CHAPTER 7

Principles of Analog Synthesis and Voltage Control

Brutal, caustic, volcanic. Evocative, flirting, caressing. Crisp, powerful, biting. Entrancing, embracing, exhilarating! Extend the stuff your music is made of with the Minimoog . . . The IN-strument of the Pros.
—Original Moog Minimoog brochure, 1971

Vladimir Ussachevsky and engineer Peter Mauzey in the Buchla studio of the Columbia–Princeton Electronic Music Center, circa 1970. (Columbia University Computer Music Center)

Electronic music is an art that marries technology and human imagination. This chapter provides a definitional background to the science behind audio phenomena and its application to the synthesis of musical sound. Understanding such fundamentals was essential to the early composers of electronic music whose equipment often had its origins in the audio engineering lab. Over time, even as the design of the instruments has become less technical and more comprehensible to the average person, the lexicon of electronic music terms and principles remains the same. Knowing the basics of waveforms, filters, cutoff frequencies, modulation, and other technical concepts is key to a thorough understanding of the making of electronic music and appreciation of the results.

In keeping with the generally chronological organization of the historical portion of this book, this chapter provides a grounding in the principles underlying the making of electronic music from the standpoint of analog synthesis and the application of these precepts to voltage-controlled synthesizers. As such, this material provides background in anticipation of discussions of both voltage-controlled analog synthesizers in Chapter 8 and their application in computer and digital synthesis as discussed in Chapters 10, 11, and 12.

UNDERSTANDING MUSICAL SOUND

The science of musical acoustics developed during the latter half of the nineteenth century in tandem with general discoveries in the field of electricity. The scientist Hermann von Helmholtz was a principal player in these discoveries and demonstrated that musical sound could be analyzed according to a few basic physical principles. Using combinations of tuning forks to illustrate his point, he showed that the quality (or timbre) of a tone was reliant on the intensity, order, and number of harmonics (overtones and partials) present in a note. Helmholtz showed that the vibrations found in a single musical tone consisted of a **fundamental** or base tone accompanied by related **harmonics** above the frequency of the fundamental. The harmonics of a tone are responsible for creating timbre or tone color. Timbre is what distinguishes the sound of a violin from the sound of a piano, even though both instruments might be playing the same note. Every instrument exhibits its own unique mixture of harmonics called its **harmonic spectrum**. Figure 7.1 visualizes the natural harmonic series of a tone.

When building sounds using electronic music techniques, the composer is working with the naturally occurring harmonic spectrum of predefined waveforms. Figures 7.2 and 7.3 depict a common method of illustrating the harmonic spectrum of waveforms, in this case a square and triangle wave. Figure 7.3 relates the harmonic spectrum inherent with each basic type of waveform to the musical scale.

The Helmholtz theory suggested that sound could be analyzed by its component parts and led directly to the engineering of electronic means for synthesizing sound, first in the form of Cahill's Telharmonium. An understanding of the wave structure of sound led to a robust reassessment of tonal systems used by composers. A technical understanding of consonance and **dissonance** stemmed from this scientific work. Helmholtz's theories also inspired a new, rational approach to analyzing sounds of all types, including noises. The Futurists categorized different types of sound for the purpose of using them in composition. Ferruccio Busoni saw in the scientific understanding of musical sound the possibility of inventing new instruments for extending the range of the 12-tone system. Busoni referred to Cahill's Telharmonium in this regard when he wrote in 1907:

> Keyboard instruments, in particular, have so thoroughly schooled our ears that we are no longer capable of hearing anything else—incapable of hearing except through this impure medium, yet Nature created an *infinite gradation—infinite!* . . . He [Cahill] has constructed a comprehensive apparatus which makes it possible to transform an electric current into a fixed and mathematically exact number of vibrations. As pitch depends on the number of vibrations, and the apparatus may be "set" on any number desired, the infinite gradation of the octave may be accomplished by merely moving a lever corresponding to the pointer of a quadrant.[1]

All of these people had set the scene many years before the arrival of composer John Cage. Cage brought an artistic clarity to the nature of creating music. He did this by professing to remove his emotions from the process of composing and objectively examining the materials of music. Cage sought ways to let sounds be themselves, allowing the listener to provide whatever emotional or intellectual context he or she needed to assess the result. In this regard, Cage directly echoed the sentiments of Busoni, who once declared that "Music was born free; and to win freedom is its destiny."[2] Cage's approach was not unlike that of a scientist studying a natural phenomenon. He observed, measured, and experimented to carry out musical hypotheses in the form of compositions.

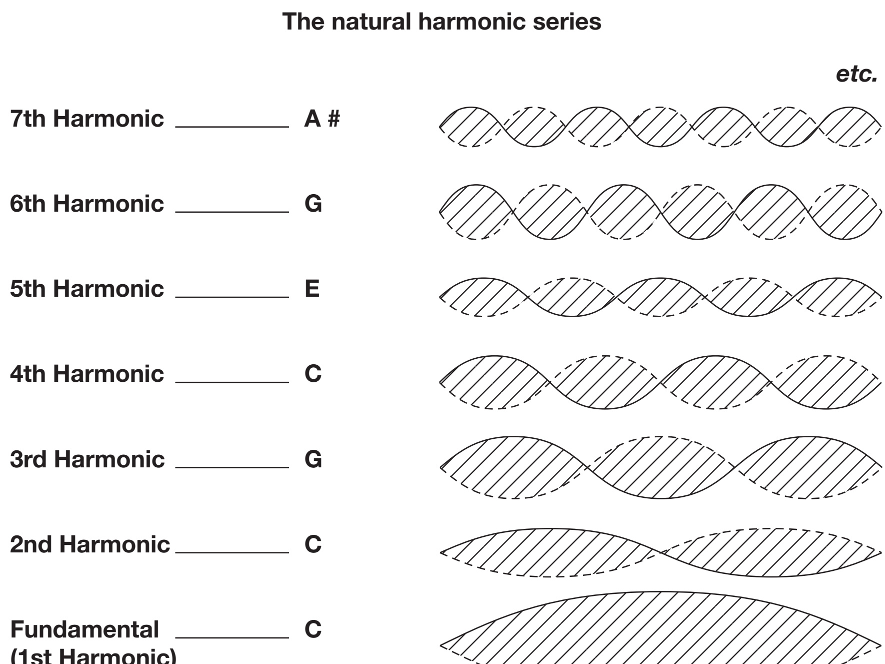
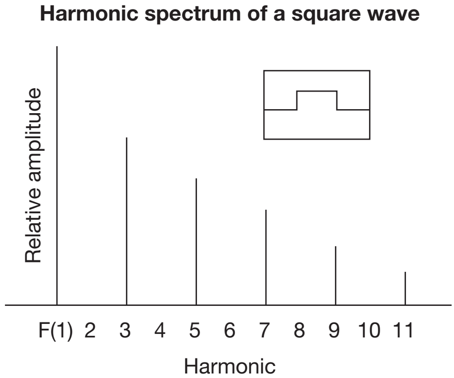


Figure 7.1 The harmonic content of a note comprises the dominant frequency known as the first harmonic, or fundamental. The first harmonic is the lowest frequency in the harmonic series of two or more frequencies that make up the content of a note. This diagram portrays the harmonic series for a note played by a string instrument. Electronic musical instruments can build notes using the addition and subtraction of harmonics to and from the fundamental tone. (after Friedman, 1986)



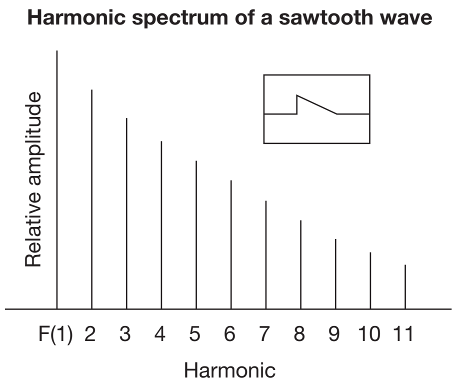


Figure 7.2 Harmonic spectra of square and sawtooth waveforms. (after Friedman, 1986)

Like Helmholtz, Cage was fascinated by the constituent parts that make up sound. In 1937, he gave a talk to an arts society in Seattle in which he suggested that music should be defined by its four basic components: the timbre ("overtone structure"), frequency, amplitude, and duration of sounds.[3] By 1957 he had added a fifth component to the list: the "morphology," or envelope, of the sound, otherwise known as its attack and decay characteristics, or "how the sound begins, goes on, and dies away."[4]

When Cage first proposed these ideas he also related them directly to the potential of using electronic musical devices to broaden our sound spectrum and create a new kind of music. The special nature of "electrical instruments" was that they provided total control over the principal components of sound. In perhaps his most prophetic statement, Cage said in 1937, "I believe that the use of noise to make music will continue and increase until we reach a music produced through the aid of electrical instruments which will make available for musical purposes any and all sounds that can be heard."[5]

