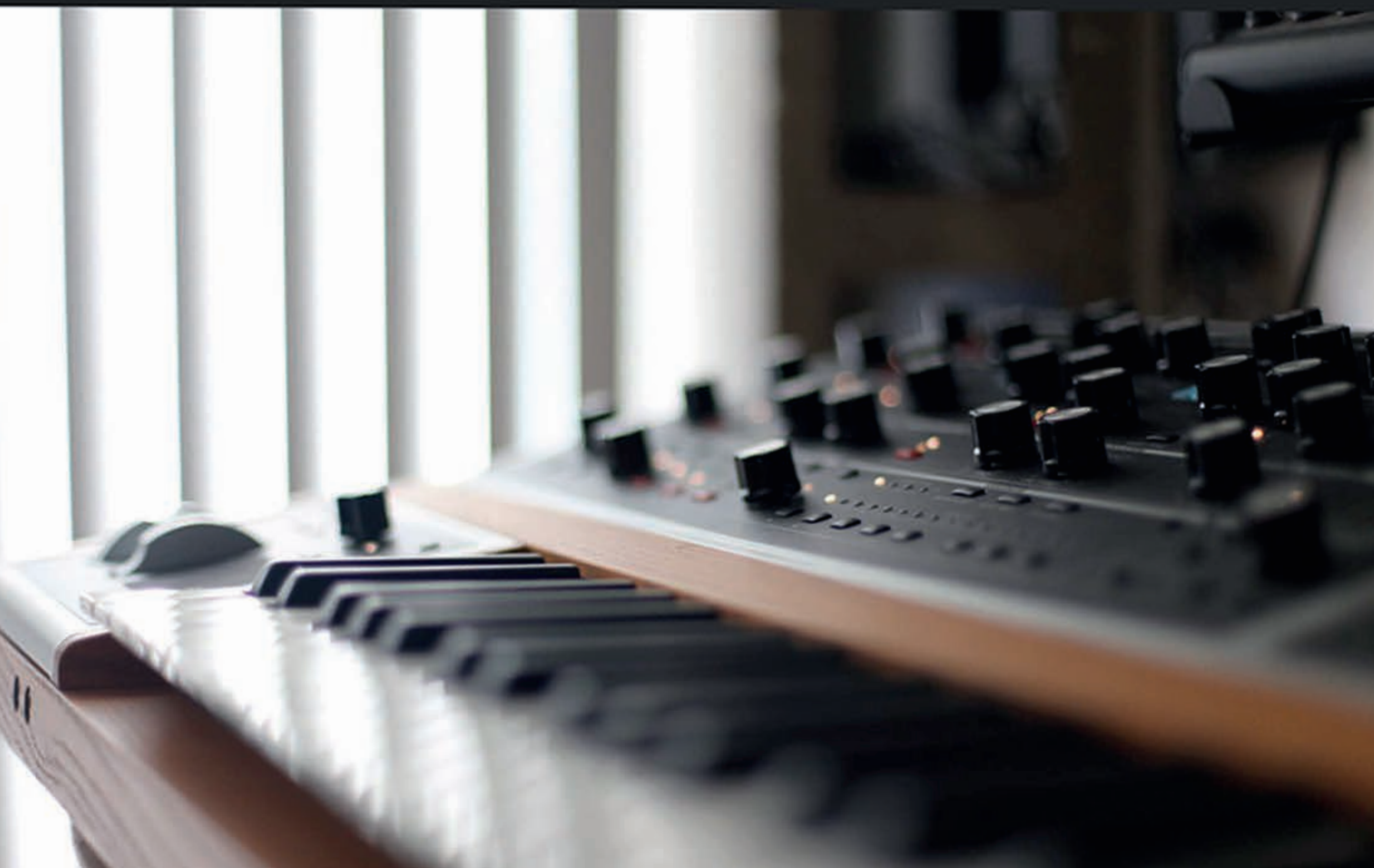
A Focal Press Book

analog synthesizers:

understanding, performing, buying

From the Legacy of Moog to Software Synthesis



Mark Jenkins

second edition

Analog Synthesizers: Understanding, Performing, Buying

Making its first huge impact in the 1960s through the inventions of Bob Moog, the analog synthesizer sound, riding a wave of later developments in digital and software synthesis, has now become more popular than ever.

Analog Synthesizers charts the technology, instruments, designers, and musicians associated with its three major historical phases: invention in the 1960s–1970s and the music of Walter Carlos, Pink Floyd, Gary Numan, Genesis, Kraftwerk, The Human League, Tangerine Dream, and Jean-Michel Jarre; re-birth in the 1980s–1990s through techno and dance music and jazz fusion; and software synthesis. Now updated, this new edition also includes sections on the explosion from 2000 to the present day in affordable, mass market Eurorack format and other analog instruments, which has helped make the analog synthesizer sound hugely popular once again, particularly in the fields of TV and movie music.

Major artists interviewed in depth include:

- Hans Zimmer (Golden Globe and Academy Award nominee and winner, "Gladiator" and "The Lion King")
- Mike Oldfield (Grammy Award winner, "Tubular Bells")
- Isao Tomita (Grammy Award nominee, "Snowflakes Are Dancing")
- Rick Wakeman (Grammy Award nominee, Yes)
- Tony Banks (Grammy, Ivor Novello and Brit Awards, Genesis)
- Nick Rhodes (Grammy Award winner, Duran Duran)

and from the worlds of TV and movie music:

- Kyle Dixon and Michael Stein (Primetime Emmy Award, "Stranger Things")
- Paul Haslinger (BMI Film and TV Music Awards, "Underworld")
- Suzanne Ciani (Grammy Award nominee, "Neverland")
- Adam Lastiwka ("Travelers")

The book opens with a grounding in the physics of sound, instrument layout, sound creation, purchasing, and instrument repair, which will help entry level musicians as well as seasoned professionals appreciate and master the secrets of analog sound synthesis. *Analog Synthesizers* has a companion website featuring hundreds of examples of analog sound created using dozens of classic and modern instruments.

Mark Jenkins has written about electronic music for *Melody Maker*, *International Musician*, *Keyboard Player* (UK), *Keyboard* (USA), and many other publications. He has performed and recorded solo and with members of Tangerine Dream, Can, Gong, White Noise, and Van Der Graaf Generator in the UK, USA, Europe, Brazil, Russia, and China, at venues including the Queen Elizabeth Hall, the London Planetarium, the Carnegie Science Center Pittsburgh, the Vanderbilt Planetarium, and the *Teatro Nacional* in Brazil.

Analog Synthesizers: Understanding, Performing, Buying

From the Legacy of Moog to Software Synthesis

Second Edition

Mark Jenkins

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Dedicated to the memories of Keith Emerson and Edgar Froese.

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Dedicated to those who have gone to meet the great synthesist in the sky: Dr Robert A. Moog, Peter Bardens, Tim Souster, Neil Ardley, Michael Karoli, Ruediger Lorenz, Michael Garrison, Richard Burmer.

About the author

Born in Wales in 1960 and influenced by the early albums of Mike Oldfield and Tangerine Dream, Mark Jenkins independently developed the tape loop recording system also used by Terry Riley and Brian Eno, and started to record in 1978 using an electronic organ and modified frequency generators from his school physics department. His first analog instruments were a kit-built synthesizer and sequencer from *ETI* and *Practical Electronics* magazines.

Gaining an Honours Degree in English Literature, Philosophy, Psychology and History of Science at the University of Leicester, he used the university's music and video studios to create a series of synthesizer and video graphic concerts. Becoming Music Editor of *Electronics and Music Maker* magazine, he reviewed and used all the major instrument releases of the time, including the first commercially available MIDI synthesizer, the Sequential Prophet 600, and met many of the electronic music scene's most influential artists. From 1983 he was a committee member, and later organiser, of the UK Electronica "Future Age Music" festivals.

As Technical Editor of *Electronic Soundmaker* magazine, he used PPG, Prophet and other keyboards for recordings and for concerts at the Festival of Mind, Body and Spirit at London Olympia. In 1985, becoming Technical Editor of the weekly national music paper, *Melody Maker*, he launched AMP Music, which remains the UK's longest-established label for synthesizer, experimental and progressive music.

Releases for various artists on the label were accompanied by concerts at the London Planetarium and elsewhere, and Mark continued to write for music, video and computing magazines, including *International Musician*, *Music Technology*, *Studio Sound*, *Music Week*, *Future Music*, *Sound On Sound*, *Keyboard Player*, *Keyboard* (USA), *Virtual Instruments* (USA), *iCreate*, and *Music Mart.*

Composing steadily for computer games, film and theatre productions, books and virtual reality systems, he performed around the UK and at the Theatre National in Brazil in 1992, starting a schedule of one CD release per year. Mark's first CD, *Space Dreams*, was chosen for sale by the London Science Museum, becoming one of the UK's most popular synthesizer releases ever, while reviews of further albums compared his music favorably to that of Jean-Michel Jarre, Jan Hammer, Mike Oldfield, and Tangerine Dream.

On the AMP Music label, Mark released two albums plus his remix of a CD single for Keith Emerson of Emerson, Lake & Palmer, completing an almost clean sweep of the original innovators of European synthesizer and progressive music with other releases, including albums from White Noise, Tangerine Dream, Daevid Allen of Gong, and Richard Pinhas of Heldon.

Mark Jenkins in his studio in 1993: Korg SQ10 sequencer, ARP Odyssey, OSC OSCar, Moog Multimoog & Sonic 6; EMS Polysynthi, Roland Vocoder Plus, Elka Rhapsody, Yamaha CS80, Moog Liberation; Simmons SDS6 sequencer, Sequential Prophet 600, EDP Wasp, Linn 9000, Oberheim SEM, PPG Wave 2.2, Sequential 700 Programmer, Oberheim MiniSequencer, Korg PS3100 and Casio AZ1; Orla DMK7 master keyboard, Roland SH101, Roland MC202, EMS Synthi Aks, Minimoog, Roland VK09 organ, Roland EM101 module, Moog Polymoog.

Mark Jenkins has recorded and performed with members of Tangerine Dream, Can, Gong, Heldon, White Noise, and Van Der Graaf Generator. He has given multimedia concerts in the USA at the Carnegie Science Center Planetarium, Pittsburgh, the Franklin Institute Fels Planetarium, the Cheltenham Arts Center, Philadelphia, the Ocean County College Planetarium, New Jersey, in the UK at the London Astoria Theatre, the South Bank Purcell Room, and the Queen Elizabeth Hall, and at many other venues.

His most recent releases are tracks on the three CD volumes of *Music for the 3rd Millennium*, on which he also included rare or unreleased music from Keith Emerson, Rick Wakeman, and many others; the multi-synthesizer studio CDs *Sequencer Loops* and *Sequencer Loops 2*; the all-virtual synthesis CD *If The World Were Turned On Its Head, We Would Walk Among The Stars*; the double CD *Live in the USA*; and CD versions of Terry Riley's *A Rainbow In Curved Air*, and Mike Oldfield's *Tubular Bells*, all available through Amazon.co.uk or direct from his own website.

WEBSITES

www.markjenkinsmusic.com
www.youtube.com/markjenkinsmusic

Introduction: what's so great about analog?

How do we usually describe modern high-tech musical instruments? Terms such as "digital", "sampled" or "virtual" are commonplace, though tell us very little about what the instrument actually sounds like. But when we hear the word "analog", we get an instant idea of the type of noise we can expect – a certain sense of power, simplicity, and richness, which dates back to the earliest days of electronic music.

Of course, those times aren't so very far in the past. We're talking about a field which has a modern history of only 60-odd years – so that a few musicians still very active in the field today have been creative throughout the whole period of its development, not something which can be said of the guitar, the folk music group or the symphony orchestra.

