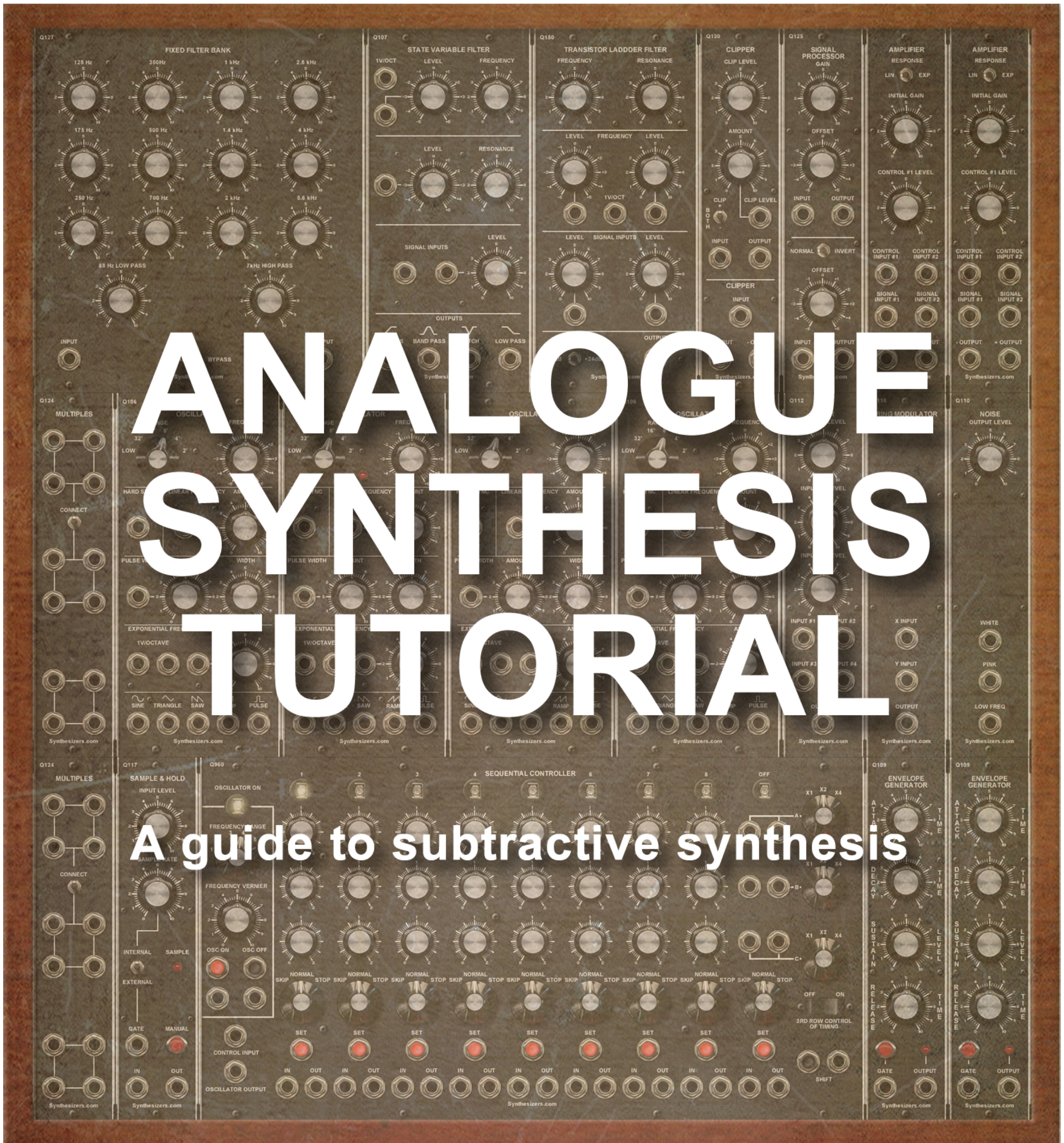
ANALOGUE SYNTHESIS TUTORIAL



A guide to subtractive synthesis

Stephen Howell

This tutorial…

The plan with this tutorial is to guide you through the basics of subtractive synthesis. That is, starting with a harmonically rich sound and filtering bits out to create the tone or sound you want.

This used to be the sole province of old analogue synthesisers but is now commonplace in a lot of modern synths and samplers. For example, the software sampler, Kontakt, can be a very versatile 'analogue' synth as well with its wide range of filters and other synth functions. Other software samplers too and even older hardware samplers such as the venerable Akai, Emu and Roland models are actually very capable 'analogue' synths. And now, of course, we are overloaded with no end of software synths that offer similar functionality.

The aim of this tutorial is to go back to basics to show how it all *used* to work with voltage control because with an understanding of those basic (and relative simple) techniques, they can be applied to almost anything whether you have picked up some crusty old analogue synth off eBay, some new modular synth or have the latest version of Kontakt or a new Access Virus, whatever.

It goes into some depth but (I hope) kept simple enough for it not to be overwhelming or intimidating.

But before you panic, why do it?

Because even in this day and age, nearly 50 years after analogue synths as we know them were developed (and after almost 150 years of the development of synths and similar instruments), there are still many who don't know an LFO from a UFO and I am contacted regularly for advice and tuition.

In 'the olde days', manufacturers such as ARP, EMS and Moog actually provided a concise explanation of the basic principles in their products' operator's manual but those days are such that all you'll get now is …

FILTER CUTOFF	Sets the filter cutoff
RESONANCE	Sets the resonance

Which, frankly, is not that useful!

So this tutorial hopes to redress that balance by providing the basics so that you KNOW what the filter cutoff and resonance (etc.) actually *does*. Some 'scientists' may contend some of the descriptions but I stand by them – they are intended to be simple and straightforward without an in-depth knowledge. You won't find any mathematical formulae here! This tutorial is for those who maybe *don't* know an LFO from a UFO and for those a little more clued up, should hopefully give a deeper insight into the wonderful world of analogue or substractive synthesis.

So dive in. I hope you find it useful and informative.

Stephen Howell

The basics....

Since the early days of synthesisers back in the 60s and before, a synthesiser can be broken down into just a handful of basic components. These are:

• SOUND GENERATORS

• SOUND PROCESSORS / MODIFIERS

• CONTROLLERS

The sound generators take the form of OSCILLATORS and also NOISE GENERATORS

The sound processors / modifiers take the form of FILTERS and AMPLIFIERS and also RING MODULATORS and, these days, effects units such as REVERB, DELAY, CHORUS, etc..

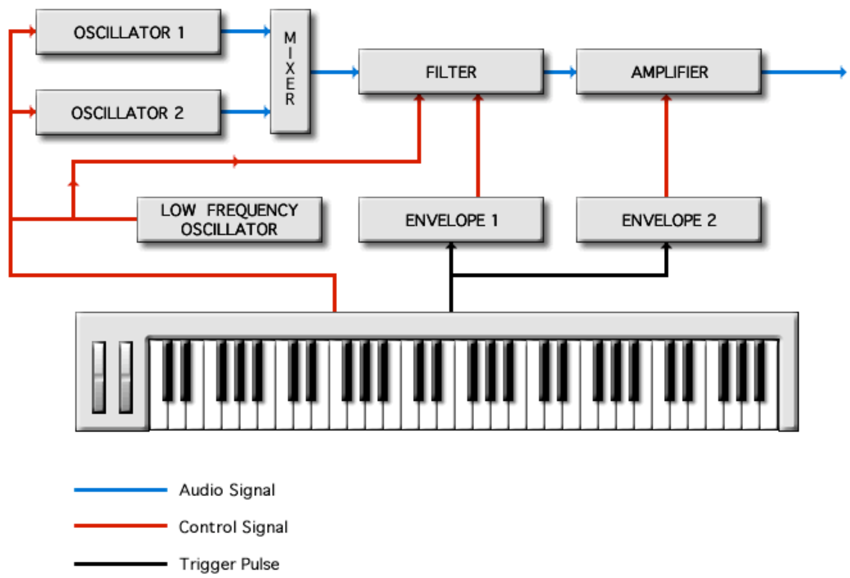
The controllers take the form of ENVELOPE GENERATORS, LFOs (LOW FREQUENCY OSCILLATORS) plus ‘real-time’ controllers such as KEYBOARD, PITCH BEND and MOD WHEELS, VELOCITY, AFTERTOUCH, etc..

Don’t worry about the jargon and terminology for now - we will look at these in detail throughout the course of this tutorial.

But first…

Voice architecture....

The typical signal flow of a typical analogue synth was pretty much defined with the MiniMoog and is something like the following:



Simplified block diagram of a typical analogue synthesiser

Two (or more) oscillators generate the basic sound and these are mixed and fed into a filter which allows you to manipulate the tone, often quite dramatically. This is then fed to an amplifier and out to the audio output(s) on the rear panel. The oscillators' pitch is typically controlled by the keyboard but can also be 'wobbled' by a low frequency oscillator (for vibrato, for example).

The filter is typically controlled by an envelope generator as is the amplifier and the envelopes are used to 'shape' the sound (i.e. determine whether it is percussive and/or 'plucky' or slow like strings... or just on/off like an organ). Combined, the different permutations of control settings on even a simple synth allow an astonishing range of sounds to be created.

We'll look at these different 'modules' in turn.

SOUND GENERATORS

The oscillators....

The oscillators define the basic pitch and tone of an analogue synth sound. You can think of them pretty much like the 'strings' of the instrument. They generate a basic waveform at a pitch set by a combination of tuning controls, the keyboard and different controllers.

Analogue synths typically offer five different waveforms. These are:

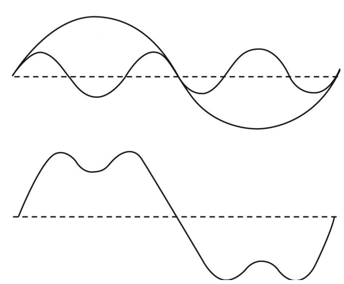
SAWTOOTH	A bright sounding waveform suitable for any number of applications such as strings, brass, pads, leadlines and more. Also knows as a 'ramp' waveform.
SQUARE	A bright, hollow sounding waveform (not unlike a clarinet).
PULSE	A thin and reedy sounding waveform. On most synths, the 'width' or 'symmetry' of the pulse wave can be varied for a greater range of sounds.
TRIANGLE	A mellow sounding waveform (flute-like).
SINE	A totally pure sound with no harmonics or overtones - the purest sound known.

The reason these waveforms all sound different is because they each contain different combinations of 'harmonics' or 'overtones' which we will look at next. But before that, a brief explanation of the nature of sound.

With the exception of the sine wave, all sounds, to a greater or lesser degree, have 'harmonics' or 'overtones' and it is these harmonics that define the tone or 'timbre' of a sound, the general rule of thumb being the more harmonics, the brighter the sound (and *vice versa*).

The predominant pitch we detect in a musical note is known as 'the fundamental frequency' and the harmonics are multiples of the fundamental's frequency.

