that will o\
nly be recognized by a unit of a particular manufacturer."),
quiz::Quiz("Tach/Tachometer","In analog tape recording, a device on the recorder tha\
t measures and regulates tape speed by emitting pulses as the tape moves across the \
head."),
quiz::Quiz("Tails Out","A method of winding audio tape so that the end of the last r\
ecorded selection is at the outside of the reel."),
quiz::Quiz("Take Notation","Writing down the takes of the tune being recorded on a t\
ake sheet or on the track log with comments. Take notation was/is recommended for an\
alog tape recording, but in most studios, this function is now accomplished on the D\
AW."),
quiz::Quiz("Take","The recording that is done between one start and stop of a tape r\
ecorder or DAW."),
quiz::Quiz("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 ha\
ve been used as an effect for guitars, but they can be used to modify other instrume\
nts, as well."),
quiz::Quiz("Talkback","A microphone in the control room carried on a separate circui\
t from the recorded channels, allowing the engineer to communicate with the musician\
s in the live room or sound booths through the monitoring system."),
quiz::Quiz("Tape Delay","A signal processing technique for creating artificial delay\
 or echoes by manipulating time delays with analog tape machines. This technique beg\
an 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 coul\
d adjust the length of the delay by adjusting the distance between the record and pl\
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ayback heads. Today, most tape delay effects in the studio are simulated digitally t\
hrough plug-ins in a DAW."),
quiz::Quiz("Tape Guide","Any stationary or rotating device which directs the tape pa\
st the heads on a tape machine, or from one reel to the other."),
quiz::Quiz("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 mos\
t of the noise floor that occurs in analog recording, and can be reduced by using ta\
pe constructed of finer magnetic particles. (See also “Noise Floor.”)"),
quiz::Quiz("Tape Loop","A length of tape with the ends spliced together so that the \
recording will play continuously."),
quiz::Quiz("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 h\
igh-frequency signals recorded above the tape hiss."),
quiz::Quiz("Telephone Filter","A filter used to simulate the audio heard through a t\
elephone receiver by removing signals at frequencies below 300 Hz and above 3500 Hz.\
"),
quiz::Quiz("Tempo Mapping","The act of programming a sequencer or DAW to follow the \
tempo variations of a recorded performance. Unlike beat mapping or beatmatching, bot\
h of which effectively adjust the recording to fit a set tempo, tempo mapping adjust\
s the tempo of the project (especially the MIDI instruments) to match the natural te\
mpo nuances of the recorded material. (See also “Beat Mapping,” “Beatmatching.”)"),
quiz::Quiz("Tempo","The rate at which the music moves, measured in Beats Per Minute \
(BPM)."),
quiz::Quiz("Terminal","1) A point of connection between two wires, including the plu\
g on the end of a cable, and the jack on a piece of equipment. 2) Refers to the keyb\
oard and monitor of a computer that enable the user to enter information and to acce\
ss data."),
quiz::Quiz("Test Oscillator","A device that generates audio waveforms at various fre\
quencies for testing purposes."),
quiz::Quiz("Test Pressing","One of a few initial vinyl record copies pressed from th\
e first stamper made, which is listened to and visually inspected to approve the qua\
lity before more copies are pressed."),
quiz::Quiz("Test Tones","1) A recording of several single-frequency tones at the beg\
inning of a tape reel at the magnetic reference level that will be used to record th\
e program. 2) Artificially generated tones that are used to calibrate an audio syste\
m."),
quiz::Quiz("Thin Sound","A vague term describing an audio signal that that is lackin\
g in certain frequencies, especially on the low end. Over-filtering a signal with an\
 EQ can produce a thin sound, for example."),
quiz::Quiz("Threaded Inserts","A common system for mounting modules into a case is c\
alled sliding rails or nuts. A number of nuts are inserted into these channels, whic\
h 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 \
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inserted into the channel that have been pre-drilled for the standard Eurorack mount\
ing hole spacing. They may be drilled for 2.5 or 3 mm screws; pay attention when buy\
ing the rails or a case that has them pre-installed."),
quiz::Quiz("Three-To-One Rule","A principle of microphone placement that says when m\
ultiple mics are used at once, the distance between microphones should be at least t\
hree 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."),
quiz::Quiz("Three-Way Speaker","A speaker system that has separate speakers to repro\
duce the bass, mid-range and treble frequencies."),
quiz::Quiz("Threshold of Hearing","Described as the sound pressure level at which pe\
ople can hear only 50 percent of the time."),
quiz::Quiz("Threshold","A threshold is generally a voltage level a signal needs to c\
ross before a module takes an action. For example, when the output of an envelope fo\