Of course, electronic musical instruments, as part of a general high-technology trend which also includes computers in general, have developed at a pace previously unknown in any musical genre. It is amusing to be able to refer as "antique" to an instrument which may have only been launched in the 1960s, remained state of the art for just three or four years, lost almost all value in the 1970s and 1980s, and is now commanding prices, adjusted for inflation, even higher than when it was launched.

But that's exactly what has happened to many instruments in the analog field, and one major reason for the publication of this book is to put into perspective the real value and musical usefulness of these original instruments, as compared to the perceived collector's value, if any, which they may have developed in more recent years.

A second reason is to act as a tuition guide to the techniques and possibilities of analog, which are often now unclear to the latest generations of musicians. In the 1960s it was possible to play for hours with a new instrument and sometimes (particularly when the handbook, if one even existed, had become mysteriously separated from the machine) completely fail to get any kind of musical sound out of it. But the general concepts of analog did become pretty familiar after a while; it was widely realised that most machines were laid out in more or less the same way, and the instruments were mastered by a whole generation of musicians who could then concentrate on making them sound more expressive and genuinely musical.

In the 1970s and 1980s, new generations of instrument designs using new techniques and technologies were introduced, some incorporating aspects of analog sound, but some diverging from it completely. Some of these digital synthesis technologies seemed even more obscure than had analog originally (to be fair, in many senses they could be more powerful and flexible too) and in the effort to master them, many musicians were forced to neglect the simpler routines of analog sound creation. To the extent that, by the mid-1980s, a whole generation of musicians had appeared who experienced as much difficulty mastering analog sound synthesis as those of 20 years before.

So what were these half-forgotten analog sounds, and why are they still worth the effort of mastering them? In the late nineteenth and first half of the twentieth century, various attempts had been made to create electronic musical instruments, mostly keyboard equipped, but these were generally large, heavy and expensive designs. The degree of control they provided over the eventual sound was extremely limited and avant-garde composers, particularly those working in Germany in the 1950s, started to look for smaller electronic components that could be adapted to act as musical sound sources.

The problem, though, with these ad hoc instruments – often adapted from electronic test equipment, signalling equipment or radio components – was one of control, resulting in the creation of a style of music that incorporated some interesting sounds, but that would inevitably tend towards the unconventional and atonal.

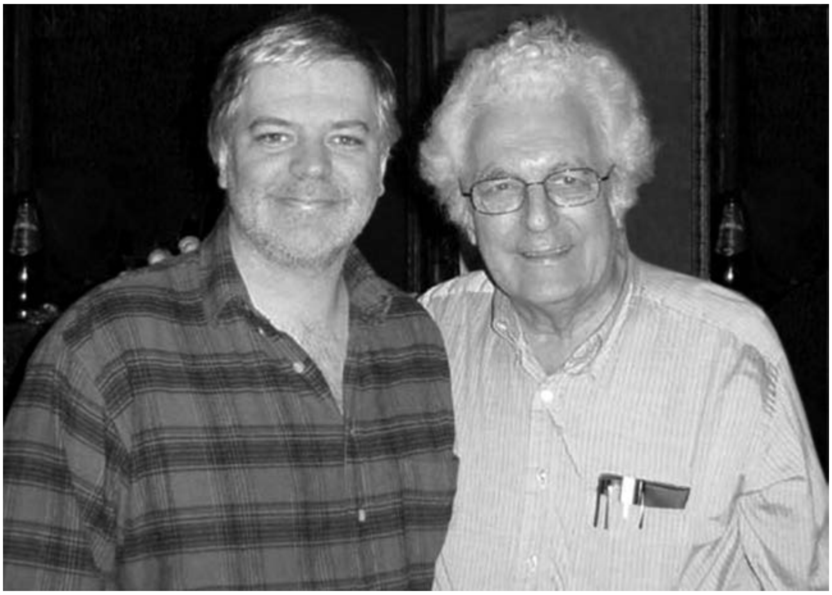
The offer made by analog sound synthesis, starting from the early 1960s, was to combine the easy control of the earliest electronic keyboard instruments with the wider sonic possibilities of the avant-gardists' self-built equipment. This would bring whole new areas of possibility to sound creation without the necessity of abandoning traditional performance and compositional styles – something which appeared highly desirable once the first flush of rock and roll sensibility had passed in the early 1960s.

Although many others had been working towards the same end, the man generally associated with the invention of analog sound synthesis was Dr Robert (Bob) A. Moog, whose early 1960s patents led to his successful creation of a modular analog synthesizer system. Despite the fact that Moog himself was a non-musician, this turned out to be no dry technical achievement; the sound of the Moog synthesizer, beyond offering the relatively easy control that had been desired for some years, was actually surprisingly rich, powerful, and flexible.

The impact of the Moog sound (the name becoming generic for the synthesizer for a while) is well documented, and to some extent has remained the standard for which to aim, despite the passing of Bob Moog himself. The basic tone is rich, strong, and powerful; the Moog "twang" is distinctive (no instrument having previously offered such marked tonal changes through the course of a single note) and the cutting quality of the Moog leadline sound has yet to be excelled.

The impact of the analog synthesizer was more or less instant. Used as much to imitate conventional instruments as to generate new sounds, within only a few years it was raising questions from musicians' unions about possible unemployment amongst their members. Within ten years it was a staple element of all types of popular and experimental music, and it certainly was replacing orchestral musicians in many applications.

Perhaps it was through attempting to compete in the orchestral music arena that analog brought about its own downfall, since the distinctive richness of analog sound is of a quite different sort to that of conventional acoustic orchestral instruments. Whatever the reasons, in the 1980s more "realistic" additions and alternatives to analog synthesis began to appear,



Mark Jenkins with Dr Robert Moog in 2004.

and by the late 1980s there were virtually no analog instruments still in production.

The story of the subsequent "analog revival" is an exciting one, driven by the new, cutting-edge styles of dance music, which were paradoxically calling for the use of sounds that had first been heard some 20 years earlier. As a result, a huge second-user market for original analog instruments came into existence and (after a considerable delay caused by market uncertainty) a few analog instruments went back into production too.

After a few years, when the analog revival was really established as more than a passing enthusiasm, some of the larger Japanese manufacturers came onto the scene, though typically their contributions involved the use of only the most up-to-date technology available. "Virtual analog" instruments initially from Clavia in Sweden sought to emulate the abilities of the classic designs, while offering stability of tuning, enhanced computer control and other facilities denied to the first generations of instruments.

Whether these latest imitative analog instruments match or exceed the abilities and audio quality of those from the 1960s and 1970s is still a subject of massive debate. Certainly, the prices being asked for the classic instruments show no sign of decreasing, except in the case of the handful of models that at some point had become really seriously overvalued.

In this book, the technical explanation of the nature of analog sound creation is followed by the story of its birth and of its subsequent development by various designers, manufacturers and performers.

The individual components of analog sound creation are then examined in detail, with step-by-step examples of sound creation techniques. Then, the modern imitative analog instruments and software instruments are examined, again with detailed information on programming and playing them. This is followed by a new Chapter 8 on the most recent developments in hardware, software, and particularly the booming Eurorack modular field. The book is completed with appendices listing the major instrument lines currently available, hints on values and purchasing, other sources of information, and a discography of readily available recordings that give good examples of analog sound synthesis, including some modern releases on vinyl.

The website accompanying the book, www.routledge.com/cw/jenkins, gives many audio examples of analog sound creation basics, as well as more advanced techniques, and of the abilities of the individual instruments associated with classical and "virtual" analog sound synthesis, and includes video demos, interviews and other material.

The history and techniques of analog sound synthesis make a fascinating study in terms of the development of modern music, the practical techniques available to sound creators today, and of the design and technology of a popular consumer product spanning the last 60 years or so. As such, it is a subject that has long deserved detailed examination.

1

What is analog?

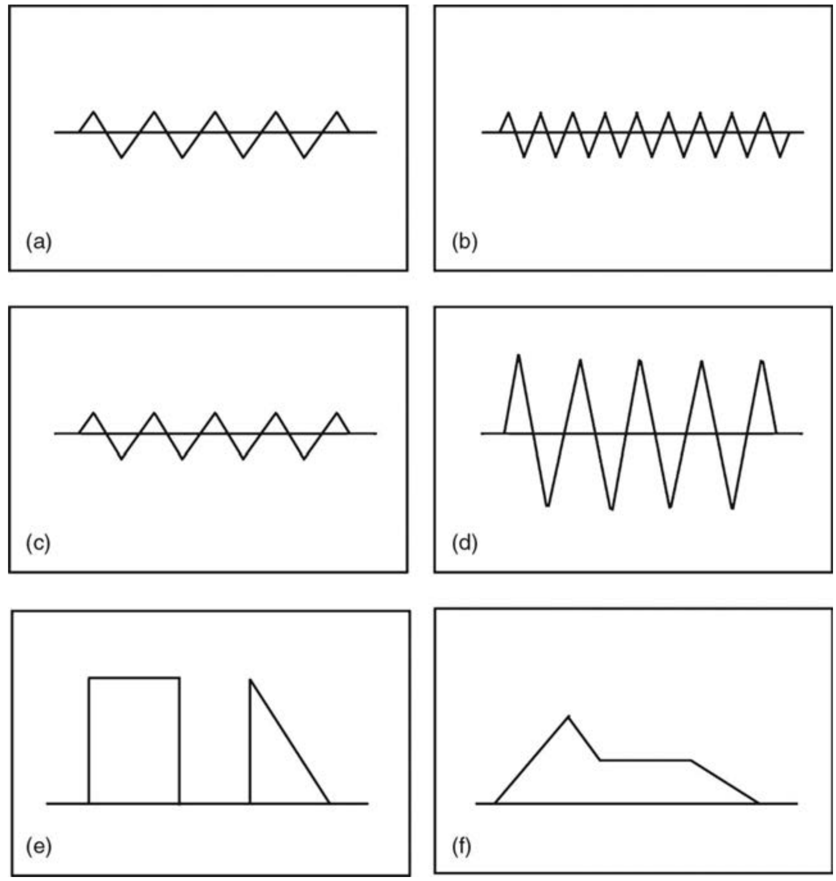
Before starting to look at the creative aspects of analog sound synthesis, it will be a huge help to develop a basic understanding of the principles of physics governing the whole subject of sound – perhaps because of all the methods of electronic sound creation, analog synthesis is probably the closest to those basic scientific principles. If that means a few hours thinking about what sound really is, how it can most easily be created electronically and how it is interpreted by the human ear, then that time spent will be more than paid back through a deeper, more intuitive approach to the handling of analog synthesizers, modules, and effects. So let's briefly take a look at some very basic physics before starting to look at analog instruments themselves.

SOUND

Sounds detected by the human ear only exist because of the medium of the air (sounds can also be transmitted through solid objects or liquids, but for our purposes we're discussing sounds created electronically and ultimately reproduced by some kind of conventional speaker system). A sound is a repeated pressure wave – a more or less regular change in the pressure of air arriving at the human ear, specifically at the eardrum.

600 BC Greek philosopher Thales finds that rubbing amber *(electron)* makes it attract small objects

When a sound strikes the ear, it arrives in the form of a rapid series of changes in the air pressure at that point in space. The most obvious way to create such pressure waves is to move an object contained in the same air medium somewhere nearby. You could do this just by striking two pieces of wood together, but in the case of electronic sound creation, the moving object is usually the cone of a speaker, which moves because it is attached to a magnet encircled by a coil of wire through which an electrical signal is passed – this is the same principle as an electric motor, but designed to create backward and forward rather than circular motion. The electrical signal, and so the sound reproduced, will have three major parameters: frequency, amplitude, and wave shape. Each is explained in the following paragraphs.