All sounds are made up of multiple sine waves at different frequencies added to each other to create harmonically rich sounds. For example, if we take a sine wave and add just one sine wave harmonic at three times the frequency of the fundamental, a new waveshape is created…

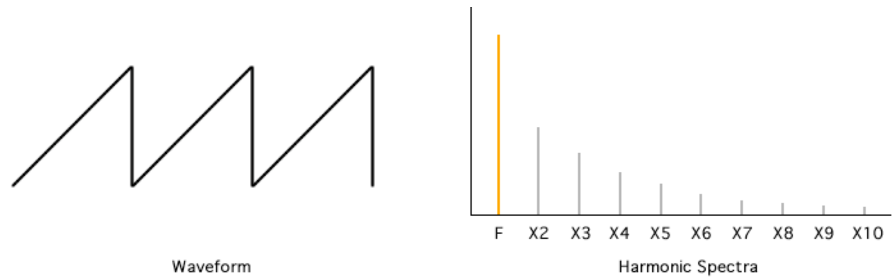


If more harmonics were added, a more complex waveform would be created.

In electronic waveforms, all the harmonics are in phase with the fundamental (as shown above) but in natural, acoustic waveforms, this isn't always the case which is why acoustic instruments can sound more 'alive' with more 'movement', more 'organic'. Synths can sound a bit 'static' by comparison … but that's often a quality you might want and there *are* ways to liven up synth sounds as we shall discover.

This is the foundation of subtractive synthesis – start off with a harmonically rich waveform, spice it up a bit and selectively filter out certain harmonics to create the tone you want. We shall look at each of the waveforms in detail next.

Sawtooth wave

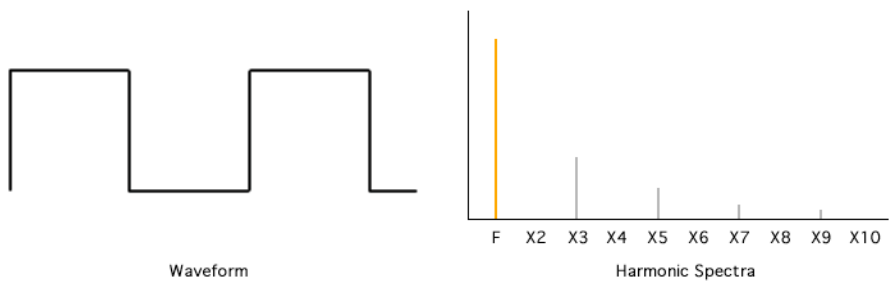


The sawtooth wave is very rich in both odd numbered and even numbered harmonics. That is, it has harmonics that are twice, three times, four times, five times (and so on) the fundamental frequency.

Thus if the fundamental (and first harmonic) frequency is 500Hz, the 2nd harmonic is 1kHz, the 3rd is 1.5kHz, the 4th is 2khz, etc.. This creates a very bright sounding basic waveform and is useful as the basis of many different sounds including strings, brass, leadlines, basses... in fact, almost anything!

The sawtooth shape of the waveform (which gives it its name) is how the signal would look if seen on an oscilloscope. If you looked at a trumpet's waveform on an oscilloscope, it would look similar.

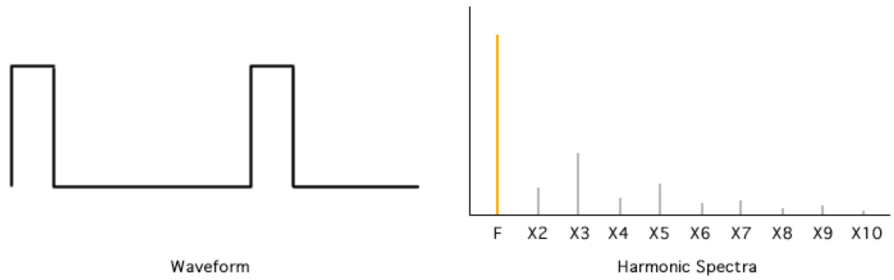
Square wave



The square wave is another very bright waveform but sounds different because it only contains odd numbered harmonics (X3, X5, X7, etc.).

As a result it sounds 'hollow' and not unlike a clarinet. It is useful for many sounds that require that quality and is very useful for reinforcing bass sounds, especially when tuned an octave down from the other oscillator(s).

Pulse wave



The pulse wave is a bit of an exception because the width of the pulse ‘spike’ can (typically) be continuously varied and the distribution of the harmonics changes according to the width of the pulse.

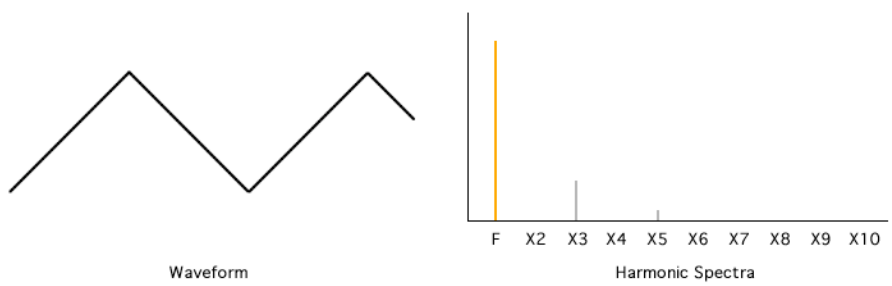
When the pulse is very thin, the sound is thin and ‘nasal’ (like an oboe) and gets fuller as the width increases. The pulse wave is also good for clavinet-like and other thin sounds.

When the pulse width is equal (i.e. 50/50), you have a square wave and you will often find that many synths don’t offer separate square and pulse waves (although some do) – instead the square wave selection serves to provide variable pulse waves as well.

One notable exception to this is the venerable Minimoog that provided a square wave and two fixed width pulse waves.

If you change the pulse width whilst it is sounding, you will hear a pleasing change in tone not unlike a chorus effect and if you use some controller to do that automatically, this is called ‘Pulse Width Modulation’ (or PWM) and can be useful for creating thick, ensemble textures as we shall see later.

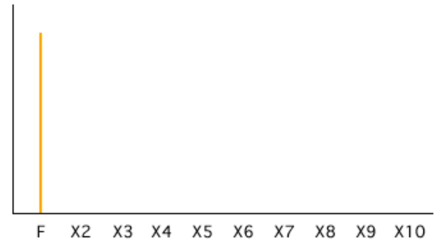
Triangle wave



The triangle wave is not unlike the square wave in that it only comprises odd numbered harmonics. However, the harmonics are very much lower in level resulting in a more mellow sound that is suitable for pure and simple sounds.

Sine wave

Waveform



Harmonic Spectra

The sine wave is the simplest waveform known and has only a fundamental with no harmonics at all. As such, it is a very ‘unnatural’ sounding waveform (there is no sound in nature or musical instrument that doesn’t contain *any* harmonics) and therefore is very good for creating pure sounds.

It is also very good for creating ‘sci-fi’ sounds because the early electronic music pioneers of the 50s and 60s only had very simple sine wave oscillators to play with.

The humble sine wave is also very useful for reinforcing the fundamental of other waveforms and comes into its own as a ‘sub-bass’ reinforcing the fundamental of a bass sound an octave down where it’s not so much heard as ‘felt’. This is not a new technique - church organists have been using it for centuries!!!

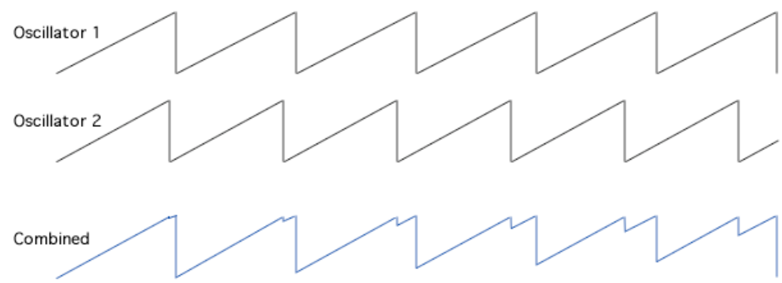
A Synthesizers.com oscillator

Oscillator pitch / tuning / frequency

As well as the basic choice of waveform, there is much you can do to govern the character of your sound by your choice of oscillator tuning.

A single oscillator on its own can sound a bit sterile, lifeless and ... well ... electronic (although this can be a quality you might want of a sound). As a result, it is common practice for most analogue synths to have two (or more) oscillators which can be detuned against each other to give a variety of chorus and ensemble sounds that are 'fatter' and 'warmer' than just a single oscillator in much the same way as an orchestral string section has a fuller sound than a single violin. The amount of detune can be ever so subtle to create a slowly changing sound or can be quite extreme to give a thick chorus effect.

The diagram below shows the effect of combining two sawtooth waveforms that are very slightly detuned against each other.



You can see the new combined waveshape constantly changing over time which creates a much more pleasing and 'animated' sound.

As well as small amounts of subtle detune, however, you can tune the oscillators apart by an octave or maybe two... or you can tune them, say, a fifth (seven semitones) or other intervals apart.

Of course, the more oscillators you have, the more scope there is for detune and tuning possibilities. The optimum number of oscillators appears to be three - fewer than that and your tuning options are limited; more than that can sometimes result in an audio 'mush'.

Although it is a good idea most of the time to employ two (or more) oscillators to create a fuller, more animated sound, sometimes a single oscillator is more appropriate. This can be especially true for creating solid bass sounds where the constantly changing phase relationship between oscillators can cause the bass sound to lose 'focus'. Alternatively, use Oscillator Sync (described on the next page) to lock the oscillators for a solid sound.

Just those possibilities - combining different waveforms at different tunings - allow you to create an enormous diversity of sounds.

But there's more......

Pulse Width Modulation (PWM)

We have already seen this mentioned in the description of the pulse wave.

The width, symmetry or 'shape' of the pulse wave can be varied from an equal square wave to a very thin pulse as shown below.

In some journals or articles, you might also see this referred to as the mark/space ratio and it represents the percentage of time the waveform is up and down. For example, a mark/space ratio of 10:90 means that the pulse wave is up for 10% of the cycle and down for 90% of it.

When this happens, there is a pleasing 'chorus' effect as the harmonic structure shifts and changes.

When setting the pulse width manually, it allows access to an almost unlimited assortment of different sounds but when put under the control of something like an LFO or an envelope generator or a real-time controller such as the mod wheel, the possibilities increase dramatically.

Controlled by a cyclic LFO, you can create lush, animated chorus and ensemble sounds.

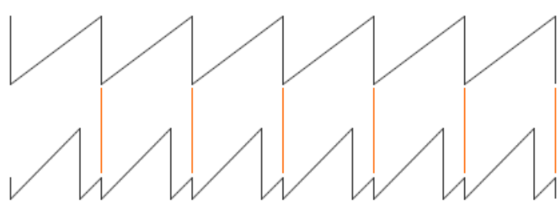
Controlled by an envelope generator, the pulse width can change over the course of a note.

Controlled by the modwheel, the pulse width can become a performance parameter.