llower (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 indic\
ating a note has started. When the output of the envelope follower falls before a th\
reshold (which may be the same or different than the note-on threshold), then the ga\
te 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."),
quiz::Quiz("Throat","In a speaker, the small opening in a horn or in a driver throug\
h which the sound wave passes from the driver to the horn."),
quiz::Quiz("Through-Zero Frequency Modulation","TZFM is the abbreviation for Through\
-Zero Frequency Modulation. Think of a patch where you feed the output of one oscill\
ator (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 i\
s added to the normal pitch control voltage for the carrier, and the pitch of the ca\
rrier goes up. As the waveform output of the modulator goes below zero, it is subtra\
cted from the normal pitch control voltage, and the pitch goes down. But what happen\
s 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 mo\
dulation, the carrier will start playing backwards – in essence, a negative frequenc\
y. This generally produces a more pleasing result, and is a desirable characteristic\
 for an oscillator."),
quiz::Quiz("Throw","1) In speakers and in microphones, describes the amount of unres\
tricted 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 de\
signed for smaller spaces has a “short throw,” while one designed for a farther proj\
ection 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 e\
xample, emits high frequencies in a limited angle of direction, so it has a “long th\
row,” while a subwoofer emits low frequencies in all directions and has a “short thr\
ow.” 3) Something a producer, engineer or musician might do with whatever is in his/\
her hand during a moment of intense frustration."),
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quiz::Quiz("Tie Lines","Tie Lines - Cables with connectors at both ends, which are u\
sually 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."),
quiz::Quiz("Timbre","This word is often used to describe the unique tonal characteri\
stic of a sound you are creating, separate from its pitch or loudness. Different sou\
nds, by definition, have different timbres. When you change a parameter of a sound t\
hat changes its tonal characteristic - such as changing the filter cutoff, pulse wid\
th, amount of wavefolding, etc. - you are changing its timbre. The timbre often chan\
ges during life of a note. The sound quality that makes one instrument sound differe\
nt from other instruments, even while playing the same pitch. The timbre of a trumpe\
t, for example, is what makes it sound like a trumpet and not like a flute. Timbre i\
s largely shaped through the presence, absence and complexity of harmonics when the \
instrument is played."),
quiz::Quiz("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 MID\
I Time Code (MTC) and SMPTE time code."),
quiz::Quiz("Time Compression / Expansion","(Also called “Time Stretching” or “Time S\
hifting“) The process of speeding up or slowing down an audio recording without chan\
ging the pitch of the sounds."),
quiz::Quiz("Time Constant","A complex mathematical ides that basically describes the\
 time delay between when an electrical voltage is applied to a circuit and when the \
circuit responds to it."),
quiz::Quiz("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 o\
r might even damage a 1/8” jack."),
quiz::Quiz("Toms","The small drums (as little as 10 inch diameter) that mount on rac\
ks above the kick drum and the large drums in a drum set."),
quiz::Quiz("Tone Generator","1) A device that puts out test tones at various frequen\
cies to align a tape machine or for other testing purposes. 2) The circuits in a syn\
thesizer that create the audio signals put out by the unit, usually to emulate the s\
ound of another instrument."),
quiz::Quiz("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 frequenc\
ies."),
quiz::Quiz("Tonguing","The technique of controlling the start of a note in a brass o\
r woodwind instrument with the tongue."),
quiz::Quiz("Total Harmonic Distortion (THD)","The measure of the difference between \
the level of harmonic frequencies at the output stage of an amplifier as compared wi\
th the input stage, a ratio expressed as a percentage. It’s a fine-tuning specificat\
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ion barely perceptible to many ears, but the lower the THD, the more accurately the \
amplifier/speaker is reproducing the sound."),
quiz::Quiz("Touch Sensitive","See “Velocity Sensitive.”"),
quiz::Quiz("Track & Hold","This is a variation of a Sample & Hold. Both have two inp\
uts – a gate signal, and a voltage reference signal – and a voltage output. When a S\
ample & Hold receives a gate high signal, it freezes and outputs the voltage referen\
ce 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, t\
he reference input it passed along to the voltage output (this is the “tracking” pha\
se); when the gate goes low, the input voltage at that instant is frozen and maintai\
ned at the voltage output until a new gate high signal is received."),
quiz::Quiz("Track Log / Track Assignment Sheet","Track Log/Track Assignment Sheet – \