(a, b) Representation of a sound with increasing frequency (pitch). (c, d) Representation of a sound with increasing amplitude (volume). (e, f) Representation of simple organ and piano volume envelopes, and a complex volume envelope showing attack and decay times, sustain level and release time.

FREQUENCY

1500 William Gilbert extends Thales' electrostatics discoveries to include sulphur and glass

If one area of increased air pressure hits the ear each second, we call this a one cycle per second (CPS) or one hertz (1 Hz) sound, and the value in Hz of the sound is referred to as the frequency, or, in musical terms, the note or pitch of the sound. In fact, a 1 Hz sound is not audible to the human ear; sounds begin to become audible at around 20 Hz. A very low hum, such as those sometimes generated by electrical equipment, will be at 50 or 60 Hz. Low-pitched instruments, such as bass drums and bass guitars, produce sounds predominantly around 100–200 Hz; 440 Hz is often used as a tuning standard (in musical terms it is an "A" and so is also referred to as A440). Human voices and stringed instruments tend to produce pitches up to around 4000 Hz (4 kHz or 4 k); high-pitched whistles and other instruments will be producing sound up to around 12 kHz, and the highest pitches audible to the human ear are at around 16 kHz (young people manage somewhat higher). Some electronic equipment is designed to handle pitches even higher than this because there is a belief that higher pitched sounds can have a psychological effect, although not consciously audible. Whether this is true or not, it is assumed that equipment that can handle pitches higher than normally necessary will be more easily capable of handling pitches within the normal range of human hearing.

1729 Stephen Grey discovers properties of electrical conductors and insulators

There is a further musical way of referring to pitch: classical church organs create sound using resonant pipes, the length of which were measured in feet. The longest pipes (perhaps 32 ft in length) give the deepest pitch, so on many analog synthesizers the pitches produced are referred to in terms of "footage", as 64', 32', 16', 8', 4', 2' and 1' – a range which covers six octaves or more (pitch increasing by one octave each time the footage is halved). Octaves are the musical intervals most clearly seen in the repeated pattern of the piano-style keyboard, and though the term derives from the inclusion of eight white keys per octave, in Western music there are five black keys per octave to be taken into account as well.

1790 Death of Benjamin Franklin, who experimented with Leyden jars, an early form of battery

Electronic circuits can very easily be designed to create variations in output across the whole range of human hearing and beyond, and we will briefly look at what sort of electronic circuits are used in the analog synthesizer in particular. When a speaker reproduces a sound at a given pitch, it has to be able to vibrate at that pitch, which if it is well constructed, it will readily be able to do. Very small speakers are unable to vibrate at very slow speeds, though, and so the lowest pitches they can reproduce may well be relatively high; we would refer to these speakers as "lacking in bass". Very large speakers can vibrate slowly and reproduce bass frequencies well, but may have difficulty in vibrating very quickly; the highest pitches they can reproduce may be relatively low and we would refer to them as "lacking in treble". For this reason, speakers of different sizes are usually found in combination, to efficiently cover all the frequency ranges within the range of human hearing: usually, a small speaker (or "tweeter") for high frequencies and a large speaker (or "woofer") for low frequencies. Sometimes other speakers are included, designed to handle the middle frequencies ("mid-range" speakers), or the very lowest frequencies ("sub-woofers") and the very highest frequencies ("super-tweeters").

AMPLITUDE

The next parameter of any electrical signal being converted into a sound is amplitude; in other words, the size of the variation in the electrical signal level, and therefore the size of the variation in the air pressure level created, or its volume. A very great change in air pressure, from very high to very low and back again, when repeated, is a loud sound; a very small change in air pressure, from high to low and back again, when repeated, is a quiet sound. Again, in electronic music we are concerned with sounds reproduced by a speaker, and the loudness literally becomes a factor of the amount of air the speaker can move. The cone in a large speaker will actually move several centimetres when in action, and can create pressure waves comprising very large quantities of air. A very small speaker may be able to vibrate at just the same speed, but since the cone is smaller, this vibration moves much smaller quantities of air and the sound is quieter.

1837 First practical electric telegraph developed in the UK by Cooke and Wheatstone

Trying to handle a very loud sound with a very small speaker can simply tear the speaker apart (it can do the same to your ears, in which the eardrum or tympanic membrane is acting like a tiny microphone, the reverse action of a speaker cone); this is even assuming that the electrical coil in the speaker can handle the very wide variation in electrical voltage represented by a very loud sound, which may simply melt the coil through excessive heating effects.

By the way, the "loudness" button found on some hi-fi systems has a related but less obvious function: it slightly boosts the highest and lowest frequencies when listening at lower volumes, to compensate for the fact that the human ear is slightly less sensitive to both of these at low listening volumes.

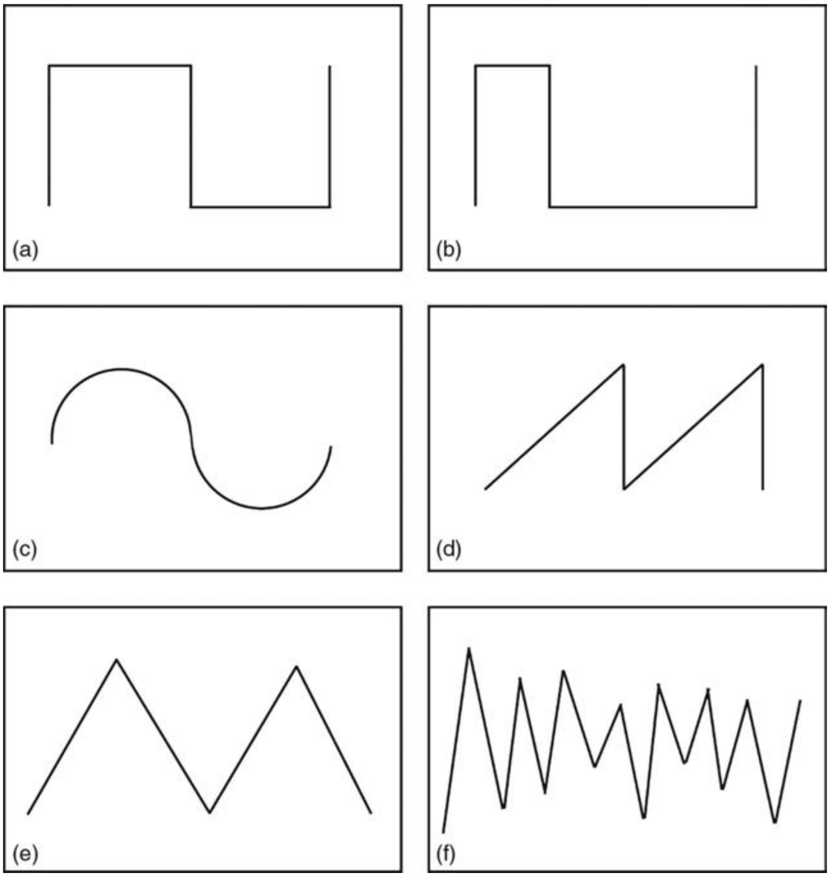
WAVE SHAPE

1867 Death of Michael Faraday, who made important discoveries in the science of electricity

The third major parameter of an electrical signal being converted into a sound is its wave shape. Two sounds of exactly the same pitch and exactly the same volume can sound quite different from one another. What exactly is happening here? The answer lies in the way in which the air pressure varies over time in each cycle. A regular variation that builds up towards a maximum value with its rate of increase slowing as it does so, reaches a peak, dies down to a minimum value, and then begins to build up again can be plotted against time to create a smooth repeated curve, which is referred to as a sine wave. This is the most basic type of wave, the one usually heard from an electronic tuning reference device; it is not particularly harsh or "cutting" and is often compared to the sound made by a flute or whistle.

But it is easily possible to make an electronic circuit vary its output in a quite different way; the signal can build at a constant rate until it reaches a maximum value, then immediately begin to decrease at exactly the same rate. When plotted against time this creates a series of triangular shapes, and so is referred to as a triangle wave. This sound is noticeably different to a sine wave – somewhat sharper and more cutting, and more comparable perhaps to the sound of a bassoon.

A third way of varying a signal is for it to move almost instantaneously from a low level to a maximum level, then fall back gradually to the low level before repeating the cycle at a constant rate. Because of the series of shapes made when this is plotted against time, this is referred to as a sawtooth wave; the opposite, building up at a constant rate then dropping almost instantaneously to the lowest level, can be referred to as an inverse sawtooth wave. Like the triangle wave, both waves sound more cutting than a sine wave, but a little more nasal.



Waveforms: (a) square; (b) pulse; (c) sine; (d) sawtooth; (e) triangle; (f) noise.

1874 Elisha Gray patents "singing telegraph", including an elementary keyboard

A fourth way of varying the signal is to bring it up to a maximum level almost instantaneously, hold it at the high level for a time, then drop it almost instantaneously to the lowest level, hold it at that level for the same amount of time, then repeat the process. Plotted against time, this shows a series of square shapes and so is known as a square wave, and sounds stronger and more cutting than any of the previous shapes. Of course, the time for which the wave holds at its maximum value does not have to be exactly the same as the time for which it holds at its minimum value; the ratio between these two times is known as the mark/space ratio or pulse width, and can be expressed as a percentage, 20% or even 10% for "thin pulse" waves, which sound progressively thinner and weaker as the ratio decreases.

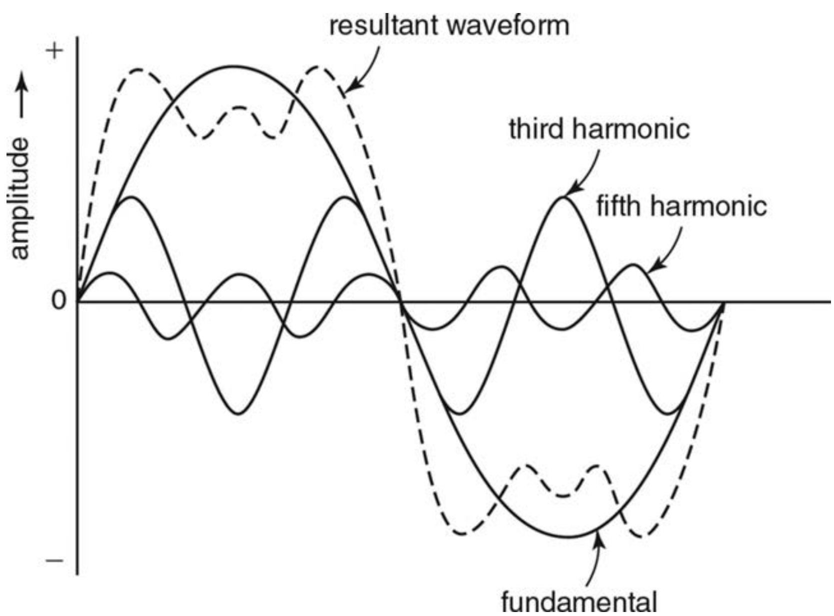
One interesting technique is applicable only to the square wave and does not readily apply to other waveforms. If an electronic circuit is designed so that the pulse width can be altered, it is possible to make the sound vary from thin and weak to strong and rich at will. Doing this under the control of another circuit is referred to as pulse width modulation (PWM), a common technique for making sounds more interesting and for introducing some apparent "movement" within a sound.