Oscillator sync

Despite recommendations to detune oscillators to create a 'fatter' sound, it can sometimes be appropriate for the oscillators to be perfectly phase-locked without any detune or 'beating'. For example, you might want to set up a solid bass sound with the oscillators tuned an octave apart. Even if you fine tune them to *exactly* the same value, there will still be some 'phasing' between them - by sync'ing the oscillators, you can achieve the solid sound you want without the slight detune and potential lack of focus. This is achieved using the SYNC facility.

When this is switched on, the oscillators' waveform cycles are locked to each other so that they are perfectly in tune.



Normally, the oscillators are continuously drifting in and out of phase but when synced, they effectively create a new waveshape. However, this has some interesting side effects and benefits.

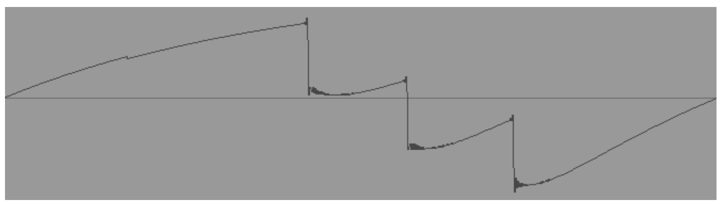
By tuning the oscillators apart in wide intervals with oscillator sync on, many interesting new waveshapes can be created that can sound distinctly 'un-analogue' and could almost have been created on a digital synthesiser. There are no rules for this - simply experiment using different waveforms and interval tuning combinations.

However, a further side effect and benefit to this is that if the frequency of the sync'd oscillator is changed during the course of a note, you get a distinctive 'tearing' sound not unlike a *very* strong flanger effect.

You can achieve this manually by controlling the pitch of the sync'd oscillator using - say - the modwheel and using the effect as a performance parameter (a popular technique with early synth players such as Jan Hammer or 80s synth pop-meister, Howard Jones, in 'What Is Love?') or you can 'automate' it using LFOs and/or envelopes to create many distinctive and classic 'sync sweep' sounds.

Super/Hyper Saw

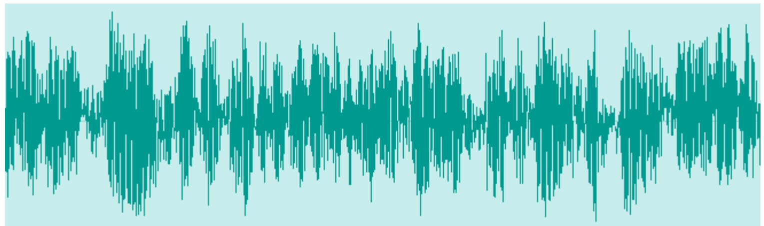
This is a fairly recent addition to the pantheon of synth waveforms.



Introduced by Roland on their JP8000 (Supersaw) and also to be found on the Access Virus (Hypersaw). It uses Digital Signal Processing (DSP) to create a thick waveform that is made up of effectively a number of detuned sawtooth waves – kind of like several oscillators in one waveform.

The waveform has made itself onto other virtual analogues and software synths and the parameters available tend to vary from one product to another but typically, most come with some sort of 'detune' controls to govern the thickness of the sound, etc..

Noise generators



So far, we have only looked at pitched waveforms. There are also sounds (such as drums and sound effects... wind, surf, etc.) that have unpitched elements. These are created on an analogue synthesiser using a noise generator.

Noise is made up of every frequency in the audio spectrum sounding at once. The most commonly known is white noise, so called because, like white light, it has an even distribution of frequencies across the spectrum. However, there are also other types of noise such as pink noise where the frequencies are balanced across the musical octaves. The technicalities are largely irrelevant - all you need to know is that white noise is bright and ‘hissy’ and suitable for wind and breath sounds whilst pink noise has more ‘rumble’ and is useful for thunder and surf sound effects.

There is also (very rarely) a low frequency (or ‘red’) noise option which is even more biased towards the low frequencies and can be seriously menacing and ‘rumbly’.

Synthesizers.com noise generator

Balancing / mixing the sound generators

You don't always want to have the oscillators at full level all the time - you will want to mix and balance their relative levels. For example, you might have a sound where one oscillator is an octave or two up but you might only want a hint of that in the sound or you may want to emphasise a low octave in a bass sound... or you may have tuned the oscillators a fifth apart but don't want the fifth element to be too prominent. Or you may have mixed in a bit of white noise that needs balancing against the pitched element of other oscillators... whatever.

All synths offer some way of balancing the relative levels of the various oscillators. Some two oscillator synths have a simple 'balance' control (which can be inflexible) whilst other synths have an oscillator mixer. Some, however, simply have an output level for each of the oscillators.

As with most things about an analogue synth, there are no rules - just adjust the relative levels according to taste and the requirements of the sound.

More modern VA synths also offer oscillator pan whereby the oscillators can be spread across the stereo image for a 'wider' sound. This can be particularly useful when creating certain large, ensemble sounds such as strings and pads to create a wide stereo sound (but can be a bit overpowering for bass sounds which typically fare better placed mono and central in the stereo image to create a solid foundation for the track). Again, no rules - just use your instinct... and experiment!

Synthesizers.com 4- and 8-channel mixers

SOUND GENERATORS - CONCLUSION

As you can see, the oscillators alone offer a huge range of sounds to be created even before we investigate the sound processors / modifiers and controllers. It’s worth getting to know (and understand) the possibilities offered by the oscillators as they are the building blocks of any sound... as mentioned, they can be compared to the strings of an instrument and so play an important part in any sound.

But don’t let this intimidate you - just experiment with different waveform, tuning and mixing combinations until you arrive at something you like and progress from there. Remember…

It doesn’t matter what you do, you can’t break anything by experimenting!

SOUND PROCESSORS / MODIFIERS

Filters

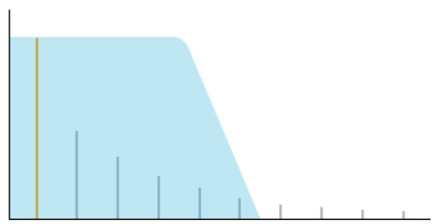
If the oscillators are the ‘strings’ of an analogue synthesiser, the filter is the heart and a synth can stand or fall on the quality of its filter(s).

Put simply, the filter is one big, drastic tone control that can modify the basic sound generated by the oscillators.

We learned about harmonics in the first section of this tutorial - the filter’s job is to selectively filter out (or sometimes enhance) these harmonics, thus changing the tone or ‘timbre’ of the sound. There are many types of filters around that perform different jobs... or rather, have a different effect on the raw sound they are processing. The most common filters are:

Lowpass filter

Allows low frequency harmonics to pass through unaffected, removing higher frequency harmonics above the cutoff frequency:



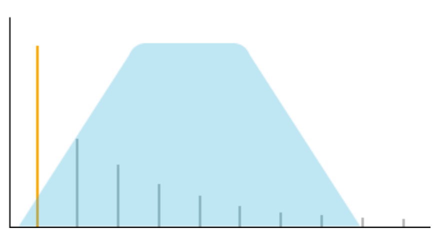
In this example, you can see that all harmonics above the 6th harmonic are ‘cut off’ or filtered out.

This is the most common filter found on ALL analogue synthesisers. It closely replicates nature in that higher frequencies tend to have less energy and so dissipate and die away quicker than lower frequencies (which is why decaying instruments such as guitar, piano, etc., become softer or ‘duller’ as the note dies away).

It is also a natural phenomenon that if an instrument is played (i.e. plucked, bowed, hit, blown - whatever) harder, more high frequency harmonics are ‘agitated’ and so the sound is brighter (and *vice versa* - if an instrument is played more softly, it sounds more ‘muted’). We can use the lowpass filter to mimic these (even if the sound can be overtly ‘synthy’!).

Bandpass filter

Allows a band of harmonics to pass through but removes harmonics either side (below and above) that band:

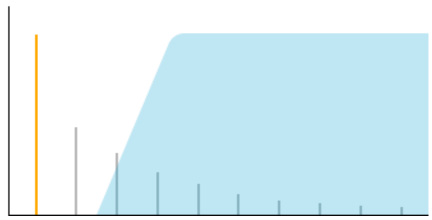


From the diagram above, you can see clearly the effect it will have on the sound - the fundamental is attenuated and harmonics above the 8th are filtered out. As a result, lacking a strong fundamental frequency, the sound is going to be a bit weak and comprising only middle frequency harmonics, can sound bright and ‘fizzy’.

That's not to say this filter is not without its uses - it's a popular filter in many dance/trance/techno genres for creating bright, 'fizzy' leadlines and chordal stabs in anthemic 'Ibiza' dance music.

Highpass filter

Allows high frequency harmonics to pass through but filters out lower frequency harmonics below the cutoff frequency:

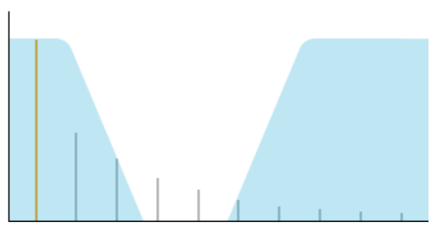


As you can see, the fundamental and second harmonic are filtered out (and the third is attenuated) which will result in a very thin sound.

The highpass filter can actually serve a useful function in a mix or arrangement in that it can attenuate the fundamental slightly to reduce a clash of frequencies (although, in fairness, a simple low cut in EQ on your mixer can do much the same). But they can be used for some characteristically thin sounds.

Band Stop / Notch filter

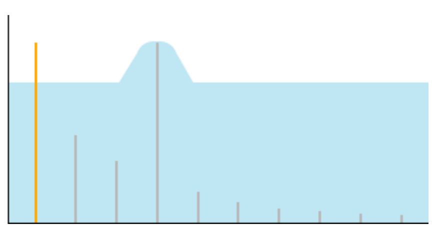
This filter type allows lower and higher frequency harmonics to pass through but removes harmonics in between:



Again, you can see the effect this filter type will have on the sound - the fundamental and the first few harmonics are preserved, some upper harmonics are removed but the upper harmonics remain intact. In practice, the effect of this filter is quite subtle but it can have its uses and when the notch is moved during the course of a note, it can sound like a mild phase shifting effect.

Band Boost / EQ filter

This is not so much a 'filter' (i.e. a device to *remove* harmonics) but more of an 'enhancer' as it actually boosts certain harmonics:



In many respects, this filter type can be used almost as a simple tone control for accentuating certain harmonics in the basic waveform. Its effect is subtle but it does come into its own when its frequency control is changed during the course of a note.