Cage was by no means working in aesthetic isolation. He had the benefit of knowing and learning from several key figures in contemporary music, including Edgard Varèse, Henry Cowell, and Arnold Schoenberg. But in analyzing sound according to the five basic parameters—timbre, frequency, duration, amplitude, and envelope—Cage defined the

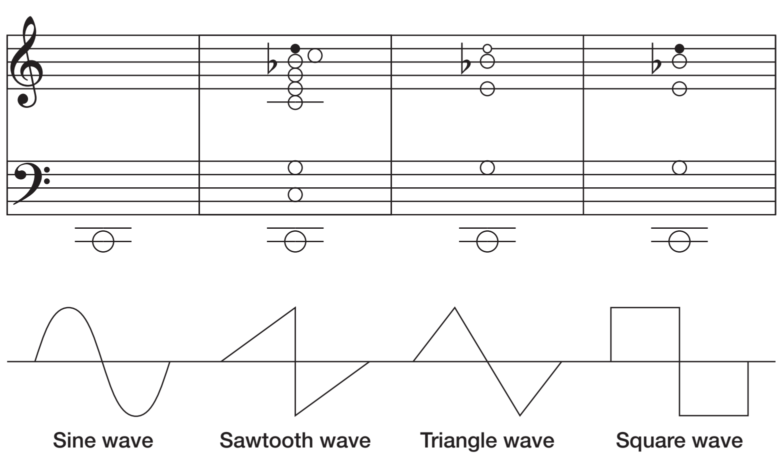


Figure 7.3 Harmonic spectra of sine, sawtooth, triangle, and square waves shown using musical notation. (after Strange, 1983)

common denominators by which all sound can be described. What set Cage apart was that he used these essentially scientific principles to rewrite the definition of music. Because all sounds are composed of the same primary components and because music is sound, then it must follow that all sounds can be defined as being musical.

The Components of Sound

Sound is produced by air pressure waves that cause the eardrum to vibrate. These vibrations are converted by auditory nerves into impulses that the brain recognizes as sounds. If a wave vibrates in a regular pattern, it is perceived as a pitched sound, such as those used in music. If the wave does not vibrate in a regular pattern, it is perceived as unpitched sound or noise.

Understanding the five components of sound is helpful for the appreciation of any music. They are especially pertinent to electronic music because the composer and musician are often working with direct control over these aspects of what you hear:

- **Frequency: the pitch of a sound**. Specifically, it is the number of vibrations per second that, when in the audible range, are detected as a certain pitch. In measuring frequency, a single vibration is called a **cycle** and the number of cycles can be expressed by a unit of measure known as the **hertz (Hz)**. In electronic music, this pitch becomes audible as an expression of the alternating electrical current that is used to vibrate the cone of a loudspeaker at a certain rate per second.
- **Amplitude: the loudness or volume of a sound and its constituent harmonics**. The simplest definition of amplitude is that it comprises the loudness of a sound and is conveyed through a loudspeaker by the distance that the speaker cone moves back and forth from its neutral position. Amplitude has multiple applications in the creation of electronic music. In addition to the overall volume of a given signal, one can selectively alter the amplitude of individual harmonics using controlled voltages, changing the timbre of a tone. In addition, amplitude may have its own shape or pattern that affects the envelope of a sound (see below).
- **Timbre: the nature or quality of a sound**. Sometimes known as tone color, timbre is what distinguishes the sounds of different musical instruments playing the same note. All sound waves are complex and contain more than just one simple frequency or fundamental tone. These additional wave structures are sometimes called *partials*, *overtones*, *harmonics*, and *transients*. If one harmonic, or fundamental, predominates, then the sound can be related to a note on the musical scale. A more complex set of harmonics—for example a sound in which the amplitudes of all harmonics have been made equal—makes it difficult to associate a tone with a specific note.
- **Duration: the length of time that a sound is audible**. Acoustic instruments have a limited ability to sustain sounds. The piano is designed with a pedal for the purpose of sustaining notes. Electronic instruments have the innate ability to sustain a sound indefinitely, making duration a key element in composition. The overall duration of a note can be further broken down into its envelope characteristics (see below).

- **Envelope: the attack, sustain, and decay characteristics of a sound**. The envelope of a sound is essentially the shape of the amplitude characteristics of a sound as it occurs over time—the way it begins, sustains, and ends. **Attack** refers to the beginning of a sound and how long it takes to reach its maximum loudness. **Sustain** is the length of time that a sound lasts at a fixed amplitude. **Decay** is the time it takes for the signal to go from its peak amplitude to its sustain amplitude. **Release** comprises the time it takes for a note to end and return to zero amplitude, for example after the finger is lifted from the key.

Fourier Analysis and Waveform Mathematics

The French mathematician and physicist Jean Baptiste Fourier (1768–1830) developed a theory of wave physics during the early nineteenth century that allowed for the scientific analysis of musical sound. In relation to the frequency relationships of periodic waveforms, the theory states that *any periodic vibration (waveform), however complex, is comprised or can be created by combining a series of simple vibrations whose frequencies are harmonically related.*

Fourier theory has two direct applications in electronic music. First, a sound wave is made of component parts and, by analyzing its characteristics (e.g. frequency, amplitude), one can measure and control such components to modify the sound. This is called **Fourier analysis**, the process of dividing a waveform into its constituent frequencies. Second, waveforms can be created with predictable and controllable results by combining simpler waves (e.g. sine waves) into more complex waves. This is a method of synthesis based on Fourier principles and is called *Fourier* or *additive synthesis*. In the case of harmonic sounds, the sidebands or harmonics all consist of integer multiples of the lowest or fundamental frequency. Non-harmonic musical sounds—such as that of a gong or bell—can be created by combining waveforms that are not integer multiples of one another. Figure 7.4 shows the results of combining several simple waveforms.

The frequency range of the 12-tone scale may contain higher and lower octaves. An **octave** is created by doubling or dividing in half the frequency of the first harmonic (fundamental) of a tone. Figure 7.5 provides a guide to the frequencies of fundamental tones in a standard set of octaves.

Making Music

It is evident from earlier chapters that there has been an evolution in the field of electronic music from the use of simpler, non-parametric instruments such as the Theremin, electronic organ, or even the tape recorder to instrumentation that provides the composer with increasingly programmatic control over the elemental components of musical sound. Allen Strange (1943–2008), in his classic text about the techniques of analog synthesis, pointed out that electronic musicians faced the same challenges as those learning conventional instruments: musical events involve practice in the making of a sound as well as the control or performance of the sound.[6] Table 7.1 provides a reminder of the many complexities of musical sound that must be managed in an electronic music environment and at the same time indicates how they are interrelated. For example, loudness is affected by both the filtering of the audio spectrum and amplitude.

Controlling all of the parameters available for generating and performing music on analog synthesizers was made practical by the introduction of the technique of voltage control.

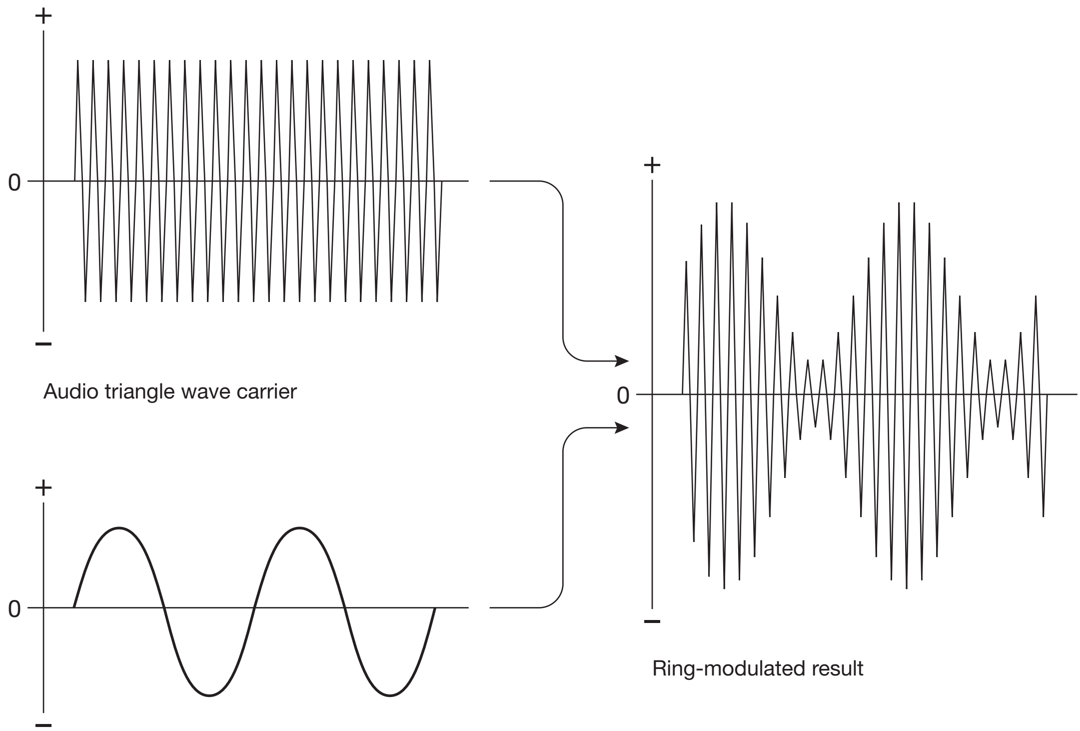


Figure 7.4 Ring modulation serves as a good illustration of the effects on waveshape when two different waveforms are combined. In this case a triangle wave is modulated by a sub-audio sine wave, resulting in a waveform that combines and subtracts elements of both source signals. (after Naumann, 1985)