HARMONICS AND OVERTONES

1876 Scots-born Alexander Graham Bell patents telephone while at Boston University

There is another way to look at the construction of wave shapes, involving the consideration of "harmonics". When we consider a simple wave such as a sine wave we know its "fundamental" pitch or frequency, as discussed previously. Another wave at twice that frequency sounds naturally "in tune" with the first; we refer to it as the second harmonic and it's musically as well as mathematically related, since it's an octave above (another wave at half the frequency is one octave below the fundamental, and so on). A wave at three times the frequency of the fundamental can be referred to as the third harmonic, four times as the fourth harmonic and so on, and these frequencies are usually generated by musical instruments to some extent, though more quietly than the fundamental.

Interestingly enough, all other waves can be created from a combination of sine waves of different frequencies or harmonics superimposed on one another. To create a square wave from only sine waves, simply generate the basic frequency (the "fundamental") plus all its odd-numbered harmonics – the third, fifth, seventh, and so on. The sine wave quickly transforms into a rounded-off square, and as more and more oddly numbered harmonics are added, becomes a more or less perfect square. The same can be done, using different combinations of harmonics, to generate the triangle, sawtooth and all other waves, and so the tone of a musical instrument or sound depends largely on its content in terms of harmonics.



Building a square wave from sine wave harmonics.

Although some advanced digital synthesizers actually work this way – this so-called "harmonic" or "additive" synthesis method was found on the Kawai K5, Korg DSS1 and a few others – in using analog synthesizers, the common wave shapes have already been created for you, and the four common simple waveforms – sine, triangle, sawtooth, and square (and its variation the thin pulse) – are all we usually need to deal with. But, of course, sounds in the real world are much more complex than this. Even musical instruments that appear to have a fairly simple tone, such as an oboe, are actually producing something much more complex than any of these basic wave shapes. In what ways are sounds in the real world more complex than these simple electronically created sounds?

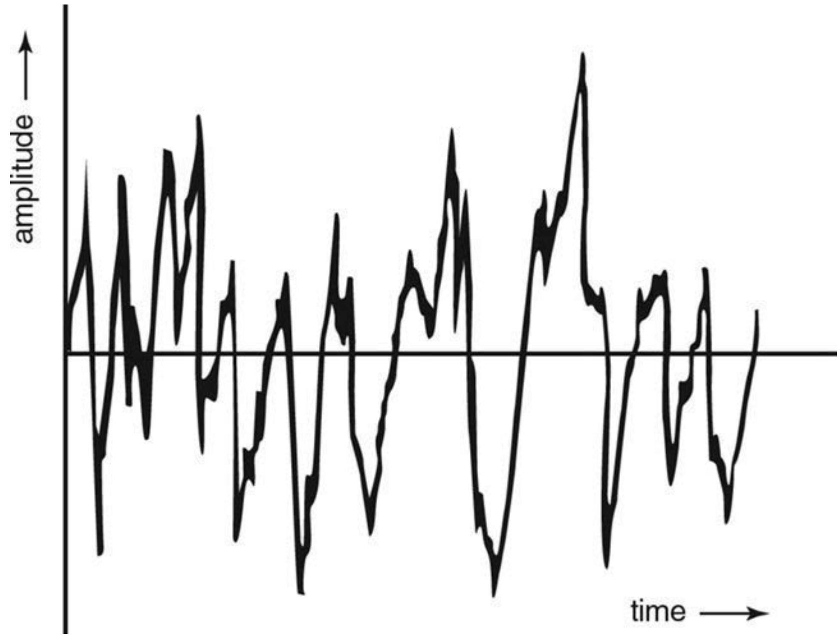
1879 Death of James Clerk Maxwell, who unified laws of electricity and magnetism

The fact is that a pure pitch at a certain frequency, such as can be created by a simple electronic circuit, will almost never be heard in the real world. Suppose a flute (constructed from a combination of wood and metal parts, and "brought to life" by the breath of a human player) is playing a note around 440 Hz. The loudest element of this note will indeed be at 440 Hz, but there will also be quieter elements at twice this frequency and at half this frequency, at three and four times the frequency, and at other intervals that will be mathematically but not necessarily musically related. The basic tone of the flute is similar to a sine wave, but these "overtones" generated at different frequencies tend to obscure this fact, and a printed display of the actual waveform put out by a flute can easily be seen to be much more complex than a simple sine wave. As for other instruments, such as an oboe, for which even the basic waveform is perhaps a more complex version of a sawtooth wave, the added overtones make the picture even more complicated. Both these are relatively simple instruments compared to something like an acoustic piano, which has a complex basic waveform as well as different sets of overtones created by the strings, body, and metal framework, in addition to interference effects created by the fact that strings are arranged in pairs each tuned to slightly differing pitches. So the overall tone or "timbre" of any acoustic instrument is a more complex wave than any of those created by the analog electronic circuits at which we will be looking, and if you combine several musical instruments together, as on a recording of a complete orchestra, the result is a waveform that is extremely complex and that becomes vastly different from one moment to the next.

NOISE

What, then, is the difference between organised sound, or music, and completely random noise? The waveform of a piece of music looks chaotic, so why does it sound tuneful coming out of the speaker? Many analog synthesizers do include a source of truly random noise, the "white noise" generator; white noise is a hissing sound like that coming from a radio tuned between stations, and is often used as a basis for creating sounds of the sea or wind. These applications give a clue to its nature, like the sea or wind, a white noise generator is creating tiny sounds at completely random frequencies and times, and the result has no easily discernible pitch. There are various different ways to create white noise electronically, and in fact it often creates itself and becomes a problem to eliminate completely from some circuit designs, but that is another matter.

1891 Stoney introduces the term electron for the theoretical smallest unit of electricity



The frequency display for random (white) noise.

But all organised sounds or musical output, however complex they may appear, will comprise many elements which repeat in an organised way for however short a time, so while they may appear random, they show much more organisation than a white noise source.

PHASE

A final consideration, particularly relevant to stereo sound reproduction, is the question of phase. Consider a simple sine wave of a particular frequency; it repeatedly reaches the peak of its cycle, then starts to die away again. Another sine wave of exactly the same frequency can also be generated, starting at a very slightly later time, so that when the first is reaching the peak of its cycle, the second is halfway down. The two signals may be of the same frequency, loudness and waveform, but we say that one is out of phase with the other. If the two are added together, they cancel each other to a greater or lesser extent; if they are completely out of phase (we say they are 180 degrees out), they may cancel each other completely, so no sound is heard at all.

This is the principle of the phase shifter or "phaser", an (originally analog) sound effects unit that splits the incoming signal, adds a very short delay to one part so it is out of phase with the original, then mixes the two back together. Making the delay length change gradually alters the degree of cancellation and produces the distinctive swirling, "phasing" effect. But phase cancellation is something to be avoided in music reproduction; if, for instance, one speaker of a stereo pair is used with its connections reversed, one half of the sound will be reproduced 180 degrees out of phase with the other, and while the effects on a complex piece of music may not at first be obvious, you can be sure that the music is not being heard as it was intended.

The conclusion to be made from all these facts is that analog synthesis, in its most basic form, generally creates sounds that are much more simple than those heard all the time in the natural world. With some artistic imagination and technical ability, they can be altered and combined to give good imitations of some simple acoustic instruments, or to create new and less easily describable sounds that have just as much interest as some acoustic instruments. But the techniques of analog sound creation using simple wave shapes are never likely to match the more complex sounds of the real world, which is why other methods of electronic sound creation, such as digital sound synthesis and digital sound sampling, recording and manipulation, were also developed.

SYNTHESIZER COMPONENTS

It is time to take a look at what types of electronic circuitry actually create the sounds heard from analog synthesizers. This is clearly the subject of a whole (more technical) publication in itself, since the subject of electronic circuit design is beyond the scope of this book. But it is enough to say that although developed relatively recently – just over 50 years ago – the basic circuits in a conventional analog synthesizer can be extremely simple.

1896 Thaddeus Cahill develops the Telharmonium to give concerts over the US telephone system

The first element is the one that actually makes an audible sound, the oscillator. As its name suggests, this is simply an electronic circuit that puts out an electrical signal which oscillates, or varies in value, in a regular and repeated way. One such circuit is known as the multivibrator. It can comprise just one or two cheap transistors (which are effectively just electrically controlled switches), a couple of resistors (which, as their name suggests, resist the passage of electrical current to a greater or lesser extent) and a handful of other components. When a steady power supply (of perhaps a few volts, easily obtained from a small battery) is applied to one end of the circuit, the other end of the circuit produces a voltage that varies rapidly in value. By selecting the value of the resistors and other components (or by using resistors that can be varied in value, otherwise known as potentiometers, which are quite simply what lie behind almost all synthesizer front-panel controls), the frequency, amplitude and wave shape coming from the circuit can be varied. Convert this to sound through a speaker and you have a simple (though not very tuneful) synthesizer, with (in musical terms) variable pitch, loudness, and tone. For reasons that we'll examine in detail elsewhere, analog oscillators are usually designed so that these various parameters can be precisely changed by varying the voltage (a measure of electrical potency rather than current flow) of an electrical signal applied to them, and so they are usually known as voltage-controlled oscillators (VCOs).

The history of analog synthesizer design is the history of the electronics engineers who created rather more complex and increasingly musical-sounding circuit designs, and the musicians (or, in some disastrous cases, accountants) who decided exactly how they should be controlled and packaged. We have already looked at the oscillator, but of course there are many other circuits to be found on an analog synthesizer. Perhaps the next most important of these is the filter, a circuit that responds differently to electrical signals of different frequencies reaching it. Trumpet players have been achieving the effect of one type of filter for hundreds of years by simply placing their hands over the open end of the trumpet; the human hand tends to stop the higher harmonic elements of the trumpet's sound from escaping, but does not do so much to cut out the lower harmonic elements. The sound becomes softer, smoother and duller, and this is what we call "low pass filtering", because low harmonic elements are passed through and can still be heard, while high harmonic elements are prevented from passing through and can no longer be heard.

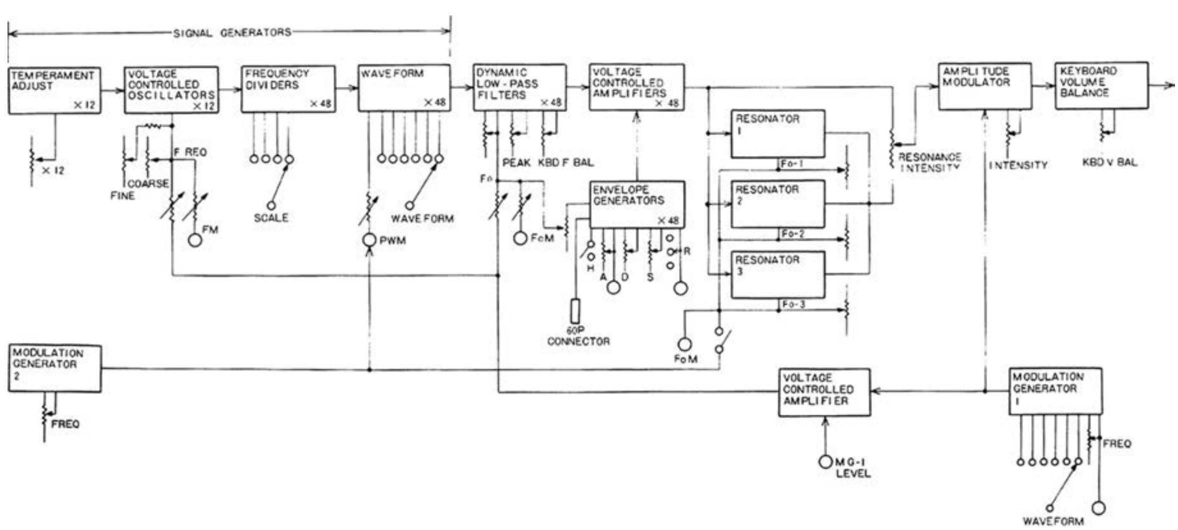


fig. 2 (A) Block Diagram (ブロック・ダイアグラム) PSU-3301 (CHANNEL 1、2、3)

A synthesizer creates sound by connecting various electronic circuits, here, in a Korg PS3300.