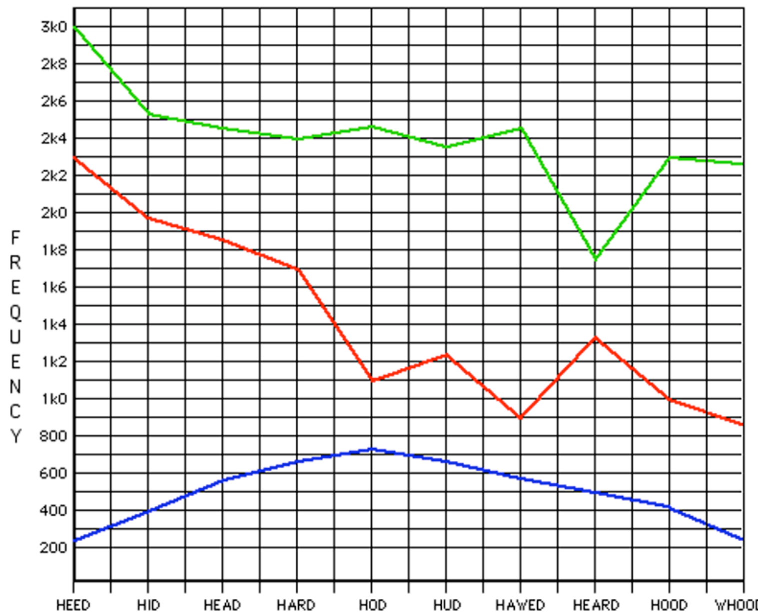
As mentioned, the most commonly used filter type is the lowpass, probably because psychologically it corresponds to what we are used to hearing when playing acoustic instruments - i.e. they tend to get mellower over the course of a note and playing harder or softer creates a brighter or softer sound respectively. But these days, most synths come lowpass, bandpass and highpass filters.

Synthesizers.com filters - one multimode and the other, a more Moog-like transistor ladder filter

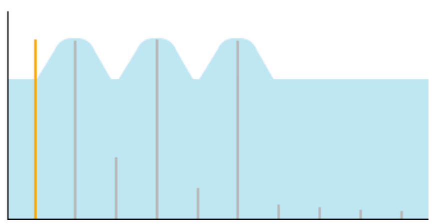
There are other filter types however.....

Vocal Formant filters

The human voice tract has three frequencies called ‘formants’ that shape the vowel sounds we make. These formant frequencies move independently of each other and it is their relative frequencies that create different vowel sounds:



Some software synths and samplers offer Vocal Formant Filters that boost and attenuate harmonics at certain frequencies to recreate human vowel sounds:



When these frequencies remain static, you might hear a certain ‘vocal’ quality but it is when they move that you can hear something approaching vowel ‘movement’ (for example, moving from ‘ooo’ to ‘aaaa’ or ‘aaaa’ to ‘eeee’, whatever).

To facilitate this, a controller such an LFO or modwheel needs to be assigned to the filter’s frequency (which we will look at later) although you can hear the effect when setting these filters’ FREQUENCY control. However, for them to be totally effective, you ideally need independent control over each band and this is not commonly available except on maybe some specialist software synths outside the scope of this tutorial.

Although these filters won’t allow the synth/sampler to ‘talk’ (!!), they do have a curious, eerie vocal quality that can be quite endearing in certain circumstances and with certain sounds.

Fixed Frequency Filters

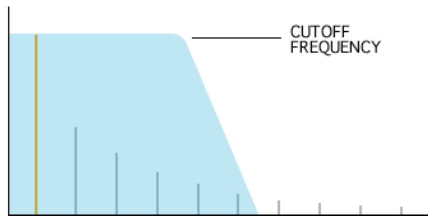
These are typically only found on larger modular systems…

In their way, they are a glorified graphic equaliser but whereas on graphic EQs, the frequencies are typically an octave apart or on larger, more sophisticated ones, 1/3rd of an octave apart, these fixed frequency filters' frequencies are chosen for their musical properties.

Synthesists such as Carlos and Tomita used fixed frequency filters extensively to create some fantastic textures and vocal sounds and it's a shame they are not more commonly available as I can vouch for their effectiveness because I have one in my own modular.

Using filters - Cutoff Frequency

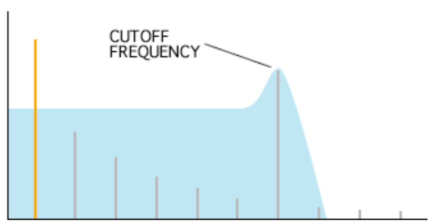
Regardless of their type, all filters pretty much work the same way. There is a CUTOFF FREQUENCY control that sets the point at which the filter starts attenuating:



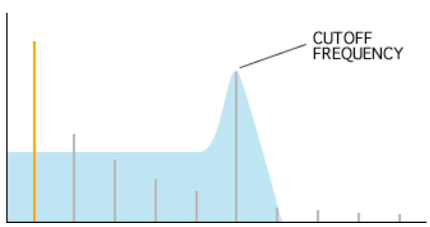
When that cutoff frequency is moved by turning the CUTOFF FREQUENCY control, you will hear the tone of the sound changing. In the example above, when you turn the CUTOFF FREQUENCY control down, you hear the sound getting gradually softer and less bright as the upper harmonics are cut. The exact effect depends on the selected filter type but generally, you hear a ‘wah’ sound as the cutoff frequency changes.

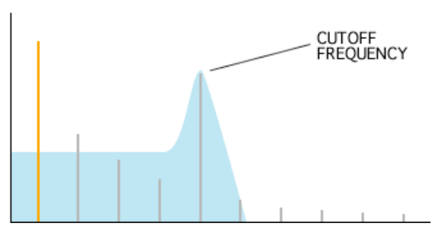
Resonance

Another control closely associated with the filter is RESONANCE (also known as EMPHASIS or ‘Q’ on some synths). What this does is boost the area around the cutoff frequency and has the effect of emphasizing the harmonics at the cutoff frequency:



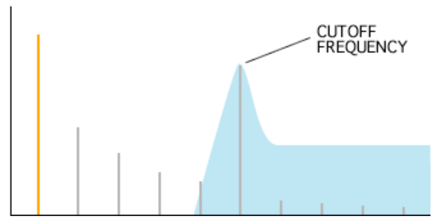
Note two things - not only is the harmonic at the cutoff frequency emphasised but the fundamental is attenuated. As you move the cutoff, so each harmonic is picked out individually especially with increased resonance as shown below:



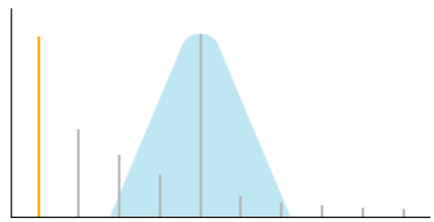


The resulting sound takes on the characteristic synthy ‘weeeeow’ sound as the cutoff changes. With certain higher settings of the resonance control, you can actually hear the harmonics being picked out and individually emphasised.

Of course, similar things happen with other filter types - for example, a highpass filter:



With a bandpass filter, the 'band' becomes narrower with higher resonance settings thus emphasising the harmonics in the area of the cutoff frequency:



Resonance is an intrinsic component of many analogue synthesiser sounds. Unfortunately, in the early days of synthesis (late 60s, early 70s), it was overused (and employed as a gimmick) and so the analogue synthesiser became synonymous with 'duck quack' and 'strangled cat' sounds. However, resonance can be used tastefully to create some truly spectacular synth textures.

Filter slope / roll-off

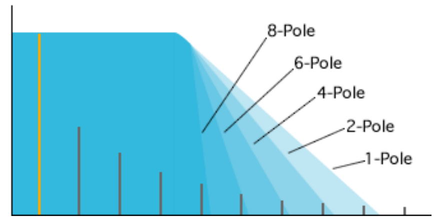
Another (and final!) aspect to filters is their cutoff or 'roll-off' slope.

In the diagrams shown on previous pages, the filter roll-off above the CUTOFF FREQUENCY is at an angle - it is not abrupt and straight down. This is known as the ROLL-OFF SLOPE and on some analogue synths and modern VA synths, it is possible to define this as an adjustable parameter. On the original analogue synths, the roll-off was usually fixed (although some synths did offer a switched option for an alternative). The most common filter slopes were:

12dB/Octave	Also known as a '2-pole' filter. The roll-off is actually quite gentle and gradually attenuates/filters harmonics above or below the cutoff frequency.
24dB/Octave	Also known as a '4-pole' filter, the roll-off is quite steep and attenuates/filters harmonics above or below the cutoff frequency more dramatically.

Of the two, the 24dB/Octave filter was generally preferred as it has a 'punchy' sound and it was common on many American-made synths from Moog and Sequential Circuits. The 12dB/Octave filter, because of its gentler roll-off, allowed more harmonics above/below the cutoff point to pass through and so was regarded by many as a bit weak and 'fizzy'. It was common on many Japanese-made synths but was also adopted by US manufacturer, Oberheim (although they offered a switchable 4-pole filter option in later models). Many Japanese manufacturers followed suit.

Almost any roll-off can be defined and we can have anything from 1-pole (6dB/Octave) to 8-pole (48dB/Octave):



And if you are wondering what the dB/Octave refers to..... it's the amount of attenuation of level per octave.

Thus a 6dB/Octave filter cuts 6dBs (decibels) for every octave and a 24dB/Octave filter cuts 24dBs for every octave and therefore has a more dramatic (some would say 'punchy') effect on the sound.

You can largely forget the technicalities though - the rule of thumb is that a 1-pole (6dB/Octave) filter is going to have a mild effect on the sound whereas a 4-pole (24dB/Octave) or higher filter is going to have a more dramatic effect.

In practice, 2-pole and 4-pole filters are the most commonly used as they are arguably the most 'musical'.

FILTERS - CONCLUSION

To regard the filters as static tone controls is only half the story - they come to life when their cutoff frequency changes over time. This can be done in several ways but almost always involves using a controller of some sort such as an envelope generator, LFO or real-time controller (such as a mod wheel).

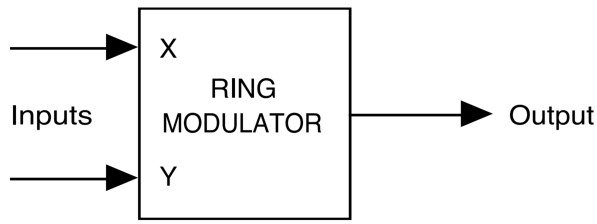
Almost all sounds vary in tone/timbre over time and the filter is the ideal tool to mimic that phenomenon. Even if your intention is not to replicate acoustic sounds, synth sounds can be significantly improved if they also have harmonic movement and change during the course of a note - for example, a resonant synth bass sound can benefit greatly from having cutoff frequency controlled by a decaying envelope as well as having the cutoff frequency controlled by velocity so that the sound is brighter when played hard and *vice versa*.

We will come to this soon when we look at envelope generators and later when we examine 'modulation'.

For now, let's look at another sound modifier.....

Ring Modulator

The ring modulator is a curious device that has been around since the early days of electronic music. It takes two audio inputs and produces the sum and difference frequencies of those inputs at the output:



So, for example, if the frequency of the signal at Input X is 440Hz and the frequency of the signal at Input Y is 1kHz, the output will have 1.44kHz and 560Hz - the sum and difference of the two respectively. Mix in the originals and you have a complex sound comprising 440Hz, 560Hz, 1kHz and 1.44kHz.

Now... because the process is just maths, the sum and difference frequencies are inevitably harmonically unrelated (inharmonic) and thus the ring modulator is very adept at making discordant sounds.