A sheet of paper kept with a multitrack tape which tells which instrument was record\
ed on each track."),
quiz::Quiz("Track","1) One audio recording made on a portion of the width of a multi\
track tape, or created as a digital representation using a DAW. 2) One set of contro\
l commands in a sequencer or DAW that is used to control one instrument over one MID\
I channel. 3) See “Band Track.”"),
quiz::Quiz("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 oct\
ave system where a rise of 1.00 volts on the pitch input should produce exactly a do\
ubling (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 volt\
age-controlled filters have a “tracking” switch for a CV input where the pitch of th\
e filter’s corner frequency only rises at 1/3, 1/2, or 2/3 of the corresponding chan\
ge 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 freq\
uency 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."),
quiz::Quiz("Transducer","A device that converts energy from one medium to another. T\
ransducers are prevalent throughout the equipment in a recording studio."),
quiz::Quiz("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, o\
r the strike of a drum."),
quiz::Quiz("Transistor Ladder Filter","This term is often used to describe the desig\
n of the much-loved Moog low-pass filter, which is still held up by many as being th\
e gold standard in low pass filter sound. Moog actually received a patent for this d\
esign (it has since expired); many of their competitors either outright copied it or\
 did their best to emulate it."),
quiz::Quiz("Transport","1) The portion of a tape machine that moves the tape from th\
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e 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, emulatin\
g the functions of a tape machine transport."),
quiz::Quiz("Transpose","In the simplest terms, to transpose the pitch of a musical l\
ine is to shift it up or down by a fixed number of semitones or octaves. This is som\
etimes referred to as “chromatic” transposition. A more sophisticated variation is “\
scalar” transposition where each note is shifted by a set number of scale steps; thi\
s differs from chromatic transposition because some scales may have differing number\
s 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 t\
hat 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 instrumen\
t, 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)."),
quiz::Quiz("Trap","1) A filter designed to reject audio signals at certain frequenci\
es. 2) An object designed with acoustically absorptive material, placed into walls t\
o reduce low frequency reflections in the room (also called “bass trap”). 3) Another\
 word for a drum set (as in “trap set”)."),
quiz::Quiz("Tremolo","This is the effect of varying the amplitude (loudness) of a no\
te. A way to create this effect on a modular synth is to patch a low frequency oscil\
lator (LFO) to one of the control voltage inputs on an amplifier. Tremolo is differe\
nt 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 i\
n a violin tremolo) or by rapid shifts in amplitude."),
quiz::Quiz("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 out\
put 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 s\
moother sine wave."),
quiz::Quiz("Triangular Wave","A harmonically rich waveform that appears triangular i\
n shape when depicted graphically, due to a combination of the presence of odd harmo\
nics and rapid rolloff."),
quiz::Quiz("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 fal\
ling back to 0 volts. It is often used to start or “trigger” the playback of a percu\
ssion 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 su\
bdivision of a note) starts. A trigger usually has a fixed duration, compared to a g\
ate 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\
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.")，
quiz::Quiz("Trim / Trim Control","A device that reduces or increases the signal stre\
ngth in an amplifier, often over a restricted range. Often used interchangeably with\
 gain, but usually referring to fine-tuning signal strength, rather than merely ampl\
ifying it."),
quiz::Quiz("Truncation","1) The shortening of an audio signal, sample or song, typic\
ally 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."),
quiz::Quiz("Tune","The act of adjusting the pitch of a synthesizer's oscillator (the\
 main pitch-generating element) to match another oscillator, instrument, or referenc\
e is known as tuning it."),
quiz::Quiz("Tuning Fork","A metal fork with two prongs that vibrate with a fairly pu\
re tone of one frequency when the fork is struck."),
quiz::Quiz("Turntable","A device to support and rotate a phonograph record during pl\
ayback."),
quiz::Quiz("Tweeter","A speaker designed to reproduce only the higher frequencies of\
 the sound."),
quiz::Quiz("Two Quadrant Multiplier","A two-quadrant multiplier performs a simple ve\
rsion of amplitude modulation (AM), where that varies the amplitude or loudness of o\
ne signal known as the carrier (typically an audio signal, swinging both above and b\
elow 0 volts) with a second signal called the modulator. In the typical amplitude mo\