A graphic equaliser on a hi-fi system is also acting as a low pass filter if you reduce the level of the top three or four bands and leave the others centred; the result, again, is a smoother and duller tone with less of a "cutting" quality. On a synthesizer, the low pass filter is an electronic circuit that allows through the slow, low-pitched frequencies that are presented to it, but prevents the passage of fast, high-pitched frequencies. The exact frequency at which the filter starts to prevent signals from passing through is referred to as the "cut-off frequency" and can usually be varied; the amount by which signals above the cut-off frequency are reduced in volume, or the power of the filter, is measured in decibels per octave (dB/oct). A 10dB/oct filter (as commonly found in graphic equalisers) will make a frequency one octave above the cut-off point, subjectively half as loud as one exactly at the cut-off point. Some synthesizer filters operate at 12 dB/oct; more powerful ones operate at 24 dB/oct.

You may ask what effect this type of filtering would have on a sine wave-like signal like a flute sound at a particular frequency, and the answer is very little, unless the filter is actually set to cut out frequencies lower than the sine wave itself, because the sine wave-like sound has very few higher harmonics to be affected. But for other types of more complex waveform that have some higher harmonic content, applying a low pass filter audibly changes their tone, and if a sound consists of a good combination of different wave shapes, filtering can change the effect of the sound substantially. If a good-quality filter is set to a very low cut-off frequency, the incoming sound can disappear completely.

An early tape-based analog electronic music studio featuring EMS and Roland synthesizers and banks of graphic equalisers.

Some filters also slightly increase the level of the signal specifically around the cut-off point, this effect is referred to as emphasis, resonance or "Q". If this setting can be varied by the user, setting a high resonance level will start to add a distinct tone to the sound, which will vary just as the filter cut-off position is varied. Using a high "Q" setting has become a staple technique of analog sound synthesis, leading to sounds described as "wet" or "squelchy", and closing down a filter with a very high "Q" setting can be a very dramatic effect. As the resonance setting becomes very strong, a whistling sound is heard at a frequency exactly around the cut-off point and, on some filter designs, going to the highest possible resonance makes the filter begin to act as an oscillator, creating its own (sometimes very loud) whistling sound. This can be useful as a sound creation technique in itself, but has to be handled very carefully as it can unexpectedly create very strong, very low or high-pitched signals that can easily damage speakers. Some synthesizer filters are intentionally designed so that their maximum "Q" setting falls just short of making the filter oscillate.

Filters can also be applied to incoming sounds much more complex than simple combinations of oscillators. If a filter is used to treat a whole piece of music, setting a high cut-off point will make no audible difference to the music. Lowering the cut-off point to perhaps 12 kHz will start to reduce the audibility of high-pitched elements such as whistles and high-hat cymbals. As the cut-off point is lowered, string sounds and human voices will become duller and smoother, and finally very little will be left except bass drums and bass guitars. Close the filter down completely and the whole piece may become inaudible. For reasons that we'll examine elsewhere, filters are usually designed so that at least their cut-off frequency, if not other parameters such as their resonance as well, can be changed by varying the voltage of an electrical signal applied to them, and so they are usually known as voltage-controlled filters (VCFs).

With slight modifications to the electrical circuit, other designs of filter are available, including: "high pass", which acts in the opposite way to a low pass filter, allowing high frequencies through while cutting low frequencies; "band pass", which is a combination of the two, cutting high and low frequencies but leaving mid frequencies unaltered (a graphic equaliser is a series of band pass filters centred around different frequency points, while a parametric equaliser is a single band pass filter with its central frequency variable); and "band-reject" or "notch" filters, which act in the opposite way, reducing the volume of certain frequencies while leaving higher and lower frequencies unaltered. A "comb" filter, used in some effects unit designs, is simply a series of band pass or band-reject filters set to different frequencies but all acting together in parallel. It's possible to build a multi-mode filter that can be switched from low pass to high pass, band pass or other responses.

The third most important circuit found on a synthesizer is the envelope shaper. This is a circuit that varies the level of a signal over a short period of time, and is most obviously used to vary the volume (loudness) of the final sound by applying its output to the amplifier circuit, which is usually a voltage-controlled amplifier circuit (VCA). We have already discussed the pitch, timbre and loudness of musical instrument sounds, but not the way in which the volume varies during the course of an individual note. The easiest example here is the sound of a simple electric organ: hold a key down and a note sounds instantly, release it and the note stops instantly. But this is not the case, for instance, on a guitar; if a string is struck, it certainly begins to make a sound instantly, but the sound does not stop instantly, rather fading away gradually to nothing. On other instruments, such as the violin, the sound does not start instantly either, it may begin quietly, then build up to its loudest volume, then fade away again.

This variation in the volume of a sound is referred to as its volume "envelope", and an electronic circuit that can control this is an "envelope shaper" or "envelope generator". In an envelope-shaping circuit with variable values, the time taken for the sound to build up to its full volume is known as "attack"; the time taken to die away to nothing after the note is no longer being played is known as "release". Other parameters can also usually be set; the time taken for the sound to die down to the "sustain" level, even though the note is still being held, is known as "decay", and the volume level at which the sound continues to play while it is still being held (if any) is known as the "sustain" level. From the abbreviations A, D, S and R for attack, decay, sustain and release, envelope shapers are sometimes know as ADSRs, but note that while attack, decay and release are expressed as lengths of time, sustain is a level represented as a percentage of the maximum available volume achieved just at the end of the decay period. Envelope shapers can be simpler, perhaps just attack-release with optional sustain (AR switchable to ASR), or alternatively more complex, offering further stages of decay or release at different points, for instance, attack-decay-sustain-decay-release, or ADSDR.

Although envelope shapers are most importantly applied to volume, they can also be used to control other parameters. The most usual one is the filter cut-off point, so that the filter cut-off frequency, and so the overall tone of the sound, varies during the course of each and every note. Sometimes a separate second envelope shaper is provided for this purpose, and sometimes the job also has to be done by the volume envelope shaper. Either way, this is a very common technique in analog sound synthesis, creating one of the most distinctive types of analog sound available, and will be looked at in much more detail elsewhere.

It is important also to understand that, to some extent, the distinction between a sound and an electrical signal becomes blurred within the analog synthesizer. If you take the electrical signal from an oscillator and apply it directly to a speaker, you will hear a sound. But if you take the same electrical signal and apply it instead to the part of the filter circuit that controls its cut-off frequency, you will hear the cut-off frequency changing more or less rapidly as a function of the frequency of the oscillator. If you apply the output of an envelope generator to a voltage-controlled amplifier, you will hear a particular volume envelope, but if you apply it to the cut-off control input of a filter, you will hear a change in tone through the course of a note. In other words, a varying electrical voltage can either become a sound in itself or a way of controlling some aspect of a different sound altogether. On a well-designed analog synthesizer, any voltage can be sent anywhere without the possibility of causing damage to an inappropriate circuit, though on some, such as the Buchla designs, audio signals are often made distinct from control signals with different types of cables or connections.

Early compact analog synths such as the Micromoog were perfect instruments for beginners.

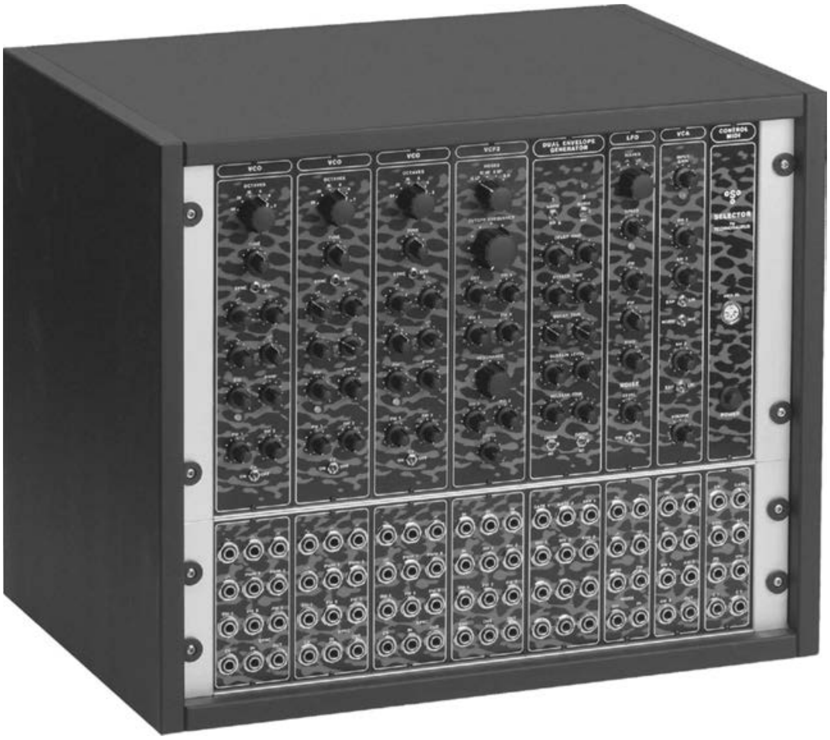
CIRCUIT DESIGN

While the earliest analog synthesizers used small individual electronic components, such as transistors, resistors, and capacitors ("discrete" components) in their circuit designs, these were quickly superseded by ICs (integrated circuits, or "chips"), which in their simplest form comprise just a handful of miniaturised transistors, resistors, and other components assembled into a small, easily handled block. Some of the earliest chips comprised a whole oscillator design in a single component, and it was not long before filters, envelope shapers, amplifiers and even entire synthesizer circuits in the form of single ICs became available through companies such as Solid State Micro Technology (SSM) and Curtis Electro Music (CEM). Although the functional difference between discrete component and IC designs matters very little to the musician, there have been endless arguments about perceived slight differences in sound quality between synthesizers using discrete components and those using ICs, and even between those using different types or versions of an IC. Only one thing is for certain: while discrete components are relatively easily replaced if they fail, some older chips are now no longer obtainable, and so some models of analog synthesizer using ICs can now be extremely difficult to repair.

SOUND DESIGN

We have looked at how the laws of physics define the nature of a sound, and how the relevant parameters are produced and controlled using electronic circuits such as oscillators, filters, and envelope shapers. Understanding what these circuits are contributing to the sound is fundamental to any imaginative use of analog instruments, and certainly to any attempt to imitate particular types of realistic instrumental sounds using analog technology. As will be seen in subsequent chapters, a good combination of basic knowledge of the physics of sound, plus some artistic imagination, needs to go into any attempt to create interesting sounds using analog techniques. For instance, imagine the process of attempting to reproduce a fairly complex acoustic instrument sound, such as that of a plucked string instrument like a bouzouki, using analog techniques. The timbre of the string is quite rich and complex, with many high and low harmonics, so a simple wave shape such as a sine wave will not be of very much help. A combination of richer waves such as sawtooth and thin pulse waves at various multiples of the basic frequency will be of more help, and will add in the overtones that give the instrument its rich sound.

But other factors add to the original sound, such as the resonance from the wooden body of the instrument, which could be simulated with a sub-octave of a smoother sine wave. The overall loudness envelope is like that of a piano, starting instantaneously but dying away slowly, and is easy enough to set. But if the instrument is to sound as if it is being played with a plectrum, the sound of the plectrum also has to be added. This will be



The Technosaurus Selector B was a powerful small modular analog synthesizer.

very short, percussive and at no particular pitch, so a different envelope controlling another unpitched sound source, such as white noise, has to be added in parallel to the first.This sort of complex task is a typical one in advanced analog synthesis, and more examples can be found elsewhere.