However, worth bearing in mind is the fact that the ring modulator doesn't just produce the sum and difference of the fundamental frequencies of both inputs but also their harmonics. Thus, feeding harmonically complex waveforms such as sawtooth or square waves into the ring modulator can produce some truly discordant sounds to the point of being an unmanageable cacophony! With that in mind, it is often best to just use simple sine or triangle waves with the ring modulator as you tend to have more control over the discordancy.

Used tastefully, the ring modulator can be used to create beautiful, sonorous bell and chime sounds. In fact, there is some confusion about the origins of the name 'ring modulator' - did it come from the ring of diodes that was used on the inputs of the original analogue designs or the fact that it produces inharmonic ringing bell sounds?

The ring modulator can also produce some spectacular early 'sci-fi' sounds reminiscent of the 50s and 60s especially if the pitch of one of the inputs changes during the course of a note. The ring modulator has been around for many decades, actually long before the synthesiser as we know it and as such, many of the sounds you can create with a ring modulator are highly reminiscent of and synonymous with early electronic music from such pioneers as Louis and Bebe Barron who created the first ever all-electronic music film score for the classic movie, 'Forbidden Planet', and also the BBC's Radiophonic Workshop who provided electronic soundtracks and sound effects for many BBC TV and radio production during the 60s including Dr Who.

As the pitch of one of the oscillators moves away from the other, you hear a kind of metallic 'squealing' sound as the inharmonic sum and difference frequencies change over time.

But the ring modulator has other uses.

As an octave splitter for example. Think about it - feed a 440Hz signal into both inputs and the sum and difference is 880Hz (one octave up) and 0Hz (no signal).

The ring modulator was (is) also famously used to create the voice of the Daleks from the classic BBC science fiction series, Dr Who. One input is fed with an audio oscillator generating somewhere in the region of 30Hz and the other is fed with a microphone with an actor speaking the lines (typically "EXTERMINATE"!!). The result is a menacing, robotic vocal effect. Similar techniques can be used using a higher frequency oscillator to create a 'tingling' vocal effect.

And of course, you're not restricted to ring modulating vocals - try drums or guitar... whatever!

Granted, the ring modulator is perhaps not the first module to reach for to create 'mainstream' sounds for your next foray into the hit parade (although the band Japan skilfully employed such sounds in their hit 'Ghosts' using Richard Barbieri's Roland System 700 modular) but it is capable of producing a veritable smörgåsbord of weird and wonderful sounds, especially if very early, pioneering electronica is your bag!

Synthesizer.com ring modulator

Amplifier

Strictly speaking, the amplifier in the audio chain of an analogue synth is not really a 'processor' as such other than it allows you to control level, typically at the end of the signal chain before the signal reaches the outside world. This is usually (if not always in most cases) controlled by an envelope shaper/generator. On some synths (modulars and older analogues), you might find a 'Gain' control that keeps the final amplifier open for drones and such like.

However, the beauty of the amplifier is that its output level can be controlled by other devices such as LFOs. We will come to this in a moment.

Many (if not most) modern VA and software synths' final amplifiers are now stereo and also offer panning allowing you to position and/or spread the signal across the synth's left/right outputs. This may also be controlled with LFOs, etc., for a wide range of dynamic stereo sounds.

On modular synths, voltage controlled amplifiers can be used to regulate the level of almost anything.

Synthesizers.com amplifier and pan modules

SOUND PROCESSORS / MODIFIERS - CONCLUSION

So far, we have looked at ways to generate a sound and then modify that in various ways. In the next phase of our voyage of discovery, we will see how we can change the nature of a sound over time as we embark upon our first foray into the world of controllers.

CONTROLLERS & CONTROL PROCESSORS

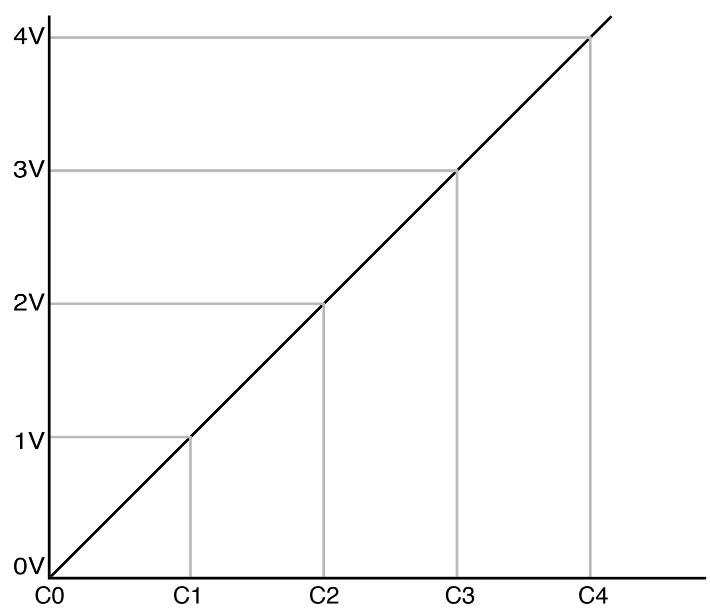
Before we look at the various controllers, we are going to take a trip back in time to take a basic lesson in 'voltage control' because with an understanding of how synths used to work, we can better understand how controllers work in a modern environment.

Today, the late Dr Robert Moog is largely regarded as the father of modern synthesisers. However, synthesisers and electronic music had been around in one form or another for a some time before Moog brought his products to the market. However, these were often test laboratory oscillators and graphic equalisers, ring modulators, simple tape delays, etc., and early electronic music pioneers had to record small snippets of sounds created with this equipment and, using tape splicing techniques, painstakingly 'assemble' a piece of electronic music. It was a laborious and time-consuming affair as you can imagine and a few seconds of electronic music could take days or more to make!!

What Moog did was 'rationalise' the process: by splitting the various elements of sound into different components such as we have so far discussed - sound generators and sound processors. However, what was unique to Moog's synthesisers was voltage control which allowed predictable control of these different components (such as pitch, waveshape, tone, amplitude, etc.).

The idea is simple - apply a varying voltage to the control input of an oscillator and the pitch will change; apply a varying voltage to the cutoff frequency of a filter and the tone will change; apply a varying voltage to the control input of an amplifier and the amplitude/level will change.

Also, Moog devised the idea of the 1Volt/Octave (1V/8ve) rule - i.e. if the voltage doubles, so does the pitch of the oscillator (or the frequency of the filter's cutoff or the amplifier's output level):

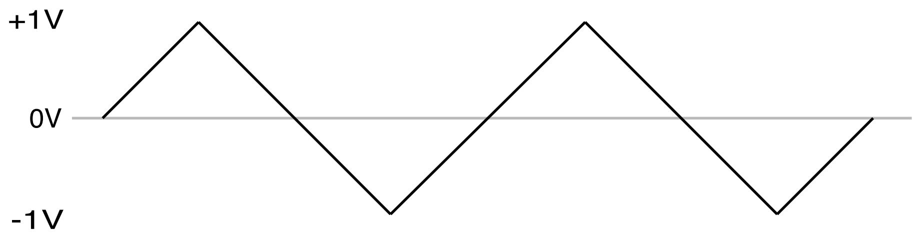


Another synth pioneer, Don Buchla, was also working in similar areas at the time. However, his designs were maybe a bit more esoteric and aimed more at 'avante garde' composers. His synths also used voltage control but they didn't always conform to a predictable 'standard' like Moog's. Korg didn't either with their MS10/20 semi-modular synths. Neither did EMS.

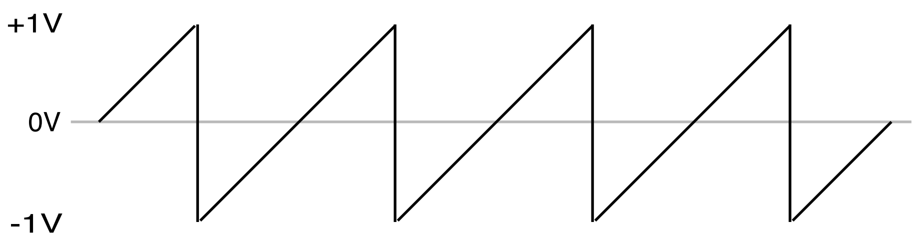
But MOST manufacturers went with the 1Volt/Octave rule

With this ‘rule’, it becomes considerably easier to control sounds. For example, if we have a keyboard that generates 1 volt for every octave, we can ‘play’ the pitch of the oscillators musically.

And what do you think would happen if we had a device that generated a slowly rising and falling voltage that was applied to pitch (i.e. oscillator frequency)? Such as this:

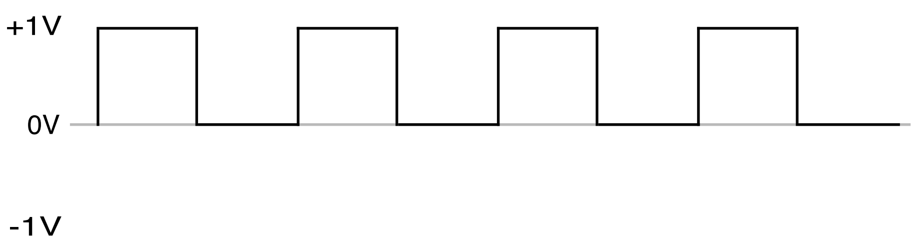


That’s right - the pitch will slowly rise an octave and then fall two octaves and then rise again, etc., as the voltage rises and falls. What about this?



That’s right - the pitch will rise slowly and then drop abruptly and rise again as the voltage slowly rises then drops suddenly.

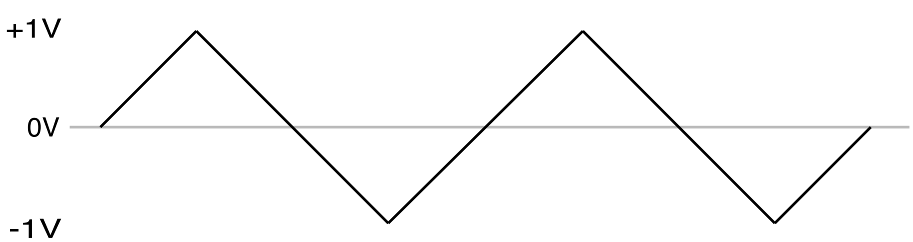
And this?



Correct! The pitch will jump up and down abruptly. In fact, given the 1V/Octave rule, this would be an octave jump/trill between the two extremes.

Congratulations! You now understand how the LFO (low frequency oscillator) works!!!