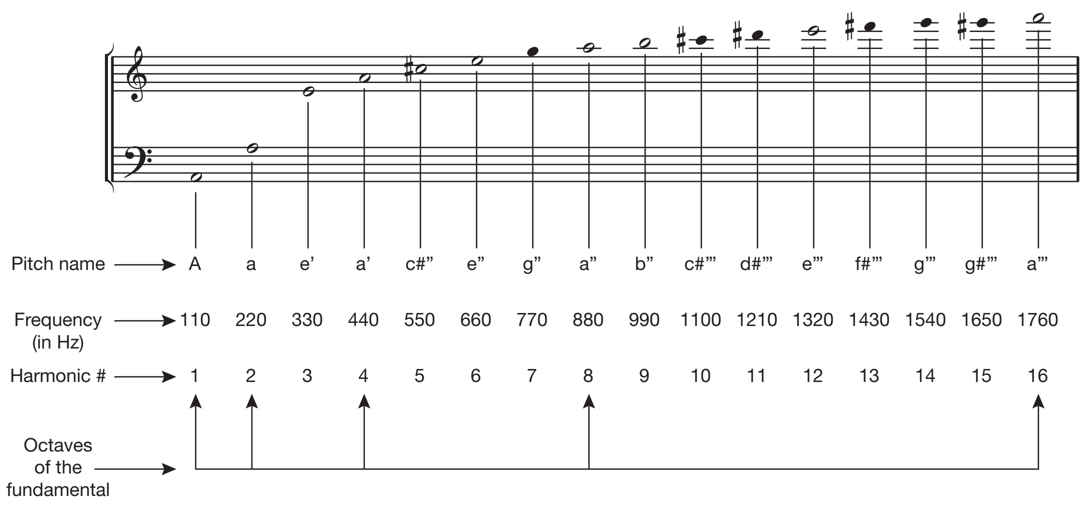


Figure 7.5 Frequencies expressed in Hz and related to the musical scale. (after Naumann, 1985)

Table 7.1 **Electronic music parameters**[7]

Frequency	*Audio spectrum*	*Amplitude*	*Structure*
Discrete pitch	Timbre	Loudness	Rhythm
Sliding pitches (portamento and glissando)	Loudness	Rhythm	Duration
Vibrato	Vibrato	Tremolo	Repetition
Timbre	Tremolo		Sequence
Associated techniques			
Frequency modulation	Band-pass filtering	Amplitude modulation	Looping
	Ring modulation	Delay	Sequencing
	Reverberation		Envelope generation
	Pulse width modulation		

ELECTRONIC SOUND GENERATION

Waveforms can be generated by an electronic circuit called an **oscillator**, which produces periodic vibrations in the form of an electric current. The resulting current precisely mirrors the shape of the waveform in a natural acoustic environment and is only audible once it reaches a loudspeaker. Oscillators can produce sounds in the full range of human hearing—from about 20 Hz to 20,000 Hz. They may also produce subsonic and ultrasonic waves, which cannot be heard but which, when combined with other waves, produce an audible result in keeping with Fourier principles of waveform behavior. Oscillators are the basic building blocks of sound in a synthesizer.

Oscillators have been made using many different techniques throughout the history of electronic music. In the late nineteenth century, Thaddeus Cahill invented the tone wheel—an electro-mechanical device that required the rotation of precisely milled notched metal cogs against a metal brush to produce pitch-making circuits. Vacuum tubes were used as oscillators in many early electronic musical instruments until the advent of the transistor in the 1950s. Solid-state oscillator circuits were found in voltage-controlled analog synthesizers from the 1960s to the mid-1980s, when digital synthesis using integrated circuits and software was adopted.

Waveforms

Common terminology is used in describing the characteristics of waveforms. Figure 7.6 provides a graphical representation of these aspects of a waveform.

The midpoint of a wave's propagation is called the **equilibrium point** and is denoted as point 0 on a waveform diagram. A **period** is the length of time required for a wave to complete one cycle from the equilibrium point to its apex, back through the equilibrium point to its base point and back again to the equilibrium point. The distance from the apex to the base of a waveform is called the **displacement**, another designation for

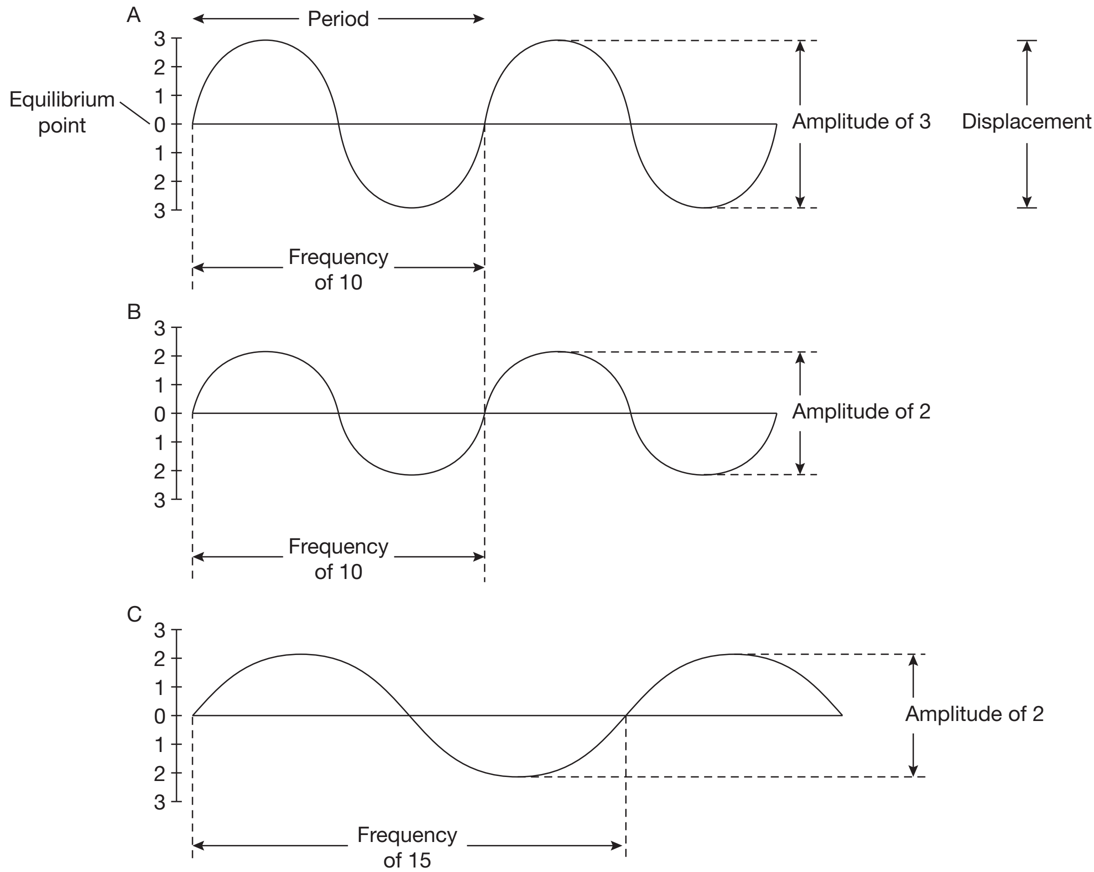


Figure 7.6 Elements of a waveform. (after Strange, 1983)

wave amplitude. The **duty cycle** of a wave is a ratio denoting the proportion of a single cycle that occurs above the equilibrium point versus time below the equilibrium point.

Waveforms can also be said to occupy a space in time, also known as the **phase**. Waveforms are said to be *in phase* if they are identical and occupy the same space and time in the conducting medium. If two identical waves are displaced slightly in the same conducting medium, one beginning before the other, they are said to be *out of phase*. This phenomenon produces audibly perceptible results and has been used as a recording technique by variably phasing two identical recorded tracks of any sound source, producing a gradually shifting spatial displacement of the sound (see Figure 7.7).

There are four basic waveforms used in electronic music composition. All of them may exist in any frequency range:

- **Sine wave**. This is the simplest type of waveform. It contains no harmonics. The sine wave undulates evenly. Although some liken the sound of a sine wave to that of a flute, even the flute has more body and depth than a pure sine tone, since it contains harmonics. The audible sine wave is a thin, precise tone, similar to a whistle. Multiple sine waves are often used as the building blocks of more complex tones.