dulation (AM) scenario, a low frequency oscillator with a positive voltage (say, bet\
ween 0v and 5v, or maybe something smaller such as between 1v and 2v) is fed into th\
e 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 p\
atching 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 wave\
form. (The case where the modulator's negative as well as positive excursions are us\
ed is referred to as a four quadrant multiplier.) "),
quiz::Quiz("Two-Way Speaker","A speaker system with separate speakers to reproduce t\
he lower frequencies (woofer) and the higher frequencies (tweeter)."),
quiz::Quiz("TZFM","TZFM is the abbreviation for Through-Zero Frequency Modulation. T\
hink 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 wavefo\
rm output of the modulator rises above zero volts, it is added to the normal pitch c\
ontrol voltage for the carrier, and the pitch of the carrier goes up. As the wavefor\
m output of the modulator goes below zero, it is subtracted from the normal pitch co\
ntrol voltage, and the pitch goes down. But what happens if the result of subtractin\
g 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 st\
art 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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quiz::Quiz("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 equ\
aling 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 c\
m) for Eurorack, 4U for Buchla and Serge, and 5U for classic modular Moogs (sometime\
s referred to as MU for Moog Unit)."),
quiz::Quiz("Unbalanced Audio","Most audio signals are passed around on cables with t\
wo wires: one for the voltage that represents the audio vibrations, and one for grou\
nd. This arrangement is often referred to as unbalanced audio."),
quiz::Quiz("Unbalanced Cable","A cable with two conductors (a signal wire and a grou\
nd wire) and connectors on each end. Unbalanced cables are often susceptible to elec\
tromagnetic interference and noise. Examples of unbalanced cables are guitar/instrum\
ent cables (also called tip-sleeve or TS cables) and RCA cables."),
quiz::Quiz("Unidirectional Pattern","A microphone pick-up pattern which is more sens\
itive to sound arriving from one direction than from any other."),
quiz::Quiz("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 t\
he output of an envelope generator – only vary between 0 volts and some maximum posi\
tive voltage; this is referred to as unipolar."),
quiz::Quiz("Unison","Several performers, instruments or sound sources that are sound\
ing at the same time and with the same pitch."),
quiz::Quiz("Unity Gain","The scenario in which there is no increase or decrease in s\
ignal strength at the output of an amplifier or device compared to the signal streng\
th at the input (typically described as 0 dB)."),
quiz::Quiz("Unity","Usually used in the phrase “unity gain” this mean a signal keeps\
 the exact same level from input to output."),
quiz::Quiz("Vacuum Tube","A diode, a glass tube with the gases removed, through whic\
h electrical current can flow. In audio, vacuum tubes are used in amplifiers, oscill\
ators, and other analog devices."),
quiz::Quiz("Vamp and Fade","A method of ending the recording of a song where the mus\
ic has a repeating part and the engineer reduces volume until the music fades out.")\
,
quiz::Quiz("Vamp","A part of a song or chord progression that is repeated, usually a\
t the end of the song, and usually the chorus or part of the chorus."),
quiz::Quiz("Vari-Speed","A control on a tape machine that changes the play speed."),
quiz::Quiz("Variable-D","A trademarked, patented technology of ElectroVoice in its m\
icrophone 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 sensi\
tivity to higher frequencies the further they are placed from the microphone’s diaph\
ragm."),
quiz::Quiz("VCA Automation","A system of mix automation in some mixing consoles in w\
hich sound levels or other functions are altered through the use of voltage controll\
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ed amplifiers."),
quiz::Quiz("VCA Group","Several VCA faders that are fed control voltages from a grou\
p master slide. A feature in higher-end mixing boards that enables the engineer to c\
ontrol groupings of independent signals by a single fader that uses VCA to adjust th\
e voltage sent to each channel."),
quiz::Quiz("Velocity Message","In synthesizers and keyboard controllers, a MIDI mess\
age that transmits data on how hard the key was struck. Velocity messages can be use\
d to transmit volume information, as well as triggering different samples on a multi\
-sampled instrument patch."),
quiz::Quiz("Velocity Microphone","See “Pressure-Gradient Microphone.”"),
quiz::Quiz("Velocity Sensitive","(Also called “Touch Sensitive“) A feature on a MIDI\
 instrument such as a keyboard that transmits a MIDI velocity message depending on h\
ow hard the key is struck."),
quiz::Quiz("Vibrato","A smooth and repeated changing of the pitch up and down from t\
he regular musical pitch, often done by singers or performed by string and wind play\
ers."),
quiz::Quiz("Virtual Instrument","(Also called Software Instrument) One of a number o\
f software-based synthesizers, samplers or sound samples that are stored and accesse\