Now... what happens if we apply this to filter cutoff:

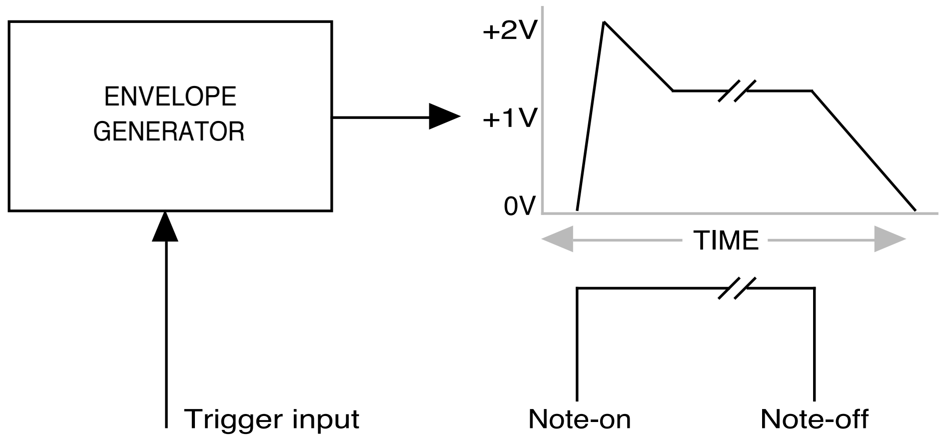


That’s right - the cutoff frequency will rise and fall (or ‘open’ and ‘close’) slowly. In other words, the sound will get brighter and then mellower as the voltage rises and falls.

And what do you think would happen if this waveform was applied to the control input of an amplifier?

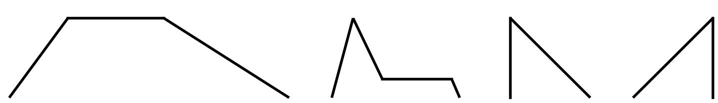
Yep! The sound would get louder and then quieter as the voltage rises and falls!

Of course, an LFO is constantly churning out a repetitive control shape which is just what's needed for some applications (such as filter sweeps, vibrato, tremolo (amplitude modulation) or panning) but we also need controllers that we can trigger ourselves when we want.



When a trigger is received at the envelope's trigger input (typically from a key on the keyboard being pressed), the envelope generator outputs a voltage which rises then falls, sustains at a given value for a while and then dies away to 0V again at the end when the key is released.

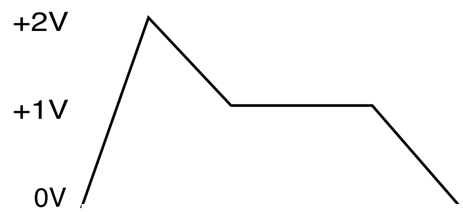
Bearing in mind what we have seen by applying a varying voltage to the control input of an amplifier (i.e. the higher the voltage, the louder the output), you can begin to imagine how the output of an envelope could affect an amplifier. That's right - the sound will get louder then die away slightly at which point it will sustain at a constant level and will then die away to silence. In this way, we can 'shape' a sound, applying different envelopes to the sound that can be slow and dreamy or plucked and percussive or anywhere in between:



However, we can also apply an envelope to filter cutoff - in this way we can vary a sound's tone over time during the course of a note. For example, you will remember that when we looked at harmonics, we discovered that in nature, high frequencies tend to have less energy and so tend to die away quicker than low frequencies and indeed, if you have played a note on a piano or plucked a string on a guitar, you will have heard this - as the sound dies away, it tends to get softer in tone over time. By applying an envelope to a lowpass filter, we can go some way to replicating this phenomenon - when the envelope receives a trigger, the voltage rises (thus opening the filter and allowing all the harmonics to pass through) and as the voltage falls to the sustain level, it will bring the filter cutoff frequency down (thus gradually reducing the higher harmonics over time) and when the note is released, the cutoff frequency will be pulled further down.... as occurs on most acoustic sounds.

These are the most common applications for envelope generators - to shape the amplitude and tone of a sound over time. However, they can also be used for other purposes.

For example, knowing what we know about how pitch reacts to incoming voltage changes, what do you imagine would happen if this shape was applied to an oscillator:

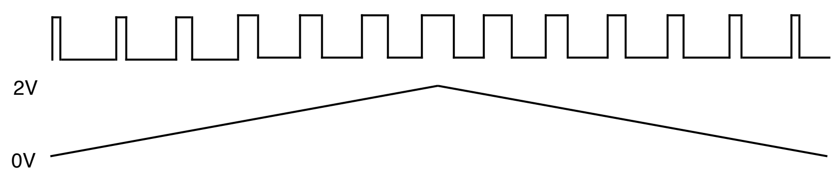


That's right - the pitch would slowly rise two octaves then drop an octave where it would remain constant. When the note is released, the sound would slowly fall in pitch again.
So we're getting a feel for this now. Let's recap....

- The higher the voltage, the higher the pitch when applied to the oscillators (and *vice versa*)
- The higher the voltage, the brighter the tone when applied to the filter (and *vice versa*)
- The higher the voltage, the louder the level when applied to the amplifier (and *vice versa*)

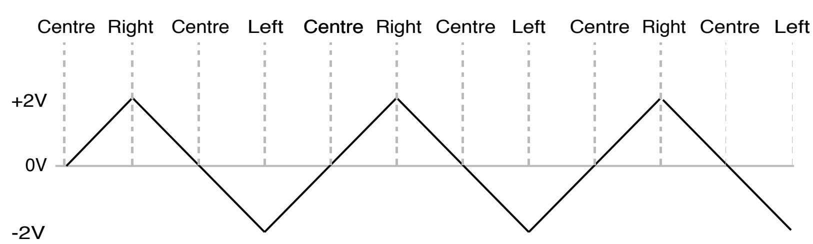
However, it is also possible to apply these controllers to other parameters.

You will remember when we looked at waveforms, we discussed variable width waveforms such as the pulse wave. By applying an LFO to pulse width, you can automatically vary the width:



You could do the same by applying the output voltage of an envelope generator to pulse width.

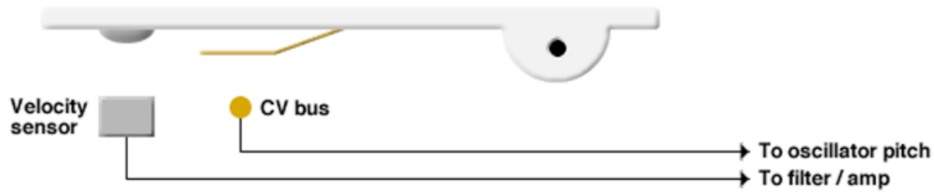
We can also apply these varying voltages to pan position:



But what of other controllers?

Well, given that we now know the basic rule about how voltage can control pitch, timbre, amplitude, waveshape and panning (and other things as we shall see), we can devise other controllers. What if, for example, we place a wheel next to the keyboard - when we move it forward, the voltage it sends to the oscillators is higher and when we move the wheel down, the voltage is lower. You've just added a pitch bend wheel! We can do the same with joysticks, rotary and other controls.

What if we make a keyboard that, when you hit it harder, it generates a higher voltage and we route that to the filter and amplifier?



We have a velocity sensitive keyboard that can control tone and amplitude according to how hard or softly we play it! Of course, because the velocity voltage can be routed anywhere you want, velocity could also be used to control pulse width, panning, even pitch.

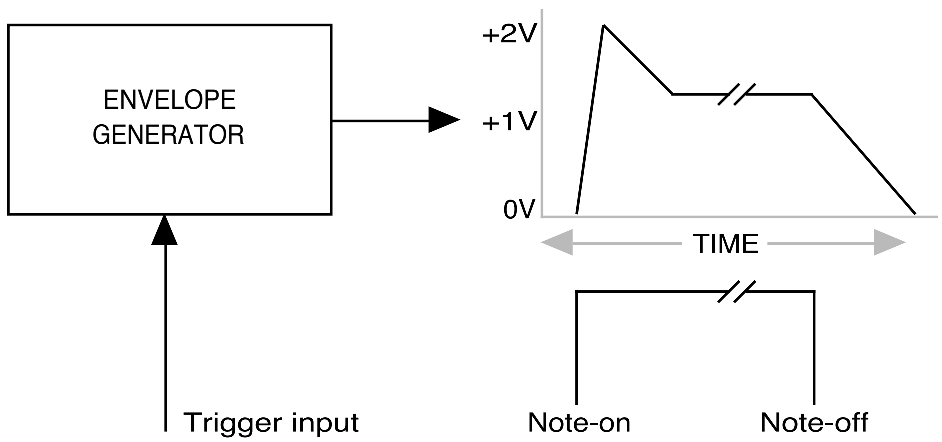
This is the principle of voltage control and it was the backbone of analogue synthesis for decades until it was replaced by the microprocessor after which such tasks were performed not using varying voltages but with digital data streams. And this is why voltage control is described here.

Even though it has been many years since we used true voltage control (except in modern modulars from Synthesizers.com, Doepfer, etc.), the way that digital control is implemented in modern synths is almost identical to the way that true voltage control worked on the original analogue synths but somehow, it is easier to understand the concept of pitch rising as a voltage rises rather than the more abstract passage of 01101010001010 accumulating in a software multiplier! Take the time to re-read this if you are unsure about the principles involved because they are the key to understanding controllers and modulation which lie at the heart of analogue synthesis.

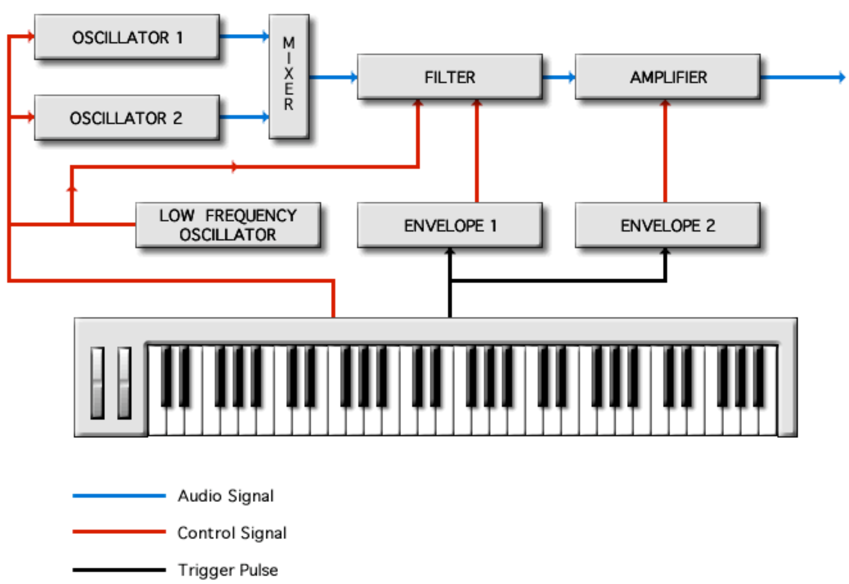
With that under our belt, we can look at the actual controllers themselves.