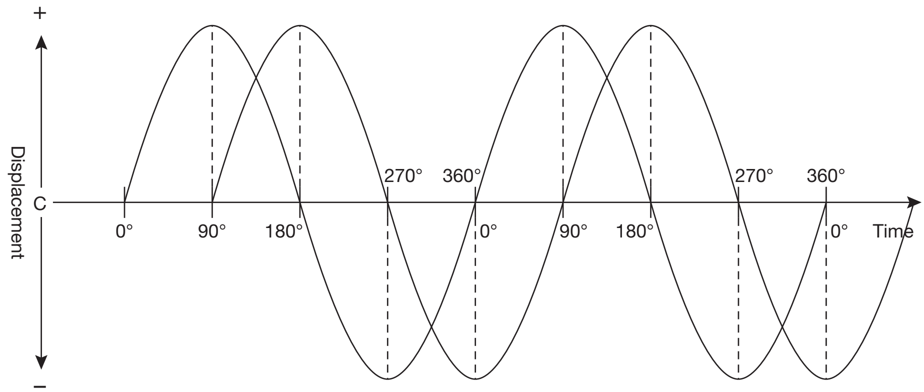


Figure 7.7 Phase relationships of two sine waves. (after Naumann, 1985)

- **Sawtooth wave**. The sawtooth or ramp wave contains all even and odd harmonics associated with a fundamental tone, making it a rich source for modeling other sounds. The amplitude of each overtone decreases exponentially as a ratio of the harmonic's frequency to that of the fundamental, providing a ramp shape to the wave. The sound of the sawtooth is rich and buzzy and is often used to reproduce the sound of reeds or bowed string instruments.
- **Triangle wave**. A triangle wave contains only the fundamental frequency and all of its odd-numbered harmonics. The amplitudes of the harmonics fall off in odd-integer ratios. The sound of the triangle wave has more body and depth than a sine wave, somewhat like a muted horn.
- **Pulse wave**. The pulse or *rectangular* wave has only the odd harmonics of the fundamental, like the triangle wave, but differs significantly in the amplitude relationships of these harmonics. Unlike sine, sawtooth, and triangle waves, which make a transition from the apex to the base of the wave cycle, the pulse wave instantaneously jumps from the apex to the base. Duty cycles of pulse waves can vary, with 1:3 being typical (1:3 indicates that the cycle spends one third above 0 and two thirds below 0 per cycle). The harmonic content of the pulse wave is determined by the duty cycle. A **square wave** is a type of pulse wave whose duty cycle is one half of the total cycle of the waveform, or 1:2, evenly divided between the upper and lower reaches of the wave, hence its square shape. The harmonic content of a pulse wave can be changed dramatically merely by altering its duty cycle. Pulse waves have a clear, resonant sound.

Each of these basic waveforms has a reliable structure that exhibits strict amplitude relationships between the harmonics and their fundamental. They can also be combined to create richer, more textured sounds or used to modulate the amplitude or frequency of another sound—techniques that will be explored below.

One more basic waveform needs to be mentioned. It is called **white noise**, and it does not exhibit the structural symmetry of sine, triangle, sawtooth, or pulse waves. In the simplest sense, white noise is to those four basic waveforms what the color gray is to the primary colors: it is a combination of all of them, with no particular element dominating the mix. White noise results when all the frequency and amplitude characteristics of a sound occur at random within the audio spectrum or contain energy at all

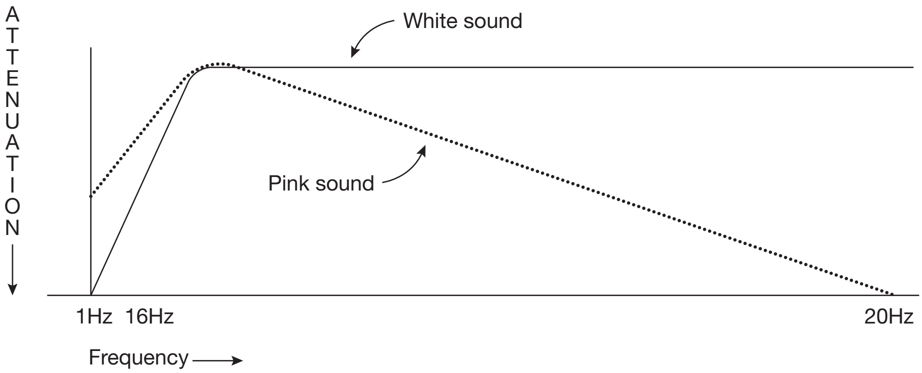


Figure 7.8 Graphic representation of white and pink noise frequency spectra. (after Strange, 1983)

frequencies within the audio spectrum. It is a continuous dense hiss. It can be filtered, modulated, and otherwise refined to sound like such things as the wind or the ocean, and is a rich source of background sound and texture for the composer of electronic music. Even The Beatles found an effective use for modulated white noise in their use of Moog-created undulating, wind-like noise at the end of "I Want You (She's So Heavy)" (1969). Composer Allen Strange defined white noise more precisely as containing all audible frequencies between 18 Hz and 22,000 Hz. A distilled form of white noise is called **pink noise**, which Strange defined as containing all frequencies between 18 Hz and 10,000 Hz (see Figure 7.8). At the other end of the audio spectrum, noise restricted to the frequency ranges between 10,000 Hz and 22,000 Hz would be **blue noise**.[8]

VOLTAGE CONTROL FUNDAMENTALS

As introduced in the previous chapter, **voltage control** is a method of applying metered amounts of current to an electronic component to govern how it operates. Using control voltages to manage an instrument became practical during the 1960s with the availability of solid-state circuitry and the ability to direct a small amount of current to the modular components of a synthesizer. Voltage-controlled technology was responsible for the commercial boom of electronic musical instruments during the 1960s and 1970s, leading to the adoption of control principles that continue to be applied, without the need for the control voltages themselves, in the algorithms used to drive digital synthesizers and software synthesizers.

A *control* voltage is discrete from the voltage used to generate an audio *signal*. Whereas the signal is the sound itself—a voltage in the audible spectrum—the control voltage affects the structure or flow of the sound and may itself be inaudible except in how it affects the audible signal. In the first modular synthesizers, patch cords were used to connect the output of one component to the input of another. Because of this, some components such as oscillators could be used as either signal sources or control sources, whichever suited the needs of the composer. Later performance instruments eliminated the patch cords and provided preset connections for governing signal and control voltages.

Voltage-Controlled Components

A significant advantage of voltage-controlled components was that special circuits could be designed to simultaneously manipulate a multitude of settings that might otherwise have been impractical to manage by hand. For example, it would be impossible to control by hand—manually turning individual dials and sliding levers—the frequencies of several oscillators, their changing amplitudes, envelopes, and filtering all at one time. Several basic types of voltage-controlled modules have been designed to automate this process.

The following voltage-controlled components are commonly used in analog synthesis. These were available as individual components (e.g. envelope generator) or packaged into a modular synthesizer with ports for connecting and combining individual components. Performing with these modules is accomplished through the use of various manual and programmable controllers (see pp. 237–9).

- **Voltage-controlled oscillator (VCO)**. A circuit for generating a periodic waveform, usually a sine, sawtooth, triangle, or pulse/square wave. Some oscillators had settings for more than one type of waveform. The VCO was the basic sound-generating source of the analog synthesizer. Typical voltage-controlled inputs would allow manipulation of oscillator frequency and waveshape.
- **Voltage-controlled filter (VCF)**. A circuit using control voltages to set the parameters filtering the audio spectrum of the sound source. A simple VCF employing a low-pass filter (allowing only lower frequencies to pass through) might only have simple settings for the cutoff frequency and resonance, with a voltage-controlled input for changing cutoff frequency. Other types of filters, such as high-pass, band-pass, and band-reject, provide other means of controlling specific ranges of the audio spectrum (see "Frequency Filtering," pp. 242–3).

- **Voltage-controlled amplifier (VCA)**. A voltage-controlled amplifier allows the musician to control the volume of a signal over a variable scale of amplitude. Amplitude is a fundamental element of sound production and rarely occurs on a scale that jumps from 0 (off) to peak (on) without some steps in between. These steps may be slow, as in a gradual swell of volume, to rapid and periodic as in vibrato. The VCA provides settings for making such gradual changes in volume possible.
- **Envelope generator (ENV)**. The voltage-controlled envelope generator is a special purpose amplitude controller dedicated to shaping the four stages of a sounds evolution: attack, decay, sustain, and release. It is most commonly associated with the characteristics of notes played using a keyboard trigger. The voltages generated by an ENV correspond to each of the multiple stages of a note's envelope.
- **Low-frequency oscillator (LFO)**. This oscillator circuit is restricted to subsonic frequencies and is an important source of modulation for other voltage-controlled modules. It is not used as an audible signal but as a control signal for other components. If fed to the input of a VCO, the LFO can control minute or radical fluctuations in the frequency of the oscillator's signal. If fed to the input of a VCA, the LFO creates periodic changes in the volume of the signal. An LFO signal fed to a VCF will modulate the filter by changing its cutoff frequency in a fluctuating pattern. If fed to a voltage-controlled pulse wave oscillator, an LFO can modulate its duty cycle and provide a pattern of changing harmonics in its output.