d via computer and performed by an external MIDI controller, rather than in a standa\
lone synthesizer or module. Because of the wide versatility available from these ins\
truments, a growing number of composers and electronic musicians are working with vi\
rtual instruments that can be stored in hard drives, rather than purchasing stacks o\
f keyboards and modules."),
quiz::Quiz("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 ins\
truments into the vocal microphone, and also reduces natural ambience and reverberat\
ion in the vocal recording."),
quiz::Quiz("Vocoder","An audio processing device effects device or plug-in that anal\
yzes the characteristics of an audio signal and uses them to affect another synthesi\
zed signal. Primarily developed for the purpose of producing synthesized voice effec\
ts from human speech, a vocoder creates the characteristic robotic vocal sound or th\
e “human synthesizer” effect that makes it sound like the synth is speaking or singi\
ng words."),
quiz::Quiz("Voice Over","The recording of vocal announcements or narration over a be\
d of music in video, film or commercials."),
quiz::Quiz("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 th\
at 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 s\
pecific sound patch available on the synth."),
quiz::Quiz("Volatile Memory","Computer memory whose data will will be lost when the \
computer is turned off. RAM (Random Access Memory) is the most common form of volati\
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le memory."),
quiz::Quiz("Voltage Controlled Amplifier (VCA)","An amplifier whose gain level is af\
fected by an external voltage being sent to it. VCAs are commonly used in synthesize\
rs, signal processors, and as a means of automation for some mixing consoles."),
quiz::Quiz("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."),
quiz::Quiz("Voltage Controlled Oscillator (VCO)","An oscillator whose frequencies ar\
e modified by voltage input. Most commonly found in synthesizers."),
quiz::Quiz("Voltage","The difference in electrical force or pressure (“potential”) b\
etween two objects, causing a flow of electric current between them."),
quiz::Quiz("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 Mete\
r. (See also “VU Meter.”)"),
quiz::Quiz("Volume","A common, non-technical term that either refers to sound pressu\
re level (which we hear as loudness), or to audio voltage level."),
quiz::Quiz("Vox","A Latin word meaning “voice,” often used as an abbreviation for tr\
ack logs in the studio."),
quiz::Quiz("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.\
"),
quiz::Quiz("Watt","Unit of electrical power."),
quiz::Quiz("Wave","This is the pattern of vibrations - up and down fluctuations in v\
oltage - output by an oscillator. Different patterns generate different sounds."),
quiz::Quiz("Wavefolder","A wavefolder is a very specific design of waveshaper that u\
ses a comparator and some other circuitry. What they do is look to see if the wave g\
oes 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 porti\
on of the wave back upon itself, creating more high harmonics and interesting spectr\
a in the process."),
quiz::Quiz("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 wa\
ve, showing the wave’s oscillations above and below the zero line."),
quiz::Quiz("Wavelength","The physical length of one cycle of a wave, measured in fee\
t, inches, etc. The longer the wavelength of a sound wave, the lower its frequency; \
the shorter the wavelength, the higher the frequency."),
quiz::Quiz("Waveshaper","It would be a bit obvious to say “a circuit that changes th\
e 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 ou\
tgoing sine wave, or to add tube-like soft clipping to the peaks and transients of w\
aves. Many waveshapers are simply intended to mangle (er, add higher harmonics to) w\
aveforms in interesting ways, creating noisier (er, more complex and bright) harmoni\
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c spectra to create new sounds."),
quiz::Quiz("Wavetable","This term can have two related but slightly different meanin\
gs. 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 digit\
al 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 crossf\
ading between them (what most digital wavetable oscillators today do). Some people r\
efer to each table as a “wave” and a set of individual waves as a wavetable."),
quiz::Quiz("Weighting","An equalization curve used in audio tests that compensates f\
or the Fletcher Munson Curve at various levels. (See also “Fletcher-Munson Curves.”)\
"),
quiz::Quiz("West Coast Synthesis","The so-called \"West Coast\" approach to synthesi\
s - traditionally associated with companies such as Buchla and Serge - is often base\
d 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 suc\
h as an AD or AR (often called slope generators) rather than four-stage ADSRs, as we\
ll 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 w\
hat's known as a Low Pass Gate (LPG). The West Coast approach also embraces non-trad\
itional controllers, such as touch plates and the such. Today it's common to mix bot\
h East Coast and West Coast approaches in the same system."),