Envelope generators

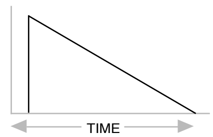
Also known as envelope shapers or (on Moog synths) contour generators, the purpose of the envelope generator is to generate a programmable control shape every time a trigger is received:



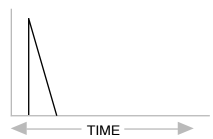
Typically, the trigger comes from the keyboard so every time you press a key, the envelope is triggered. Almost without exception, an envelope generator is connected to the final amplifier to ‘shape’ the sound’s overall amplitude but another envelope generator will almost always be connected to the filter to ‘shape’ tonal variation. Let’s just remind ourselves of the basic synth layout again:



Every sound has an envelope. A piano, for example, has a fast attack and a long decay:

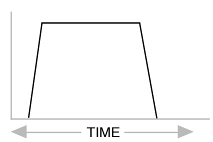


A guitar or harp or bass guitar has a similar envelope. A plucked (pizzicato) violin has a similar shape but is much shorter because there is not enough physical energy in the short, thin strings for it to last long:

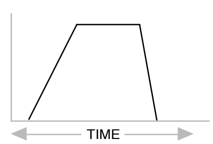


A drum or a marimba and other percussive instruments will have similar envelopes.

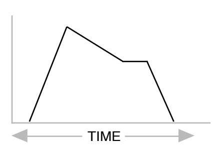
Some instruments, however, have slower envelopes (depending on how they are played). For example, a legato string orchestra will have a slow attack but can sustain:



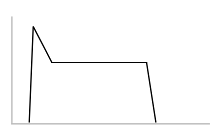
In very slow pieces, the attack can be quite languid:



However, strings (ensemble or solo) rarely sustain at full level and tend to swell and then fade slowly to a sustain:

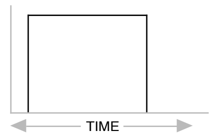


Woodwind tends to have an initial attack that is soft but has an initial 'burst' that settles down quite quickly:



Pipe organs are similar as are many brass sounds.

Of course, the simplest envelope of all is the electric organ - on, full sustain and then off:



This envelope shape is also used on very simple synth sounds.

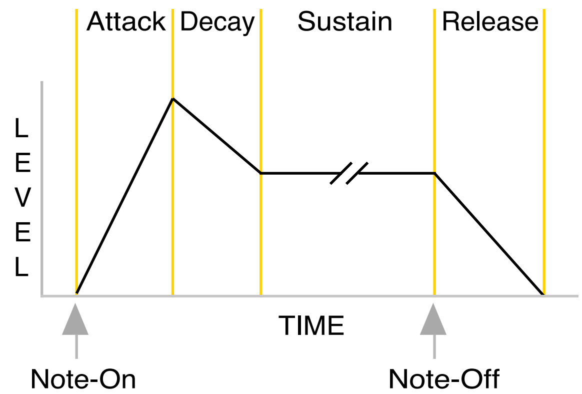
It is the envelope generator that allows us to create all these different shapes and more.

An envelope generator is typically equipped with four controls:

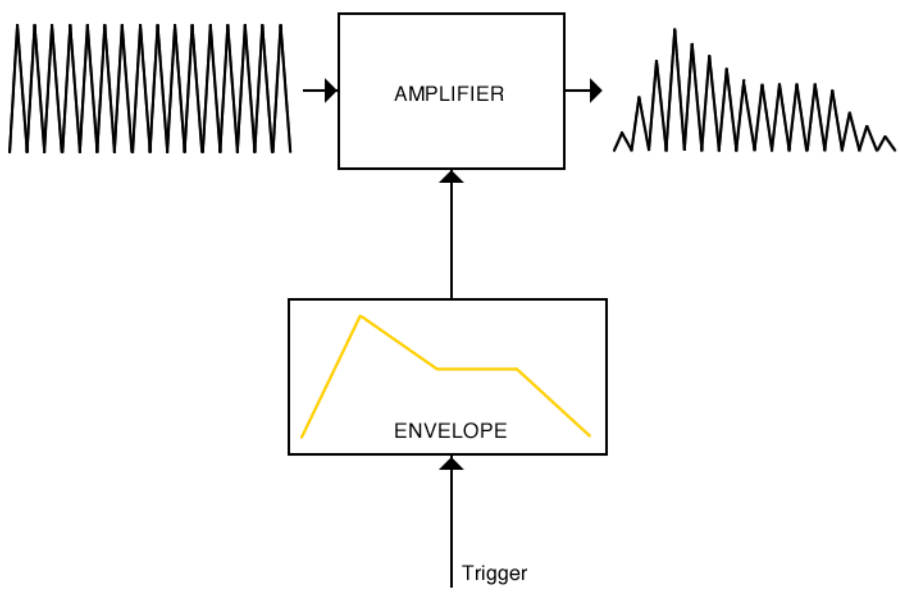
ATTACK	Sets the time it takes for the signal to reach full amplitude.
DECAY	Sets the time it takes for the signal to die away to the sustain level.
SUSTAIN	Sets the level of the sustained portion of the sound (if any).
RELEASE	Sets the time it takes for the sound to die away after the note has finished and you take your finger(s) off the key(s).

Synthesizers.com envelope generator

The Attack, Decay, Sustain, Release phases are often abbreviated to ADSR.



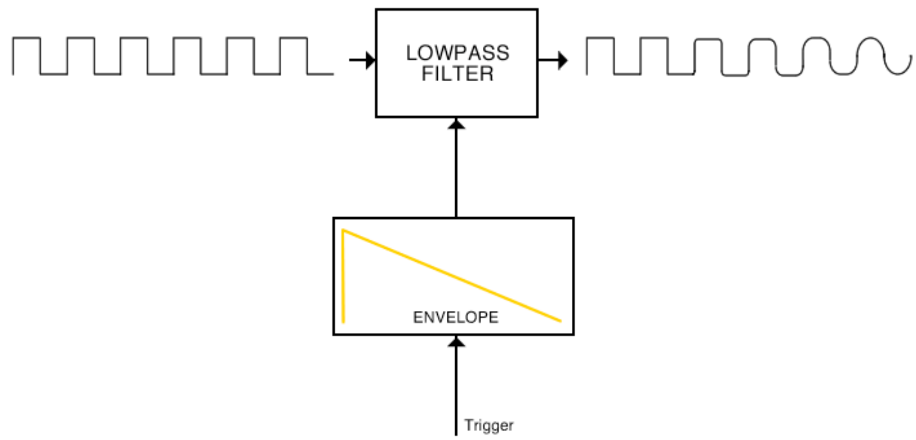
When applied to an amplifier, we can see how it works:



The constant level output from the oscillators/filter is fed to the final amplifier, the output of which is controlled exclusively by its envelope generator. When the envelope is triggered, it opens the final amplifier allowing sound to pass through and the 'shape' of the sound at the output of the amplifier is determined by the control shape of the envelope as set by the ADSR controls. These four controls allow an astonishing number of permutations of envelope shapes to be created and it is surprising how a sound can be transformed simply by trying different envelope shapes - an aggressive bass can become a slow pad whilst a slow pad can be transformed into a spiky arpeggiator sound. In fact, when wanting to make quick, radical changes to a sound, the amplifier's envelope is always a good first port of call.... it is surprising how much this can transform a sound.

The same is true of the filter envelope (if applied) - quite dramatic changes to the sound can be made with simple manipulation of the filter's envelope parameters.

Here's an example of a lowpass filter being swept by a decaying envelope:



At note-on, the envelope opens and so opens the filter allowing the sound to pass through unaffected. As the envelope decays, it brings the filter cutoff frequency down and so the upper harmonics are gradually filtered out and the sharp, harmonically rich square wave gradually becomes a sine wave with no harmonics over the course of the note.

Envelope generators - conclusion
That just about concludes our look at envelope generators for the moment.

On an original analogue synth, this is pretty much all you would need to know (in fact, on an original analogue synth, this is pretty much all there *is* to know!). However, whilst the basic principles still apply, a modern VA or software synth can offer so much more such as velocity (and other external) control of envelope times but to discuss those here might only serve to confuse the issue.

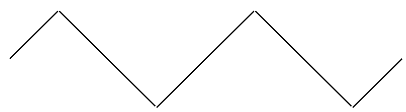
For now, we'll conclude by saying that the envelopes are typically responsible for 'shaping' a sound's tone and/or amplitude - so... get messing with those ADSR controls!!!

Low Frequency Oscillators (LFOs)

In the early days of modular synths, there was no such thing as an LFO - you simply put the audio oscillators into a very low frequency mode so that instead of the oscillators chucking out their waveforms at 100 or 440 or 1,000 cycles per second, whatever (i.e. 100Hz, 440Hz, 1kHz respectively), they produced much slower cycles.... like much, *much* slower cycles... like one cycle every few seconds! In this mode, the audio oscillators were running at fractions of a cycle per second ... like 0.05Hz (one cycle every 20 seconds!).Thus they could be applied to filter cutoff for long filter sweeps. Running slightly faster at, say 6Hz (still sub-audio), they could be used for vibrato (pitch modulation) and tremolo (amplitude modulation) or filter ‘bubbles’.

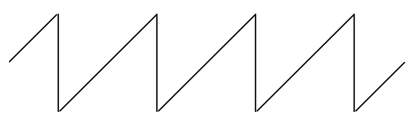
However, as synths developed, it was deemed unnecessary to employ an expensive audio oscillator and so some manufacturers included simpler low frequency oscillators with a limited frequency range purely for the purpose of controlling other modules.

An LFO is almost identical to an audio oscillator except, as mentioned, they operate at a much lower and slower frequency or *rate*. The waveshapes are identical but, because they are running so slowly, you can predict the effect they will have when applied to pitch, tone, amplitude, whatever. For example:

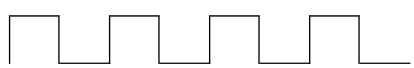


If applied to oscillator frequency, this waveform will cause pitch to rise and fall according to the rate of the LFO.

If applied to oscillator frequency, this sawtooth wave will cause pitch to rise slowly and then fall abruptly according to the rate of the LFO.

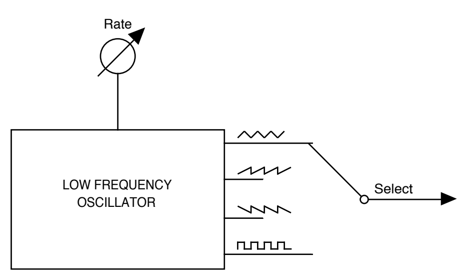


A square wave would cause the pitch to jump abruptly between the two extremes.



As such, LFOs are simple to understand - just looking at the waveshape allows you to predict the effect it will have.

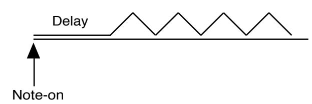
The LFO as a ‘module’ in a modern synth is very simple and looks a little like this:



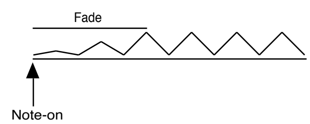
Typically, it offers a triangle wave, rising and falling sawtooth waves and a square wave which can be selected with a switch. Some LFOs offer more waveforms such as sine wave and/or pulse wave (though some offer less - just triangle and square!). The frequency of these waveforms is governed by the RATE control and typically they cover a range of one cycle every few seconds to around 10Hz (10 cycles per second). 5-7Hz is commonly used with a triangle (or sine) wave for vibrato.

And that's about it! Or rather, that used to be about it.