Sources of Control Voltage

The voltage-controlled modules described above could be managed by the composer through several means. One of the most flexible, and sometimes confusing, aspects of voltage-controlled systems is that voltage signals can be used for many different functions, often concurrently. For example, a voltage-controlled oscillator could be adjusted manually using a rotary dial to change its pitch or it could be triggered by a voltage source outside of the oscillator itself, such as a keyboard or sequencer. The same can be said for other voltage-controlled modules for generating or modifying the sound.

Sources of voltage control fall into two categories: *manually operated (kinesthetic) controls* or *programmable controllers*.

Manual Controls

Manually operated (kinesthetic) controls are those that are adjusted or played by hand in real time.

Keyboards

The organ-style keyboard was the most common voltage controller found on analog synthesizers and had obvious advantages for playing music. Every key was a voltage generator and could be used to trigger a specific note by sending a signal to the voltage-controlled oscillator. The earliest analog synthesizers were **monophonic**, capable of outputting only one voltage at a time, conventionally the lowest key to be depressed at any given moment. **Polyphonic** keyboards were capable of playing more than one note at a time but were often limited to no more than ten voices—one per finger—in the earliest models. The octave range of a keyboard could be scaled up or down in frequency range and keyboards on the most advanced analog synthesizers could also be split so that different parts of the keyboard were assigned to different instrumental voices. Some manufacturers provided keyboards that could modify the scale—frequency steps between notes—making composition possible with microtonal and other alternatives to the 12-tone scale. Because a synthesizer keyboard was essentially no more than a source of voltage output, it could also be used for managing modules other than pitch generators, providing timing triggers of preset parameters to VCAs, VCFs, and other components, permitting many actions to occur simultaneously.

The first commercially available synthesizer keyboards were not **touch-sensitive**, but by the 1970s this had become a common feature. There were two aspects of touch sensitivity important to voltage-controlled keyboards. A **velocity-sensitive keyboard** generated a voltage for a note that was proportional to the speed with which the keys were depressed. A **force-sensitive keyboard** produced a control voltage proportional to the amount of pressure put on a key. Both types of keyboard sensitivity could be included in the same keyboard.

Most keyboards also had expression controls such as wheels or levers for providing pitchbend or modulation:

- **Pitchbender wheel**. The pitchbender wheel allowed the performer to slide a note up or down, gliding the frequency smoothly between pitches. The control did this by sending a higher voltage to the VCO to raise the pitch or sending a lower voltage to lower the pitch. The range of the pitchbender could be adjusted either through preset switches or a sliding control. In the most flexible systems, pitches could be bent from a range as small as two adjacent keys to several octaves.
- **Modulation wheel**. The modulation wheel adjusted the amount of voltage from an LFO used to modify a VCO, VCA, or VCF. The audible result on the waveform depended on which voltage-controlled module was being modulated. If the VCA was modulated, tremolo was produced. If the VCO was modulated, vibrato was the result. If the output of the modulation wheel was sent to the VCF, a filter sweep was the result.

In addition to keyboards, several other unique methods of kinesthetic controls were developed as voltage sources for synthesizers. The Moog **ribbon controller** was a monophonic device for the linear control of voltage and essentially served the same function as the keyboard but without the keys. It was used by sliding a finger up and down a slender metallic ribbon to cause changes in pitch. Wavering the fingertip along the surface could create vibrato. This was a popular control technique that was modified and adapted by several manufacturers.

Joysticks

Joysticks were adapted for use on some performance synthesizers and combined both pitchbend and modulation voltage sources. Moving the joystick from front to back controlled one voltage source while moving it from right to left adjusted the second. Having one control for two manually adjusted parameters made the control of these voltage sources much easier for the performing musician. Theoretically, the joystick could be used to send voltages to any two voltage-controlled modules. Typically, the joystick was connected to an oscillator source to control pitchbend when it was moved in one direction (e.g. left to right) and was connected to a VCA to control amplitude when moved in the other direction (e.g. front to back). An infinite number of positions were available between the absolute front-to-back and right-to-left planes, providing many subtle combinations of the two control sources.

Other Kinesthetic Inputs

Buchla pioneered several early alternatives to organ-style keyboards including the Kinesthetic Input Port, which used flat, membrane contacts arrayed in the configuration of an organ-style keyboard. Unlike conventional keyboards, the Kinesthetic Input Port was equipped with outputs for connecting the membrane "keys" directly to other voltage-controlled inputs, allowing the port to act as both a performance interface and a simple, programmable aid for triggering other functions on the synthesizer. A simplified version of the membrane keyboard was used on the portable Buchla Electronic Music Box (1973), a self-contained synthesizer suitable for live performance.

Programmable Controllers

Another aspect of playing a synthesizer is the ability to program sounds, patterns, and modulations so that they can be performed automatically or possibly stored for retrieval and playback later. This element is widely accepted today in the design of computer-based instruments and music software. Prior to the application of computers to music, the programming of synthesizers was not as easily done, yet many innovative solutions were devised for applying voltage control to automate important aspects of creating electronic music.

Sequencers

The RCA Mark II Electronic Music Synthesizer and its coded paper input device was an early attempt to provide control over pitch, amplitude, timbre, and the organization of musical tones. Raymond Scott reportedly accomplished something similar in his home studio, as did the engineers of the Siemens Studio für Elektronische Musik in Munich around the same time. In all of these cases, the instruments were hardwired to the sound-generating and modifying circuits, greatly limiting their adaptability to all but certain preset values determined by the circuit builder. Nonetheless, all three attempts underscored the value of programmability to electronic music—one of its inherent traits. What RCA, Scott, and Siemens had done was demonstrate the potential usefulness of a control module or **sequencer**.

Buchla and Moog independently developed voltage-controlled sequencers for their synthesizers. The sequencer provided a means for structuring a sequence of voltage control signals that were then fed as control signals to other voltage-controlled modules. A number of schema were provided, from straightforward voltage pulses to controllers that also provided time settings for varying the duration of a given increment in a sequence. Most sequencers could be set to trigger control voltages in 8, 12, or 16 increments and there were often three such arrays available at a time (Figure 7.11 shows a 16-track set-up). Despite a limitation of 8, 12, or 16 steps, patches could be used to effectively string out all three rows into single long sequence comprising three times as many steps.

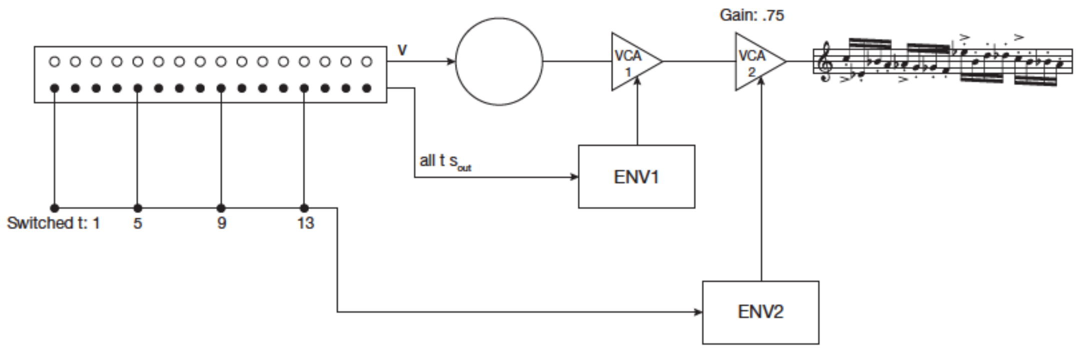


Figure 7.11 Setup for a 16-track voltage-controlled sequencer using a signal source, envelope generators (ENVs), and voltage-controlled amplifiers (VCA) to produce a sequence of accented notes. (after Strange, 1983)

LISTEN

ANALOG SYNTHESIS AND SOUND MODIFICATION

1 *Cartridge Music* (1960) by John Cage
A work for amplified small sounds that used phono cartridges as contact microphones

2 *The Wolfman* (1964) by Robert Ashley
Acoustic feedback was used as the primary source of audio material for this work

3 *Safari: Eine kleine Klangfarbenmelodie* (1964) by Hugh Le Caine
Used extensive additive synthesis and texturing by means of the Sonde, an instrument equipped with 200 closely tuned sine tones

4 *It's Going to Rain* (1965) by Steve Reich
Tape piece experiment with tape loops and phasing of vocal passages

5 *Pendulum Music* (1968) by Steve Reich
Used acoustic feedback

6 *Cambrian Sea* (1968) by Peter Klausmeyer
Extensive use of modulated white noise and a Moog voltage-controlled envelope/amplitude generator

7 *Ambience* (1968) by Richard Allan Robinson
Transformed electroacoustic sounds using voltage-controlled ring modulation, filters, and additive synthesis

8 "I Want You (She's So Heavy)" (1969) by The Beatles
John Lennon added a modulated sequence of Moog-generated white noise to the last part of the song, providing a sound like that of relentlessly blowing wind

9 *Toneburst* (1975) by David Tudor
Used feedback circuits

10 *Repeat* (1999) by Toshimaru Nakamura
Used feedback circuits via the composer's "no-input mixing board"

Sequencers were a versatile source of output voltage and could be combined in banks so that the output of one could start and stop another. Sequencer outputs could be fed to any other voltage-controlled component for generating, modifying, mixing, and distributing sound.