quiz::Quiz("wet sound","Sometimes people will say a filter has a “wet” sound. This u\
sually refers to a fewer-than-4-pole filter sound - often low or bandpass - with res\
onance 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."),
quiz::Quiz("Wet","A sound with effects (such as reverb) mixed is referred to as \"we\
t\"; a sound with no effects is referred to as \"dry.\" Effects units or mixers ofte\
n have wet/dry mix amounts that set the ratio between the original, unprocessed soun\
d and the fully-effected sound. Refers to a signal that has the full amount of an ef\
fect (like reverb) applied to it, as opposed to “dry,” which refers to the un-effect\
ed sound. Many times, the preferred sound in mixing will be a blend of wet and dry s\
ignals. (See also “Dry.”)"),
quiz::Quiz("White Noise","Noise is a random signal that does not have a distinct pit\
ch, 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 differe\
nt 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 contain\
ing an equal spread of energy across all audible frequencies. Like pink noise, engin\
eers often send a white noise signal through audio equipment for tuning and calibrat\
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ion purposes, or in EQ-ing a live audio space. (See also “Pink Noise.”)"),
quiz::Quiz("Whole Step","A change in pitch equivalent to two half steps, or the diff\
erence in pitch between two piano keys."),
quiz::Quiz("Wild Sound","In film and video, audio that is recorded separately from t\
he visual that may be added to the audio track later, and does not need to be synchr\
onized with the picture."),
quiz::Quiz("Wind Controller","A device that is played like a wind instrument to cont\
rol a synthesizer, module or virtual instrument via MIDI signals, as opposed to a ke\
yboard controller."),
quiz::Quiz("Windscreen","A covering that fits over a microphone to reduce the excess\
ive noise resulting from wind blowing into the mic. Typically used for recording in \
outdoor locations."),
quiz::Quiz("Wireless Microphone","A microphone that transmits its signal over an FM \
frequency to a receiver offstage, rather than traveling over an audio cable."),
quiz::Quiz("Woofer","A speaker that is designed to reproduce bass frequencies only."\
),
quiz::Quiz("Write Mode","A mode of operation in an automated console where the engin\
eer is in control of channel gain and the computer is recording the gain changes ove\
r time."),
quiz::Quiz("XLR Cable","A balanced microphone cable utilizing XLR connectors. (See a\
lso “XLR Connector.”)"),
quiz::Quiz("XLR Connector","A balanced cable connector consisting of 3 or 7 pins, mo\
st commonly used in microphone cables."),
quiz::Quiz("XY Miking","A coincident stereo microphone placement technique in which \
two cardioid microphones are placed with their heads toward each other at a 90-degre\
e angle, and as close together as possible. (See also “Coincident Miking.”)"),
quiz::Quiz("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 compo\
nents."),
quiz::Quiz("Zenith","In analog tape recording, refers to the tilt of the tape head i\
n the direction perpendicular to the tape travel."),
quiz::Quiz("Zero-Order Hold (ZOH)","Refers to the mathematical expression of the sig\
nal processing done by a conventional digital-to-analog converter (DAC)."),
};


int main()
{
        std::random_device rd;
    std::mt19937 gen(rd());
    std::uniform_int_distribution<> distria(1, 4);
    std::uniform_int_distribution<> distrib(0, game.size()-1);
        std::shuffle(std::begin(game), std::end(game), std::default_random_engine());
```

```
        std::vector<std::string> answers;
        std::string question;
        uint32_t n;
        uint8_t correct;
        uint32_t score=0;
        uint32_t tqs=0;

        for (uint32_t ctr=0;ctr<game.size();++ctr) {
                answers.clear();
                correct=distria(gen);
                for (uint8_t i=1;i<=4;++i) {
                        if (i == correct) {
                                answers.push_back(game[ctr].getA());
                                question=game[ctr].getQ();
                        } else {
                                answers.push_back(game[distrib(gen)].getA());
                        }
                }
                std::cout << "\33c\e[3J";
                if (tqs != 0) {
                        std::cout << "[QUESTIONS: " << tqs << " / " << game.size() << " SCORE: " <<
 << "]\n";
                }
                std::cout << "Question #" << tqs+1 << ": " << question << "\n\n";
                std::cout << "Answer #1.\n" << answers[0] << "\n\n";
                std::cout << "Answer #2.\n" << answers[1] << "\n\n";
                std::cout << "Answer #3.\n" << answers[2] << "\n\n";
                std::cout << "Answer #4.\n" << answers[3] << "\n\n";
                std::cout << "What answer is correct (q=quit)? ";
                std::cin >> n;
                if (n == 0) {
                        break;
                } else if (n == correct) {
                        score++;
                }
                tqs++;
                std::cout << n << " is the answer you gave. And the correct answer is: " << correc\
t << '\n';
        }

         std::cout << "\33c\e[3J";