These days, LFOs are far more sophisticated and offer other parameters such as delay/fade. There is, in fact, some confusion between manufacturers about this. Some use DELAY where the onset of the LFO is delayed - i.e. after a note-on, nothing happens and then the LFO kicks in at full level:



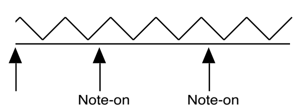
Others, however, use a FADE function where the LFO slowly swells:



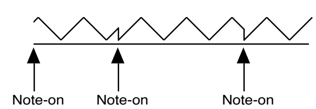
The latter is far more 'natural' for vibrato as this mimics the action of players of stringed instruments such as violin who tend to introduce vibrato slowly after the start of a note.

Some LFOs offer both!

In 'the olde days', an analogue synth's LFO used to be free running in the background and whenever you played a new note, the sound just picked up wherever the LFO was in its cycle:



However, more recent synths offer a 're-trigger' facility where the LFO's cycle is reset to its start with every note--on:



Both have their uses. As an example, the Minimoog didn't have an LFO - instead, you had to use Oscillator 3 in its 'low frequency' setting. As such, when used as an LFO, it was free running with no re-trigger. However, its rival - the contemporaneous ARP Odyssey - had an LFO dedicated to the task but it re-triggered with every new note. Many claimed at the time that the Minimoog had a more 'fluid' sound as a result - vibrato was somehow more 'natural' because it picked up at random in the cycle and slow filter sweeps flowed with the musical phrase.

And of course, in 'the olde days', synths generally had just one LFO which had to be shared between different duties and so it was always a compromise - if you wanted a slow filter sweep, you had to forego vibrato; if you wanted vibrato, you had to forego slow filter sweeps. Many modern synths have two while others may have many more LFOs and, as a result, many things are possible. For example, one LFO can provide vibrato, another can be providing a slow filter sweep whilst another is modulating pulse width at another rate and another can be panning the sound left-to-right at a different rate again!!!

These days as well, the rate of an LFO can be synchronised to MIDI clock allowing filter sweeps, panning, whatever to be in time with the music (in days gone by, we had to guess!!!).

More recently, many LFOs allow their rate to be controlled by some other controller. For example, you could control LFO rate with velocity so that playing the keyboard hard will make the LFO rate faster and *vice versa* ... playing softer yields a slightly slower LFO rate. Or it could be controlled by another LFO or envelope generator or a real-time controller (see below)... endless possibilities.

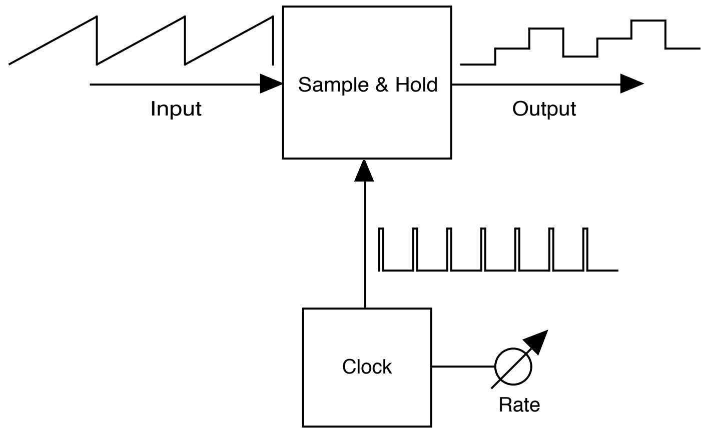
A few synths allow the 'shape' or symmetry of the LFO's waveforms to be altered as well for even more control flexibility but that's probably enough to be going on with for now!!!

A theoretical modular systems LFO.

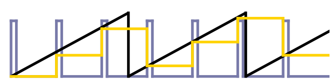
Typically, however, modular systems tend to employ audio oscillators in a 'low frequency' mode.

Sample & Hold

This is another old module from the early days and it falls somewhere between being a controller and a control *processor*. Basically, it takes an incoming control signal, 'samples' it and produces a stepped output according to the rate of the clock:



Here, with a slowly rising control input signal, we get a rising stepped output which, if connected to pitch, would give us a kind of arpeggio effect. This is how they all relate:

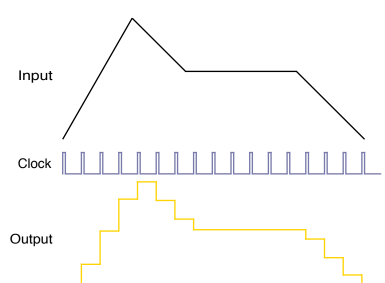


The S&H processor takes ('samples') the level of the incoming signal at each clock pulse and holds that value for the duration of the clock step whereupon it takes the value of the incoming signal at the next clock pulse and holds that level and so on.

So, in this case, the signal is low at the first clock pulse so the output is held low until the next clock pulse. At that point, the input signal is higher so that level is sampled and held. At the next pulse, the input is slightly higher so that is held but at the next pulse, the input signal is lower as is the output level... etc..

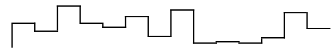
Because the speed of the input signal is not synchronised to the clock, some interesting 'alleatoric' (i.e. unpredictable, almost 'self-compositional') results can be created where the arpeggios are constantly weaving and changing and evolving over time. By playing with the rate of the incoming signal and the S&H's clock rate, some very interesting sounds and patterns can be created and this was a popular technique in the early days of avant garde electronic music composers to create abstract soundscapes.

Almost any signal can be fed into the S&H input. For example, can you predict what the output might be with an envelope signal as the input? That's right:



If the input was the pitch bend wheel, you could create stepped pitch bend (or glissandos).

However, if the control input is a totally random signal such as noise, the signal at the output is a totally random stepped control waveform:



Ideally (and in theory), there should be no 'pattern' and no repetition. When applied to pitch (especially high pitched sounds), this gives the classic (and clichéd!) 'computer' effect (even though no computer makes this sound!!!). However, the random S&H effect can have many uses when applied to filter cutoff for random tonal changes or to panning so that the sound is 'bouncing' across the stereo image at random.

Of course, this is how it all *used* to be done on big modular synths. As synths became simpler over time, S&H was only offered as the 'RANDOM' waveform selection as part of the LFO(s) which, whilst capable of producing the 'standard' S&H random effects was/is ultimately limiting if you want to step beyond that and be more experimental.

Synthesizers.com sample & hold module

Other (real-time) controllers

There are other controllers known as 'real-time' controllers These can take many forms and are:

VELOCITY

How hard you strike the keys affects the sound. Typically, velocity is routed to the filter and the final amplifier to control tone and amplitude respectively. However, velocity can also be used to control other parameters such as tuning, pulse width, panning, LFO rate, whatever.

PITCH BEND

This usually takes the form of a 'wheel' or a 'bender' lever to the left of the keyboard and allows you to 'bend' a note's pitch much like a guitarist bends the strings of his instrument.

MODWHEEL

This usually takes the form of a 'wheel' to the left of the keyboard that allows you to introduce LFO control of the oscillator(s) to add vibrato. However, the modwheel can also be used to open and close the filter or change pulse width or control oscillator sync sweep - whatever. Most synths allow this controller to be totally assignable to almost anything you want.

KEYBOARD

This uses keyboard position as a controller. Normally, the keyboard controls the pitch of the oscillators so that you can play the synth in a conventional musical way. However, it is also possible to use the keyboard to control for example) panning so that low notes are on the left of the stereo image and high notes are on the right. Or it could be used to control pulse width or filter cutoff (or both).... or LFO rate or envelope time.... whatever!

AFTERTOUCH

This allows you to press harder on the keyboard after a note has been played to effect some change in the sound. One example might be to use aftertouch to introduce vibrato... or it could be used to open/close the filter or affect amplitude or panning or pulse width or LFO rate... whatever.

The slight downside to this is that all notes are affected equally. That is, hold down a C Major chord and dig in on the G and all notes will be affected. On rare occasions, 'polyphonic aftertouch' was a feature where, in the above example, only the G would be affected. This is extremely expressive and was arguably one of the best features of the Yamaha CS80 so beloved of Vangelis and heard to great effect in his score for 'Bladerunner'...

Sadly, polyphonic aftertouch is rare (it is expensive to implement) but the old Ensoniq SQ80 also featured it even if the keyboard was a bit noisy and crude.

FOOT PEDAL — This allows you to use a variable foot pedal to add vibrato, open/close the filter, control amplitude, pan the sound left-to-right, alter pulse width, LFO rate/shape.... whatever.

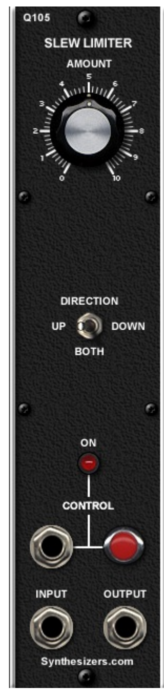
ASSIGNABLES — Many modern synths offer front panel assignable controls whose purpose is to allow certain ‘key parameters’ to be available for quick and easy tweaking and adjustment. These can typically be assigned to almost any function such as envelope attack, envelope release, pulse width, LFO rate - whatever - allowing ‘hands-on’ control of those parameters quickly and easily.

In most cases, the function of the real-time controllers is totally assignable to do whatever it is you want - modwheel and/or aftertouch to open the filter or aftertouch to introduce vibrato, for example.

Portamento/Glide

This is a simple effect found on most synths and basically sets the time it takes for one note to slide to another. Typically, there is just a TIME control which sets the portamento or glide speed and it's just a matter of adjusting that until you get the sound you want.

On modular synths, things are a little complicated and the effect is created using a separate module called a Slew Limiter (also known as a Lag Processor). It has just an input and an output and will process any voltage fed into it. To achieve the classic portamento/glide effect, the input would be the keyboard voltage with the output feeding the oscillator(s). However, the beauty of modular is that it can be used to process ANY voltage thrown at it.



Synthesizers.com Slew Limiter

CONTROLLERS - CONCLUSION

Understanding the way the various controllers affect pitch / tone / amplitude is crucial to getting the most out of your synth. There are no hard and fast rules and so it is difficult to be specific but hopefully the explanations of the various controllers can give you some clue as to the power that is latent within your synth to create a broad range of sounds.

Of course, many of the possibilities will result in totally unusable sounds but this is the power of the synthesiser - you are largely designing your own instrument.

ANALOGUE SYNTH TUTORIAL - CONCLUSION

Hopefully, this tutorial will have given you a deeper insight into the workings of your synth.

Apologies if it has been a bit technical at times but it's only with a thorough understanding of the basic principles that you can begin to explore and understand the depths of your synth whatever it might be. Hopefully now you know why a sawtooth wave sounds different to a square wave or what the different filters are all about.... or why envelope generators and LFOs are important.

The only way to really learn about synthesis, however, is to experiment. Always remember this one thing - whatever you do, you cannot damage or break your synth!!!! Sure - you may do something and end up with no sound (or a total cacophony) but you can always start over.

Besides which, some great sounds are made by accident!

Whatever you do and whatever synth you may have, have fun … which is what it's all about.