Sequencers were typically triggered by a **timing pulse** output by a manual controller, such as a keyboard. This enabled a performer to trigger a sequence of control signals by only touching a single key. Pressing a key could trigger any variety of control sequences, from automatically playing an arpeggio, triggering a rhythm pattern in another module, changing the envelope of the sound, or activating a filter sweep. Any module that was voltage-controlled could be triggered by a sequencer.

Timing pulse generators were LFOs dedicated to generating pulses for controlling tempo or other repetitive processes commonly used in music. As a control signal, the timing pulse consisted only of a binary On/Off signal.

Sequencers were programmed either manually using a panel of rotary dials or by playing a sequence of voltages on the keyboard. When using the keyboard, a sequence could be recorded in *real time* as it was played or one note at a time using *step programming*. In either case, the sequencer acted somewhat like a player piano roll, keeping a record of the key depressions but not recording the sound of the notes themselves. This allowed a sequenced pattern to be used with any patch, regardless of the instrumental voices chosen. The tempo and key could each be changed without affecting the other.

Sequencers were forgiving when it came to recording key strokes. If the keyed notes were not precisely in correct time, a feature called **quantizing** was used to align each note to the nearest beat in a preset tempo, locking all key strokes into a perfect tempo. **Looping** was another feature that allowed a sequence to repeat as long as desired, providing a steady rhythmic backdrop for a piece of music. It was also possible to link multiple sequencers so that one could trigger the others, providing a cascading series of programmed sequences with nearly limitless possibilities. In addition to providing the fixed sequential output of a signal sequence, some sequencers could also be set to output a given sequence in *random* order.

SIGNAL PROCESSING

If audio signals may be considered the raw material of electronic music, signal processing represents the ways in which these signals can be dynamically modified and shaped. Signal processing is primarily aimed at modifying the frequency, amplitude, and timbre of sound. This is done through the use of a variety of circuits to modify the electrical voltage of a sound, or its digital equivalent in computer-based instruments.

The electrical signals generated by an audio circuit are not strong enough to drive a loudspeaker on their own and require amplification. **Gain** is the amount of voltage or power that an amplifier provides to increase the strength of a signal. The audible effect of increasing gain is a corresponding increase in volume from a loudspeaker. Within the circuits, however, gain is a factor in modifying other aspects of a waveform because it affects the amplitude of the signal to be modified.

Frequency Filtering

A **filter** is a specialized amplifier that controls the amount of gain to prescribed frequency ranges of a sound. Making such adjustments changes the balance of harmonics found in the source sound signal. Adjusting the perceptibility of harmonics is key to modifying the identity or timbre of a sound, making filters one of the most important sound modification components available to the composer.

Stereo systems are often equipped with a rudimentary filter called an **equalizer** for adjusting the amount of bass, midrange, and treble frequencies that will be heard in a piece of recorded music. Filters associated with electronic music can generally be adjusted to finer settings than those on a conventional stereo system. Some kinds of filters are designed for passing only certain ranges of frequencies and provide precise settings that can be easily repeated whenever needed. A **cutoff frequency** is the point at which a filter begins to omit a prescribed frequency range. Theoretically, a filter should attenuate or cut off a range of frequencies at the prescribed point, but this is not the case. Passing of the frequencies occurs as a **roll-off slope** that is generally equivalent to about 3 dB attenuation per octave. The precise roll-off specifications for a filter depends on its circuit design and will vary from manufacturer to manufacturer.

Some typical types of filters include the following (see also Figure 7.12):

- **Band-pass filter**. Allows only those sounds *between* specified high- and low-frequency cutoff points to be heard. It removes the high and low frequencies from a signal at the same time.
- **Band-reject filter**. Allows only those sounds *above* or *below* specified high- and low-frequency cutoff points to be heard. It removes the midrange frequencies from a signal.

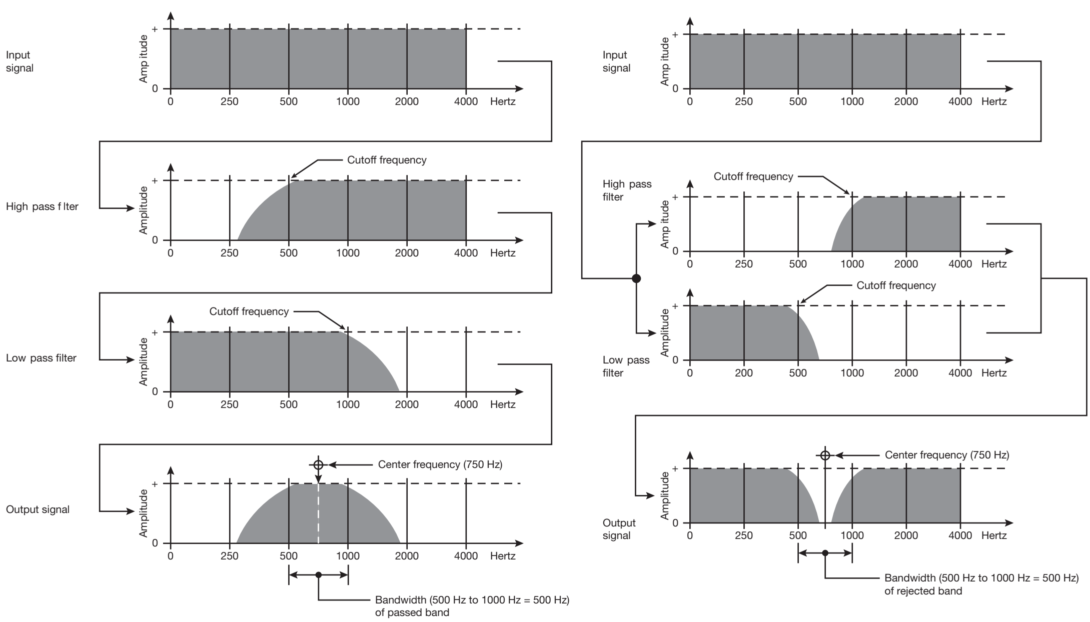


Figure 7.12 Band-pass filter (left) and band-reject filter (right). (after Naumann, 1985)

- **Low-pass filter**. Allows only frequencies *below* a specified cutoff point to be passed. It removes the high frequencies from a signal.
- **High-pass filter**. Allows only frequencies *above* a specified cutoff point to be passed. It removes the low frequencies from a signal.

While the degree of attenuation of a filter was sometimes a permanent fixture of its circuit design, most low-pass and band-pass filters also had a variable *regeneration* or *resonance* control sometimes referred to as the **Q factor**. This control changed the perceptible sharpness of the filtered sound. The Q factor was determined by dividing the center frequency of the filtered band by the bandwidth. For example, if the center frequency of a filtered band was 150 Hz and the bandwidth was 75 Hz, the Q factor was 2 (150 Hz/75 Hz = 2). Increasing the Q factor narrowed the width of the passed band, increasing the Q factor and further accentuating the remaining sidebands, giving the sound a hollow, harmonic chiming quality. Another technique was to keep the Q factor constant while varying the center frequency, resulting in a change to the bandwidth of the passed band while it maintained the same Q factor relationship with the center frequency.

Filters may be part of a synthesizer console, a software component for processing sounds, or a standalone device used like an effects box between an instrument and the mixing board or loudspeaker system.

Envelope Shaping

The envelope of a sound is the way the sound begins, continues, and then ends. It is the pattern of loudness of a sound. For example, a note played on the piano will begin sharply (attack) and will also end abruptly (release), but the middle part of the note can be extended by pressing the pedal (delay and sustain). Electronic musical instruments offer unique control over the envelope characteristics of a sound. This technique can be used to change the attack characteristics of all discretely generated sounds. Envelopes may be adjusted manually or programmed using an envelope generator.

Most envelope generators have four settings for different stages of a sound:

- **Attack**. The start of a sound as defined by the time it takes for the signal to go from zero amplitude to peak amplitude.
- **Decay**. The second stage of a sound as defined by the time it takes for the signal to go from its peak amplitude to its sustain amplitude.
- **Sustain**. Once a sound has passed through the attack and decay stages, it may be sustained at a fixed amplitude for as long as the note is held.
- **Release**. The end of a note's envelope, which drops off rapidly to zero amplitude. The term "release" is equivalent to releasing the key on a synthesizer.