         if (tqs != 0) {
                 std::cout << "[QUESTIONS: " << tqs << " / " << game.size() << " SCORE: " << score\
```

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

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 (÷6), eighth notes (÷12), eight note triplets (÷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 that 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 the 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 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 Verbos 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 it 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 the 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 the 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 it 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.

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) 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 frequncies 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.

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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 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 use 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.

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. Companding 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 "compliation" 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 is 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 is 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 to 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.

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, and 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 if 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, and 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 if 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.

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.

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.

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
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Hum
1) The low-frequency pitch that occurs when power line current is accidently 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 pick up 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.

IADSR
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.

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 in 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.

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.

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.

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 −10dBV (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 +/−0.447 volts is considered to be at −10dBV. By contrast, a typical oscillator signal in a modular synthesizer is +/−5 to +/−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.

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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 SMTPE 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 the 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 effecting 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 the 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 possiblity of one input being switched between multiple outputs. A sequential switch is a type of multiplexor, 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.

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.)

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 internal. 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.")

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 though 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 tunes 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 tunes 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 I 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.")

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.

Rack Ears
Rack Ears/Rack Flanges – Mounting brackets that can are 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.

Ringing 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.

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.

Sibliance
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 it 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 fives time 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 sounds 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.

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 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 ides 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.

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.

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 Mirophone
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 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.

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.

Woofer
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.

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.")

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.

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).

Appendix C: URL for Source Code

00-signalgenerators.cpp
https://gist.github.com/chbtoys/978de513b61ae51d5a3f6e8aecd67861

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