Given a basic understanding of the principles of physics behind the creation of sound, and of the type of electronic circuit used in the analog synthesizer to generate and modify these sounds, we are now ready to look in Chapters 3 and 4 at the designers, manufacturers, and artists involved in the field, but firstly in Chapter 2 at how the wonders of analog sound synthesis can actually be applied to making music (or at least musical sounds), and hopefully to having a little fun.

2

Aspects of analog sound

In Chapter 1 we looked at the principles of physics that underlie the electronic creation of sound, and the simple circuits used to do this. In this chapter we will look more systematically at the facilities available on analog synthesizers, and give some ideas about how to apply them practically and imaginatively.

As we will see in Chapter 3, the first analog synthesizers were modular – they were divided into individual circuits, each one carrying out a different task – which had to be connected or "patched" together in order to create a complete sound. This may well have been because Robert Moog solved the problems of designing a voltage-controlled oscillator, a voltage-controlled filter, and a voltage-controlled amplifier one at a time, but whether this is true or not, the fact is that the modular synthesizer has advantages and disadvantages. A disadvantage is that it will not make a sound until some substantial work has been done to set it up; an advantage is that it is extremely flexible and rewards experimentation and imaginative use.

1920 Leon Termen develops the gesture-responsive Theremin in Russia

For practical reasons there was soon a lot of pressure towards the introduction of a non-modular synthesizer, and from the Minimoog onwards the majority of synthesizer designs were not modular, a trend which has now markedly reversed. But even in the early days, the legacy of modular design persisted, even if individual elements of the synthesizer were internally connected and did not need to be patched together, the front panels of instruments still tended to be marked up as if the instrument comprised separate modules. In fact, the Minimoog more or less defined the standard arrangement of facilities on analog synthesizers: one or more oscillators to create a basic sound, leading to a filter to alter its tone, then an envelope shaper to control the loudness and filter setting during the course of a note, and perhaps with white noise or an external sound source mixed in.

1928 Keyboard-equipped Ondes Martenot invented in France, influenced by Theremin

But as early as the time of the Minimoog design, some of the fine details of modular synthesizer design were being obscured. On a truly modular synthesizer, volume levels are controlled by voltage-controlled amplifiers, and to alter a level during the course of a note, these have to be "patched" to an envelope generator. On the Minimoog and later designs, it is assumed that one of the envelope shapers will always be used to control output volume, so the voltage-controlled amplifier for audio output is permanently connected to an envelope generator, not separately accessible as such (see Appendix C for a photo of the more modular Minimoog Model B prototype).

Other modules are also conflated on integrated instrument designs. Some low-frequency oscillators (LFOs) used for modifying pitch or filter cut-off levels will offer a "random" setting as well as the more conventional sine, sawtooth or square wave shapes. But this obscures the fact that random voltages in strictly analog systems were generally created using three separate circuits: a white noise or random voltage source; a sample-and-hold circuit, which sets itself to the level of the white noise source in a fraction of a second and then holds at that level; and the LFO itself, which controls the frequency with which a new sample is taken (of course, the term sample-and-hold is not to be confused with the more modern definition of "sound sampling", though it does have some aspects in common). This sort of design can obscure rather than clarify the facilities available on a particular instrument, for instance, some instruments which clearly must have an internal random or white noise source do not make this available as an independent audio signal.

For these reasons it's best to study the details of analog synthesizer design using a fully modular system, applying the lessons learned to simpler integrated instruments later on. Rather than appearing to favour a current manufacturer, in this chapter many of the examples are taken from the long discontinued Selector system by Technosaurus, a Swiss manufacturer which began to introduce its designs in 1995. The Selector resembled the earliest Moog modular systems using full-sized, quarter-inch jack sockets, which helped make it both more reliable and easier to patch into other professional audio systems than smaller modular systems made elsewhere in Europe and the UK, using mini-jack patching, or US-made designs such as PAiA and Buchla using banana sockets. Also, the Selector placed its interface sockets in groups away from the related controls, so its layouts are very clear. Alongside this powerful but now discontinued system we'll compare the modern Studio Electronics Boomstar SEM, a small MIDI-equipped monophonic module released in 2018, which offers similar facilities in a compact integrated format not requiring patch cables.

1929 Laurens Hammond launches tonewheel organ in the USA

1934 Birth of Robert Moog in Flushing, NY

1939 Second generation of Hammond organs now using valve technology

VOLTAGE-CONTROLLED OSCILLATOR

The voltage-controlled oscillator (VCO) is the basic building block and most frequently used sound source in any synthesizer application, and its output will exhibit at the very least the three most obvious parameters described in Chapter 1, frequency, wave shape, and amplitude, or, in more musical terms, pitch, tone, and loudness.

On most synthesizer designs the VCO would more correctly be described as a voltage-controlled audio oscillator because it is designed to create electrical signals, and so eventually sounds more or less within the range of human hearing. The low-frequency oscillator (LFO), described later, uses a very similar circuit, specified simply to give an output with a much lower speed of variation, and so generally directed towards controlling other synthesizer functions rather than itself creating a sound. On some synthesizers, such as the Minimoog, one or more oscillators are switchable between LFO and VCO applications, but when only audio

1947 Selmer Clavioline invented and licensed for manufacture in several countries

applications are being considered, there are various ways of adjusting a VCO to span only the range of pitches used in normal musical applications.

As we discussed in Chapter 1, the range of human hearing, and so the range of sounds that we consider musical, spans from somewhere below 50 Hz to somewhere above 16000 Hz. As a doubling in frequency represents an increase in pitch of one octave, this represents a range of something over seven octaves – exactly as found on the full-length, 88-note piano keyboard. This is quite a range to be covered with any accuracy by any single variable control, and so most synthesizer designs provide a selector for the rough frequency range, often marked in "footages" (a term derived from the length of the different pipes found on church organs), coupled with a more precise control for fine-tuning.

On the Technosaurus Selector VCO module, the rotary range switch is marked 64, 32, 16, eight, four and two (feet). This control sets the rough range of pitches, while other control voltages (usually from a keyboard that may be three, four or five octaves long and so extend the range accordingly) will determine the actual note played by the oscillator. The 64 (feet) setting represents an extremely deep bass sound, modifiable from the keyboard until it is so low as to apparently change from a musical note into an individual series of clicks. This low setting would generally be used to create very deep bass drones and organ bass-pedal effects.

The 32 (feet) setting is more appropriate for bass guitar, synthesizer bass, and similar applications, while the 16 (feet) setting is appropriate for repeated sequences and accompaniments. The 8 (feet) setting is used for melody parts (perhaps for synthesizing flutes, oboes, strings, and similar instruments), while the 4 (feet) setting would be appropriate for high strings, whistles, and prominent lead parts. The two (feet) setting, particularly when extended higher up the keyboard, is extremely high

The Technosaurus Selector VCO module alongside the SE Boomstar, SEM version.

pitched and only occasionally used in musical applications, or to create higher overtones for lower-pitched sounds.

As described, the overall range control is usually teamed with a fine-tuning control. This is the case for several reasons. Firstly, the very wide range that has to be covered by an audio oscillator usually puts quite a demand on the basic circuit design, and at very high or very low ranges, slight retuning may be necessary. Secondly, random drifting in pitch, sometimes caused by the temperature (or lack of temperature) in the room, has been a long-standing problem for analog synthesizer designs despite many attempts to cancel out its effects by making parts of the circuit run hot compared to average room temperature, and so retuning may be periodically necessary due to temperature-induced drift. Thirdly, when several oscillators are playing together, it is not always optimal to have them perfectly in tune with one another, so the facility to slightly detune an oscillator is usually required. This creates slight pulsing or "beating" in the sound, the result of the oscillator wave shapes adding to or cancelling out each other's output volume as they slip in and out of phase with one another, and this is an important reason why 2-oscillator synthesizers sound much more powerful than single oscillator designs. Listen to track 4 on the website for a demonstration.

On the Technosaurus Selector VCO, the tune control covers a narrow range, just sufficient to correct tuning or to introduce intentional detuning. On some other synthesizer designs, the range and fine-tune controls are combined (on the EMS VCS3 and Synthi A an expensive multi-turn or "vernier" potentiometer is used, which gives better resolution), while on others, such as the Sequential Pro One, the range control covers a somewhat reduced range of octaves, while the fine-tune control covers a much wider range.

1948 Transistor invented by Bardeen, Brattain and Shockley at Bell Labs

The next control generally found on a VCO governs the selection of output wave shape, which determines the basic tone of the sound created. The wave shapes generally available from simple analog circuits are sine, sawtooth, triangle, square and its variation the pulse, and as discussed in Chapter 1, these have distinctly different tones: the sine is extremely smooth and bland, like a flute; the sawtooth and triangle are slightly more cutting; the square wave is distinctly harsher; and the pulse wave sounds progressively thinner and weaker, as its "on" times become shorter.

1950 Robert Moog launches his own business building Theremins

On many synthesizers the wave shape has to be selected for each oscillator individually, so there will usually be a rotary switch giving a choice of one or another – or, in some cases, a variable control that changes the wave shape from one to another continuously. The rather featureless sine wave is sometimes omitted (although the Minimoog does offer it) and the square wave is sometimes fully variable in width using a separate control, or sometimes (again as on the Minimoog) offered in a variety of fixed widths. If there is more than one oscillator on a synthesizer they can almost always be set to different wave shapes, and this is ideal so that a sound could have a smooth element and a more cutting element. But the most luxurious option would be to have all wave shapes available simultaneously, in different amounts, from all oscillators, and this was the case on the Technosaurus Selector. Sawtooth, triangle, sine, and square waves each had their own level controls (so there's no overall oscillator level

1953 Karlheinz Stockhausen creates electronic tape compositions at WDR in Koln

1959 Yamaha launches the first D1 organ model in Japan

1963 Keio Organ company, later to become Korg, builds drum machines for Yamaha

1963 Robert Moog opens Trumansberg factory building Theremins

control as you'd find on most synthesizers), while a separate pulse width control alters the tone of the square wave. Listen to track 2 on the website for a demonstration of the different oscillator waveforms typical of any modular synthesizer.

As mentioned in Chapter 1, a special technique is applicable only to the square wave, referred to as pulse width modulation. Since the tone of a regular square wave is fuller than that of a thin pulse wave, varying this pulse width creates a change in tone, and changing it continuously creates a continuous shifting in tone that can sound similar to chorus or phasing. On most square wave oscillators the pulse width is made voltage controllable, so it can be modulated with an LFO for chorus-like effects, or by an envelope shaper so that the tone of the oscillator varies through the course of a note. It would even be possible to vary pulse width from the keyboard, so the oscillator tone becomes thinner on higher notes, or from a sequencer playing a series of notes, so that it appears to vary in regular patterns; on the Technosaurus Selector VCO there are two inputs for pulse width modulation (PWM), each with its own level control, so two or more of these techniques could easily be applied simultaneously.

Some synthesizers that only have a single oscillator (such as the old Roland SH101) try to strengthen their sound by adding a sub-oscillator. This comprises the sound of the main oscillator divided by a frequency-dividing circuit (more on this later), usually to a pitch either one or two octaves below the main oscillator. This can add to the strength of the sound, particularly in terms of creating strong bass, but it does not have the same effect as a second oscillator since the sub-octave cannot generally be slightly detuned from the main oscillator sound and there is usually no choice of wave shape being limited, due to the nature of the frequency-dividing circuit, to a square wave. On some synth designs, such as the Roland SH3 or ARP Explorer (check track 46 on the website), several sub-octave sounds of different octave values can be faded up simultaneously; this is rather similar to playing several pipes on a church organ together.