These four stages of envelope generation are collectively known as the **ADSR** (attack, decay, sustain, release) characteristics (see Figures 7.13 and 7.14). Settings for the attack, decay, and release properties of a signal govern the duration of a sound regardless of how long a key is depressed. The sustain setting denotes a peak amplitude for as long as a signal is held.

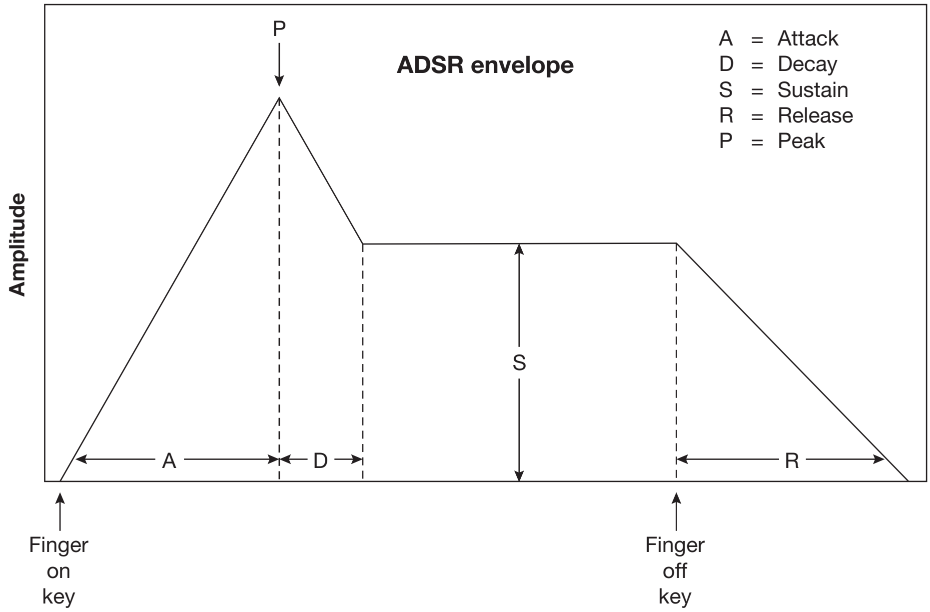


Figure 7.13 Envelope characteristics of a sound that are controllable by a synthesizer, including attack, decay, sustain, and release (ADSR). (after Friedman, 1986)

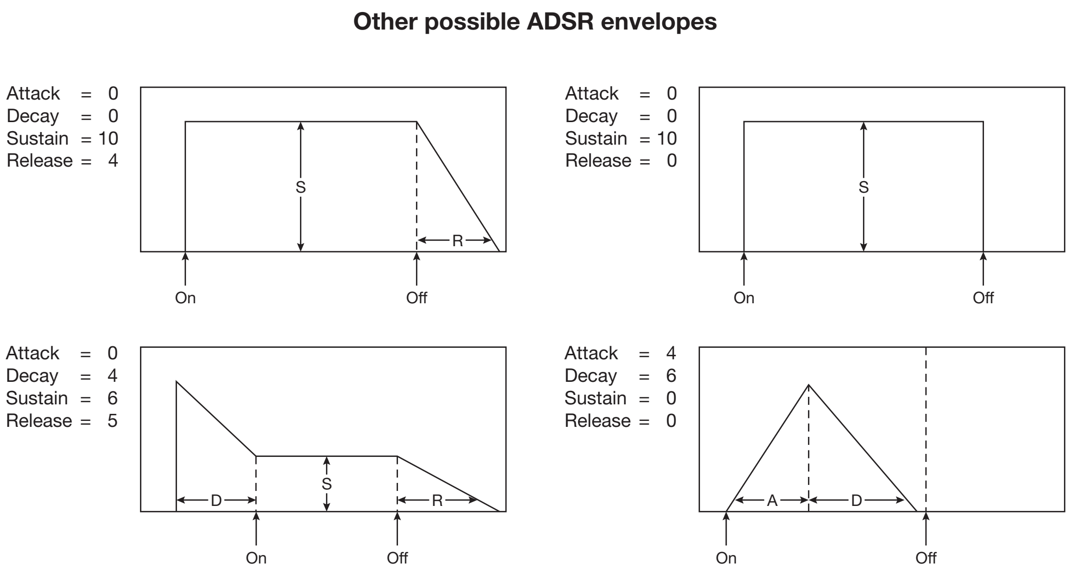


Figure 7.14 Sample ADSR settings for shaping sounds. (after Friedman, 1986)

Envelopes can be changed for any given sound signal. The attack, sustain, and decay characteristics are individually adjustable, providing the composer with infinite possibilities for altering a given sound source. In the voltage-controlled synthesizer, the envelope generator can be triggered by control voltages from other components, such as a low-frequency oscillator.

Echo, Reverberation, Looping, and Delay

The techniques of echo, reverberation, looping, and delay originated with tape composition and are described in that context in Chapter 5. When described using signal processing terminology, the definitions of these techniques can be applied apart from the tape recorder to both analog and digital signal processing.

Echo and *reverberation* comprise different degrees of the same phenomenon—the effect of reflected sound on the perceived depth or character of an audio signal. Reverberation comprises the sum total of all such reflections as expressed by a prolongation of the sound, where individual reflections are not discretely perceivable. The length of the reverberation is determined by the distance of the listener from the sound source, and the type of surrounding reflective surfaces. The length of reverberation is measured from the start of the sound to the point when it decays to 60 dB below its original amplitude. Echo is a form of reverberation in which the individual sound reflections, rather than being compressed into a short lapse of time, are spaced by 50 milliseconds or more, at which point they can be perceived individually.[13] Artificial reverberation and echo can be produced using a tape recorder or circuits designed to provide adjustable settings for room size and reflectivity.

The term *delay* is borrowed from the tape composition practice of stringing a length of recording tape through two tape recorders, recording a sound on the first machine, playing it back on the second, and then simultaneously feeding the signal back into the first machine where it is recorded again. The signal that is repeatedly re-recorded eventually diminishes with each generation of re-recording. Tape delay has been replaced with analog and digital delay circuits that reproduce the same effect with controllable parameters for the pace, duration, and rate of disintegration, if any, of the delay signal.

Looping a sound is similar to the use of a delay system except that the original signal is not re-recorded with each pass. Rather, a loop repeats without any loss of fidelity for as long as it is played. The concept of looping originated with locked grooves in turntable discs and was translated to the tape medium by splicing a short length of tape end to end so that it would play repeatedly.

Signal Modulation

The term **modulation** is used in music to denote a change from one key, or tonal center, to another—a technique that is commonly heard in the performance of popular music. In electronic music the term is borrowed from the field of telecommunications and refers to the use of one electronic signal to modify another, such as the output of an LFO changing an oscillator's frequency. Changes in pitch, amplitude, and timbre can all be controlled using modulation.

Amplitude Modulation

Amplitude modulation (AM) is the use of a control voltage to alter (modulate) the loudness of another signal. The sound that is being modulated is called the *carrier* signal. When a sub-audio signal is used to modulate a given sound wave, the result is a slow, undulating effect called tremolo, in which the volume of the sound becomes alternately louder and softer but without changing the pitch. The loudness rises and falls around a central amplitude.

All types of waveforms can be used as control signals. Using a sine wave to modulate the carrier will cause the loudness to rise and fall very smoothly. A triangle wave will effect a gradual rise in loudness that sharply turns down and gradually falls, only to switch directions again very sharply. The use of a pulse wave as an amplitude-modulating signal eliminates the various gradients between loud and soft, and causes the carrier to switch instantly between the two extremes.

When the control signal is a waveform in the audible range, the changes in loudness become much more difficult to perceive because of their rapidity, and the resultant effect is a change in the harmonic structure of the carrier through the creation of audible sideband frequencies. Sidebands are the partials or harmonics that make up part of a total sound but do not dominate it. They change the tone color or timbre of the carrier. Sidebands are mathematically related to the carrier: the upper sidebands are equal to the sum of the carrier and control frequencies, while the lower sidebands are equal to the difference between them. When sidebands become audible, the carrier signal still remains the dominant signal.

Frequency Modulation

Frequency modulation (FM) is the use of a control voltage to alter the frequency (pitch) of the sound. A sub-audio control voltage (less than 20 Hz) will produce a vibrato effect, which is an undulation of pitch around the carrier tone. As in amplitude modulation, when the control voltage is in the audible frequency range, the resultant signal contains sidebands of the carrier wave and the very rapid undulation of pitch is perceived as a change in timbre. The complexity and harmonics of FM sidebands are much more intricate and rich than those produced by AM. Unlike AM, FM sidebands may actually dominate the carrier tone. The degree of undulation of the pitch will vary in proportion to the amount of attenuation of the carrier as well as the type of waveform being used. Figure 7.15 visually shows the effect of using different waveshapes on FM modulation.