We've looked at how the rough and fine pitch of the oscillator and its volume and tone can be selected. The next task is to actually play it in a musical fashion and perhaps to add some expression. Generally, the oscillators on any synthesizer design are controlled from a keyboard and the connection is internal and scaled, so that a one-octave range on the keyboard creates a one-octave change in pitch from the oscillator (on the ARP 2600, for instance, this scaling has to be set by the player). Sometimes (as on the Minimoog and Sequential Pro One), it is possible to disconnect one or more oscillators from the keyboard (on the EMS VCS3 and Synthi A, the oscillators are not controlled from the keyboard at all unless specifically connected, so they drone on one note or are controlled from other sources), but on many integrated synthesizers the connection between keyboard and oscillators is permanent. This led Bob Moog, from his very earliest designs, to establish as standard that a change in voltage of one volt from the keyboard circuit would cause a change in oscillator pitch of one octave, so it can be assumed that a five-octave keyboard creates a voltage five volts higher from its highest key than from its lowest key (but see the note later regarding other arrangements). On very few synthesizers it is possible to readjust this scale so that an octave on the keyboard does not correspond to an octave change from the oscillators, or even (as on the ARP 2600) to reverse the scale, so that playing higher up the keyboard produces a lower note. On the Technosaurus Selector the keyboard voltage, scaled to one volt per octave, is connected to the oscillator through a socket specifically marked CV.

But there is another obvious purpose in controlling the pitch of an oscillator, which is to create a small, regular change in pitch known as vibrato. This is a common technique that can be produced on stringed instruments, such as the guitar or violin, by slightly varying the length or tension of the string in a regular manner, which makes a sound richer and apparently more expressive. A musical-sounding vibrato will vary the pitch by perhaps one-tenth of a semitone (the interval between two notes on the piano keyboard) or less, and at a rate of perhaps 7 Hz (seven times per second), in a regular manner represented by a sine wave or perhaps by a triangle wave. Any great variation from these figures – in terms of depth, speed, or waveform – will sound distinctly unnatural, although this may be exactly the effect required. The regular variation of pitch is also known as frequency modulation. Listen to track 3 on the website for a demonstration.

1963 Don Buchla develops modular synthesizer for Morton Subotnick

On some synthesizers the oscillators can only be modulated, apart from by the keyboard, using the LFO. But there are other ways of controlling pitch too, so having more than one input for oscillator pitch modulation (or frequency modulation, FM) is preferable. On the Technosaurus Selector VCO there are three inputs for oscillator pitch modulation: one fixed and two with level controls marked FM1 and FM2 to adjust their range. If one input is used for vibrato modulation from an LFO, the others can be used (remembering that the additional control input from the keyboard is internal and permanently connected) for alternative control, perhaps from an envelope generator. Chapter 1 explained how envelope generators give a control voltage output that varies over time during the course of a note; applying the output of an envelope generator that starts at a high level and falls away quickly to an oscillator will make it shoot up in pitch then fall rapidly, an effect often compared to an electronic tomtom or "syndrum".

On the Technosaurus Selector VCO there are two further facilities: a CV Off switch, which as we discussed can disconnect the normal control signal from the keyboard; and a Sync switch, which puts the oscillator into synchronisation with another. This means that the frequency of the oscillator is locked to the frequency of another oscillator. This is not a very interesting facility in itself – in fact, it removes some of the interesting effects that can be obtained by playing two slightly detuned oscillators together – but as soon as any attempt is made to change the pitch of the synchronised oscillator, it begins to generate interesting and sometimes quite striking overtones. If the change in pitch is an octave or more, these overtones can become extremely harsh and metallic. Controlling the pitch of the synchronised oscillator with an LFO creates regularly varying overtones; controlling it from an envelope generator creates a harsh, clanging sound on each note; and controlling it from a pitch bender (without bending the pitch of the oscillator to which it is synchronised) creates an expressive, screaming effect that can help generate very distinctive lead or bass lines. Oscillator synchronisation was available on the Moog

1963 Walter Carlos composes early pieces for tape and electronics

Prodigy and is a commonly found custom modification on the Minimoog. Examples of oscillator synchronisation on the website include some of the sounds in tracks 51, 52, 68, and 79.

On the SE Boomstar the VCOs have footages marked from 32' up to 2' as well as a Lo position. Since OSC2 can modulate OSC1, both can also be modulated by the LFO, and OSC Sync is available, a wide range of tuned or more abstract sounds are possible. In this way, many of the possibilities of a fully modular system can be offered by a smaller instrument without the need for patching cables; it is also possible to route MIDI control information to many parameters to vary the sound with performance controllers. Examples are on the website, tracks 83–85.

1964 Paulo Ketoff designs Synket in Rome, later used in performances in New York

VOLTAGE-CONTROLLED FILTER

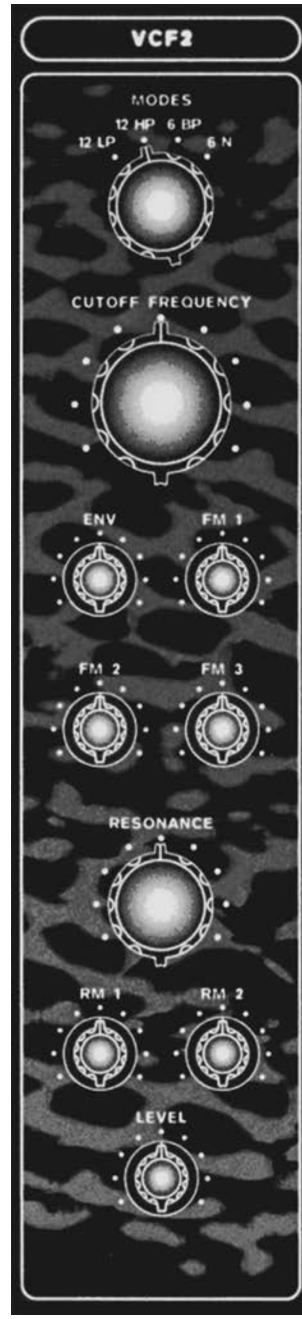
In the standard analog synthesizer configuration, the audio signal from the oscillator or oscillators is passed straight to a voltage controlled filter. As explained in Chapter 1, this circuit acts to cut out certain frequencies from the incoming signal, doing very little to a plain sine wave, but smoothing the tone of a sawtooth, triangle, or square wave, or any combination of these, which contains some higher harmonics. If a powerful low pass filter is set to its lowest cut-off level it can completely silence the signal fed to it; often a source of confusion when first learning the control settings on a new instrument, as it is assumed that either the oscillators are not turned up or that the keyboard is not working.

1964 Robert Moog gives a talk at the Audio Engineering Society on the Voltage Controlled Amplifier

Many integrated synthesizers have a single low pass filter, while modular systems sometimes have separate modules for low pass, high pass, band pass, and sometimes band reject (or "notch") filters. Some filters, known as multi-mode filters, can be switched from one mode to another and on the Technosaurus Selector the VCF2 module could be switched to low pass, high pass, band pass, or notch (demonstrated on track 5 of the website). On the Boomstar SEM version the filter is specifically designed to emulate that of the classic Oberheim SEM module and only has low pass mode, though there is a separate band pass switch, as well as a notch setting for a much wider variation in sound.

The voltage-controlled filter is usually modulated by an envelope and by other voltage inputs. On the Technosaurus Selector the envelope level is controlled from one control and there are three scalable voltage control inputs marked FM1, FM2, and FM3. Exactly as for vibrato modulation of an oscillator, these can be fed by an LFO, a sine, or triangle wave, giving a regular variation in tone, referred to sometimes as "wow" or, if very gentle, as a tremolo. Examples are on tracks 6 and 7 of the website.

The envelope modulation input is used to create one of the most distinctive sounds in the whole field of analog synthesis – a sound that changes very markedly in tone as the note progresses, something highly unusual in the world of acoustic instruments. This is particularly striking if the filter's resonance is turned to a high setting. Check out track 26 on the website. Track 27 has a similar effect "upside down", with the envelope sweeping the filter upwards instead of downwards.



The Technosaurus Selector VCF module alongside the Boomstar SEM's filter section.

On some filter designs, resonance can also be voltage controlled, and on the Technosaurus Selector VCF there are two scalable inputs for resonance modulation marked RM1 and RM2. Finally, there is an output level control that controls the audio output level from the filter, which you can use to "make up" some output volume if you strongly filter the incoming sound. On the Boomstar, switches route the output of VCO2 or the LFO to control the filter and determine the degree of keyboard tracking, MIDI controllers can also be routed to the filter.

ENVELOPE GENERATOR

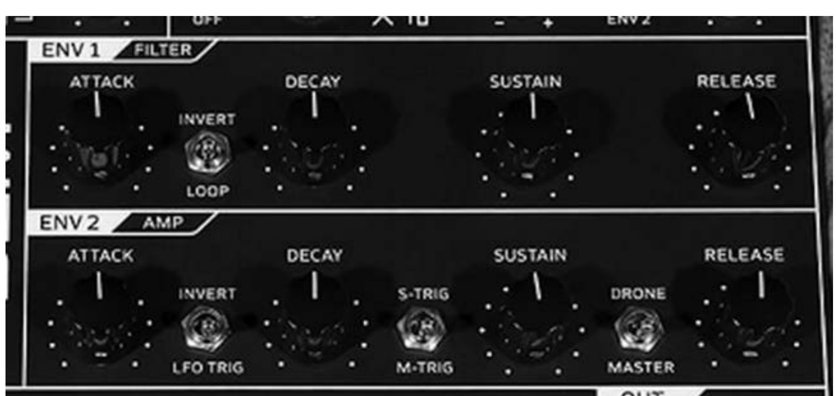
Envelope generators differ from VCO and VCF designs in that they control other functions but generally do not have voltage or audio inputs other than for a voltage trigger or "gate", a brief, relatively high voltage "spike", which starts the envelope generator (EG) going through its cycle (though more advanced EGs with voltage controllable parameters are also available). The most obvious source of this spike is a keyboard. Every time a note is played the keyboard generates a trigger or gate that sets off the envelopes, which ensures that, usually alongside adopting a particular pitch, the new note also has its own loudness and filter envelope. But it is also possible to trigger envelope generators from other sources, sometimes from an LFO on every cycle, sometimes from an external source such as a sequencer, as described later.

As discussed more fully in Chapter 1, envelope generators typically have four parameters for attack, decay, sustain, and release (ADSR). These are all periods of time, other than sustain, which is a level expressed as a percentage of the full envelope level reached at the end of the decay phase.

As mentioned, envelope generators can sometimes be more complex and on the Technosaurus Selector system there is a dual envelope generator in which each envelope also has a delay phase, so it can be described as DADSR (there is something similar on the Korg MS20). This simply means there is a variable time period after the note is played, before the envelope triggers, so, for example, you could create a rather muted sound which abruptly brightens up after a second or so. On the Selector each envelope can also be switched to a simpler AR format, and there is a gate input, and normal and inverted voltage outputs for each envelope. For convenience, there is an LED light to indicate each time the envelope triggers. On the Minimoog, the envelopes only have ADS stages, if you switch in the release phase it has the same length as the decay stage. Bob Moog corrected this limitation and others on both the Crumar Spirit and the Moog Voyager.

On the SE Boomstar the filter envelope, as mentioned before, can be looped, while the VCA envelope can be repeatedly triggered from the LFO, so smaller integrated instruments aren't denied this wide variety of options. On some synthesizer designs there is a choice of overall rates for the envelope, speeding up its period to cycle, while some keyboards will offer a choice of ways in which the envelopes can be re-triggered: on the high, low, or any new note, repeating with a key down only, or repeating at the speed of an LFO, as found on the ARP Odyssey. It is annoying to sometimes find modular designs in which the LFO output is not enabled to repeatedly trigger the envelopes, though it's sometime possible to shape and amplify the LFO waveform into a trigger pulse that the envelope generators will accept. See the section on keyboards for more details.

Envelopes are generally applied to the VCA to alter volume during the course of a note and to the VCF to alter the tone during the course of a note. Examples of varying volume are on tracks 8 and 12 of the website, examples of varying tone are throughout – on tracks 22, 29, 34, 35, 63, for instance – as this is a very common technique of analog synthesis.



The Technosaurus Selector dual EG module and the Boomstar's envelope section.

Voltage control of the individual ADSR parameters of an envelope generator is more rarely found, although the idea of being able to vary the attack or decay time of a note using an LFO or sequencer pattern is interesting, as you could create patterns of notes that automatically or randomly vary from the slowly swelling and sustained to the brief and staccato.

VOLTAGE-CONTROLLED AMPLIFIER

The VCA generally has few functions as it is normally connected directly to one of the envelope generators, so it often appears just in the form of a final volume control. On the other hand, if it is required to create a genuine tremolo, in which the output volume varies rapidly just as the oscillator pitch would vary rapidly in a vibrato, then it is useful to be able to access the VCA independently. On the Technosaurus Selector the VCA was a module separate from the envelope generators and comprised an input level control with a dual-colour LED that indicates if it is overloading (which would cause audible distortion), four audio inputs, three voltage inputs with scaling controls for amplitude modulation (AM), exponential/linear selection, and normal/invert switching. In this way the VCA can be used to combine the output of two, three, or four oscillators, control their levels together using an envelope generator (the relative levels of the oscillators are mixed at the oscillator output, since there is no audio mixer as such in the system illustrated) and add tremolo using an LFO, at the same time as using another level control such as a pattern of differing levels sent from a sequencer. The invert setting can be used to set up two VCAs to opposite phases to create stereo panning.

On the SE Boomstar, typically of smaller integrated instruments, the VCA isn't separately marked, though there is a master Volume control and on the top panel a minijack socket for external voltage control marked AMP AM – amplifier amplitude modulation – offering either control of volume from a voltage pedal or regular tremolo using an external LFO. There is also a Drive setting, which slightly overdrives the output VCA for musical distortion, an incidental though popular facility of the original Minimoog, here made much more explicitly available.

LOW-FREQUENCY OSCILLATOR

As previously discussed, the LFO is simply a voltage-controlled oscillator designed to operate at lower speeds and intended for use as a control source rather than an audio source. On the Technosaurus Selector LFO, the wave shape was selectable using a rotary control for sawtooth, triangle, square, or sine waves, but these are also available simultaneously at independent voltage output sockets. The speed is variable and the pulse width of the square wave can be varied manually or automatically using

The Technosaurus Selector VCA module.

a voltage at the PWM input socket. Unusually, the LFO speed itself is also voltage controllable, so this is more properly a VCLFO or voltage-controlled low frequency oscillator. This can be an interesting facility because the speed of the modulation created by the LFO (whether on a filter or an oscillator) can itself vary regularly and all sorts of complex patterns can be created.

On the Boomstar, although the LFOs shape control is a continuously variable knob, it offers nine individual shapes, including Random, with flexible routing to the pitch and pulse width of both oscillators, the cutoff of the filter, and the repeat of Envelope 2. The LFO rate can also be synchronised to an incoming MIDI clock. So again, an example where most of the facilities of a full modular system are readily accessible, though some more advanced features such as voltage controlled speed and waveshape are not.

The differing waveforms output by an LFO are useful for different purposes. The sine or triangle is most often used to create vibrato on any oscillator, tremolo/wah-wah effects on a filter, or chorus-like pulse width modulation effects when applied to the pulse width control input of a square wave oscillator. A square wave LFO output, however, produces trills when sent to an audio oscillator (tuned ones, if they are scaled correctly), while if there is a thin pulse or other LFO wave shape available, this has fewer obviously musical uses. On the EMS synths the envelopes could re-trigger automatically (referred to as a trapezoid generator) and in this case themselves become LFOs capable of outputting extremely unusual shapes, this facility re-appears on the Boomstar, given its ability to play the filter envelope in repeated Loop mode.

Some LFOs, for example, on the Korg MS20, also output a random waveform, useful for creating bubbling, windchime, and similar sounds. The availability of this output from the LFO waveform selector often obscures the point that it is created by a combination of the LFO, a source of white noise, and a sample-and-hold circuit all working in concert (more on this later), though in modern designs while the audio oscillators may be truly analog, the LFOs are often digital and the random waveform is generated using any one of a number of digital techniques (and may not be all that random, sometimes it's possible to hear an apparently random pattern actually repeating over a fairly short period).

Listen to tracks 3, 6, and 7 on the website, which illustrate simple modulation of an oscillator's pitch from an LFO, simple modulation of a filter's cut-off frequency from an LFO, then more complex filter modulation from several LFOs simultaneously. Very fast modulation of either pitch or filtering can sometimes create human voice-like effects, as heard on tracks 9 and 15; the second sound of track 31 features square wave pitch modulation, while most of the sounds elsewhere that exhibit regular variations of pitch or tone will also depend on the use of an LFO.

On the Technosaurus Selector LFO module there was also a noise source, switchable from white to pink, which appears at an audio output socket with output level governed by its own control. While several synths group the LFOs and white noise source together, there is no particular necessity for doing so.

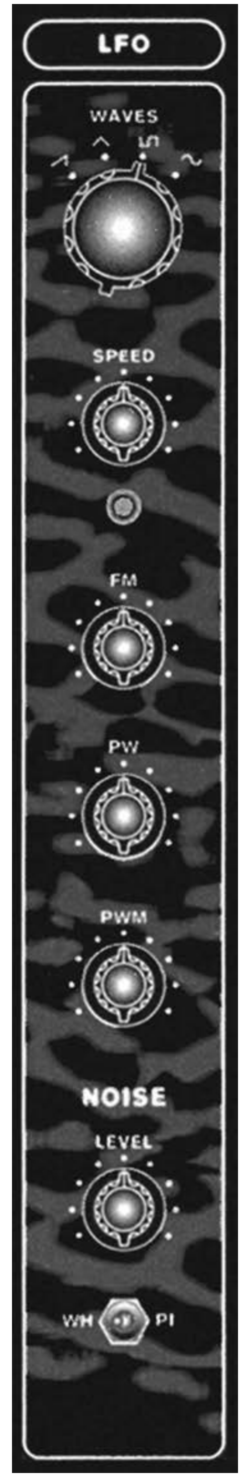
WHITE NOISE SOURCE

Apart from oscillators (and possibly a filter set to self-resonate) the white noise generator is the most common sound source found on analog synthesizers. Just as white light comprises all colours of light mixed together, white noise consists of sounds at all frequencies randomly mixed together. This random sound, which resembles a sea or wind noise, can also be heard from a radio set tuned between stations and is also generated as an unwanted artifact of many electronic designs. The simplest white noise generator circuits just use a "noisy" diode, amplifying its imperfections as much as possible to create a strong noise signal. But there are several other ways to create white noise, playing random numbers at suitable speeds from a digital sound generation circuit can work equally well.

There are also two less commonly found variations: pink noise, which offers equal energy level at all frequencies (found on the Delta Music Research modular system, for example, and used for testing acoustic systems); and red noise, which boosts the lower frequencies, and so sounds damped compared to white and pink noise (found on some of the EMS synths).

White noise is endlessly useful above and beyond the obvious creation of sea, wind, and explosion effects. All sorts of percussive and acoustic sounds contain an element of white noise and it often needs to be incorporated into more complex imitative analog sounds. For instance, the "chiff" on the start of a flute sound is best created with white noise passed through a short envelope and an imitation of human whistling benefits from the addition of white noise. All sorts of drum and percussive sounds need a white noise element, indeed, satisfactory short high-hat cymbal sounds can be created using nothing more. A synthesized snare drum sound (as heard on the Simmons and other electronic kits) typically comprises a tuned oscillator sound, to replicate the sound of the skin being struck, balanced against a burst of white noise to replicate the sound of the snare, which is a rattle made of metal wires. Check out the website tracks of the JHS Drum Synth. On the SE Boomstar the white noise section isn't obvious, but it's there, as one channel on the small built-in mixer. Small integrated monophonic systems like this will often find use for percussive effects, so those which leave out the white noise option are missing a trick. Vocoders also need a white noise input to help generate sibilant ("s" and "t") vocal sounds.

Since white noise in an electrical circuit comprises random voltages it can also be used as a source of random effects. But the random elements in white noise arrive much too rapidly to be of much use in musical terms, so it's necessary to pick and choose from them, as we see next.



The Technosaurus Selector LFO module included a White Noise output.

SAMPLE-AND-HOLD

The term "sample-and-hold" is usually associated with the creation of random sound effects, though this module can be used for much more. It is

The Technosaurus Selector sample-and-hold module.

not generally appreciated that most analog synthesizers already have one sample-and-hold circuit associated with the keyboard; when a new key is played, the keyboard sample-and-hold circuit detects and resets itself to the voltage produced, then holds that voltage as an output to the oscillators, setting them to the pitch of the new note until a different one is played. If the note of the oscillators drifts or "droops" it is often because the keyboard sample-and-hold circuit is faulty, sometimes because a capacitor (an electronic component that stores electrical charge) is slowly leaking away its charge.

But it is true that an independent sample-and-hold module is most often found in very close conjunction with a source of white noise, which is effectively a producer of rapidly varying random voltages. Some synthesizers do offer a white noise source but no sample-and-hold facility – the Sequential Pro One is an example – and on the Moog Prodigy there are, unsurprisingly, no random modulation options because there is no white noise source at all. On the Minimoog the white noise source can be patched directly to control the oscillator frequencies, but with no independent sample-and-hold circuit this just produces a rather jittery and unmusical variation in pitch (Moog later produced an optional sample-and-hold unit, the Model 1125 used by George Duke and others, which is actually quite versatile).

An independent sample-and-hold circuit, as its name suggests, typically looks at the random voltage coming from a white noise source, which is varying very rapidly, matches its output to that voltage and holds that output voltage steady for a variable amount of time. The speed at which the sample-and-hold circuit updates the output voltage is controlled either from an internal or an external LFO, or other source. When the output of the sample-and-hold is applied to an oscillator the result is a series of random pitches, when applied to a filter, the result is a series of random tones. This latter technique is very useful, particularly when a repeated note is playing, as the impression can be given of a much more varied series of sounds. There's a famous example in the middle of Emerson, Lake & Palmer's track *Karn Evil 9*. Examples on the website include tracks 27, 39, and 57.

On most integrated synthesizers that are equipped with a sample-and-hold circuit, a white noise source is permanently connected as its only input signal. But to do this is to lose a lot of the potential of the circuit. On the Roland SH3, ARP Odyssey, and just a few other synths, the sample-and-hold can also be fed from an LFO or another source. The resulting output depends very much on the wave shape of the LFO input; inputting a sine wave results in a series of output voltages that rise and fall repeatedly, generating regular patterns when applied to a filter, or a series of notes from an oscillator, though not necessarily in any regular musical scale. A sawtooth wave would create a repeated rising series of notes; an inverted sawtooth a repeated falling series of notes. This is sometimes referred to as a glissando and so the sample-and-hold circuit can become a rather less musically controllable substitute for some kinds of sequencer or arpeggiator.

On the ARP Odyssey (and the more recent Korg clone) the sample-and-hold circuit is particularly flexible and can become quite baffling. It can