Ring Modulation

Ring modulation is a form of amplitude modulation in which special circuitry suppresses the carrier signal and reproduces only the sidebands. Two additional frequencies are created in place of the original carrier signal. One is equal to the sum of the two input frequencies, and the other is equal to the difference between them. If the input signal has many harmonics, such as a guitar or the human voice, the resulting output signal is complex and rich—a kind of ghost of the original sound. The analog ring modulator made by the Moog Music Co. has a second input signal in the form of an oscillator. This can be adjusted to narrow or widen the distance between the two frequencies generated by the effect.

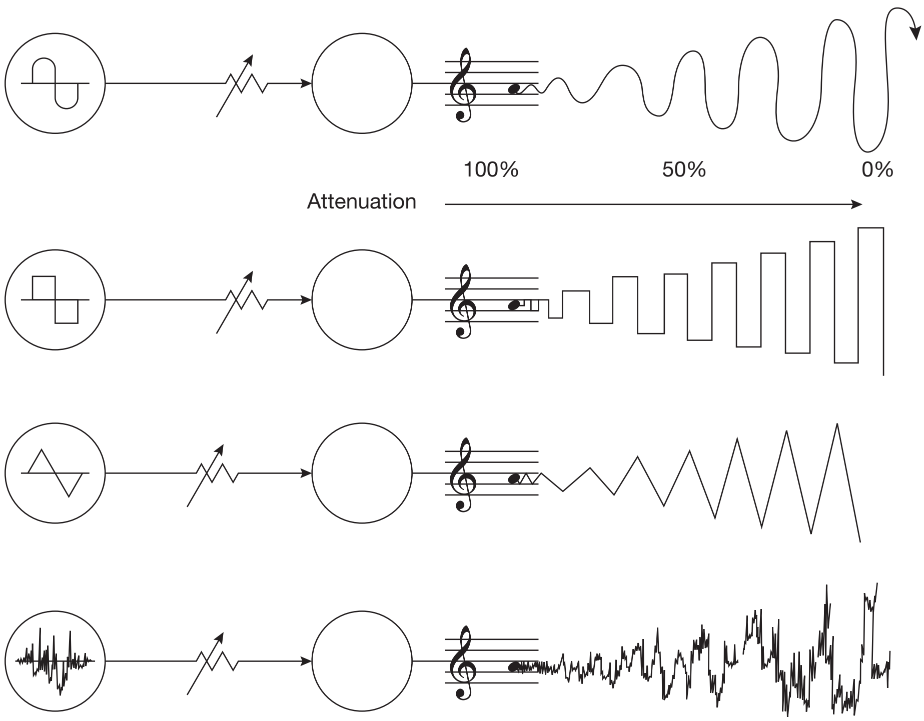


Figure 7.15 The effects of frequency modulation (FM) using different waveforms on a given signal source and attenuation increasing from 0 percent to 100 percent. (after Strange, 1983)

Pulse Width Modulation

Pulse width modulation (PWM) provides another technique for modulating the timbre of a frequency. This form of modulation takes advantage of the fact that the harmonics of a waveform will change according to the duty cycle of a pulse wave. The duty cycle—and pulse width—can be modulated by a low-frequency oscillator to provide subtle, although detectable modifications of the harmonic spectra associated with a pulse wave.

ANALOG SOUND SYNTHESIS

Synthesis is the ability to use the fundamental building blocks of sound to construct new sounds. Most electronic music composers prior to the 1960s had no purpose-made synthesizers at their disposal. Armed only with the basic building blocks comprising waveform oscillators, filters, tape recorders, and various other sound processing devices, they learned how to combine and modify existing sounds to make new ones from the simplest component parts. Through the development of solid-state miniaturization, early analog synthesizers provided many of the same audio-processing components as an entire studio but in the guise of a few, integrated modular desktop components. But the actual synthesis of sounds relied on the same trial and error process that had been in use since the early 1950s.

The term "synthesis" connotes a desire to create unique electronic instrumental voices. Such voices may be designed by the composer to imitate the timbre and response of conventional instruments such as those found in the classical orchestra. But the possibility of modeling equally compelling new sounds is equally plausible. Prior to the advent of digital sampling and synthesis (see Chapter 11) the techniques for crafting electronic instrumentation using analog techniques were challenging, required much patience on the part of the composer, and were sometimes difficult to reproduce due to the precision needed to devise—and repeat—the parametric settings needed to produce the desired sound.

The simplest form of sound synthesis is the combination of two or more sine waves into a more complex waveform. This process is called **additive synthesis** and can be used to create diverse sounds by building up layers of many individual sounds. Additive synthesis is based on the observation from Fourier theory that a periodic sound is composed of a fundamental frequency, which is dominant, and partials that have a mathematically harmonious relationship to the carrier. In the electronic music studio, the individual frequencies and their amplitude relationships can be manipulated in such a way as to duplicate or modify the sound synthetically. Synthesizers allow for the construction of complex sounds from simpler individual components and offer the ability to manipulate their frequency and amplitude interrelationships. Additive synthesis was the method used by many of the earliest electronic music composers. Stockhausen's first experiments with sine wave generators began as exercises in additive synthesis.

Subtractive synthesis is another technique used since the early years of electronic music. Just as waveforms can be constructed by the addition of one sound to another, they can also be altered through the systematic elimination of certain parts of the sound, such as overtones or the fundamental frequency. Subtractive synthesis begins with a complex waveform and subjects it to filtering using any one of the techniques described earlier in this chapter. French composer Eliane Radigue (b. 1932) is a classic analog synthesist who has used subtractive synthesis as the focus of her works. She has been working with an ARP 2500 analog synthesizer since the early 1970s and makes use of the instrument's manual controls for mixing waveforms into gradually changing sound textures. Radigue first learned electronic music composition in Paris from Pierre Schaffer and Pierre Henry, but her affinity for music consisting of slowly unraveling processes is distinct from classic *musique concrète* in which tape manipulation and editing are such important elements.

Early performance synthesizers, such as the Moog Minimoog, incorporated some of the first logical steps away from the use of patch cords and manually controlled parameters to preset controls for instrumental voices based on additive and subtractive synthesis techniques. The Minimoog had no patch cords and although it did not include specific preset voices it greatly simplified the modification of sounds by providing only rudimentary controls over envelopes, amplitude, and other modulation. For example, an Emphasis dial with ten settings could be used in conjunction with a Cutoff Frequency (filter) dial to produce a sharp resonance in the filter. Eliminating the patch cords with preset circuits for controlling waveshaping parameters greatly freed the performing musician to concentrate on playing. This was an innovative improvement, but playing the Minimoog was still not as simple as flipping a switch to get the desired sound. By way of an example, note the following instructions taken from the Minimoog operating manual for adjusting the attack characteristics of a sound:

> The ATTACK TIME control determines the duration of the initial rise in volume to a peak. Turn off the Noise Source and turn on Oscillator 1. Move control back and forth while repeatedly pressing down a key. Notice the different qualities which a note takes on as a sharp attack becomes a slow crescendo.[14]

The Minimoog was soon followed by more advanced analog synthesizers by Moog and other companies that incorporated an increasing number of presets made to approximate the distinctive voices of many instruments, such as violins, horns, and pianos, among others. By the end of the 1970s, the availability of increasingly affordable computer circuits began to improve the programmability and sequencing features of analog synthesizers, eventually leading to fully digital instruments using a new wave of diverse synthesizing techniques (see Chapter 11).

COMPONENTS OF THE VOLTAGE-CONTROLLED SYNTHESIZER

All analog, voltage-controlled synthesizers, whether modular or integrated by design, were comprised of several common building blocks. Although the specific way in which each manufacturer engineered these components varied, the expected results could be managed through the application of basic principles of voltage-controlled sound processing. The most common sound modules included the following:

- **Two or more oscillators for generating raw sound material**. The waveforms normally offered included sine, sawtooth, square, and sometimes triangle. Those waveforms could be combined to create variations on the default waveshapes through modulation.
- **Preset sounds, or instrumental "voices**." Modular synthesizers from the 1960s only came with basic waveform generators from which a composer would construct desired instrumental sounds. By the mid-1970s, the use of preset waveform generators and memory chips introduced the availability of preset voices requiring no additional programming.
- **White noise generator**. Variations on white noise generators—usually applying preset filters to produce specific bands of the noise spectrum—were offered by many manufacturers.
- **Voltage-controlled amplifier (VCA)**. Adjusted the loudness of a signal in proportion to a control voltage input.
- **Voltage-controlled filter (VCF)**. Provided a cutoff frequency that was adjustable in proportion to a control voltage input. Most VCFs also included voltage-controlled resonance, which accentuated frequencies near the cutoff and provided a hollow, ringing quality to the sound. VCFs were often designed for specific filtering functions and included band-pass, band-reject, high-pass, low-pass filters intended to pass only certain ranges of the sound spectrum.
- **Envelope generator (ENV)**. Controllers for modifying the way a sound starts, continues, and ends. Whereas an envelope generator is used to shape the loudness curve of a sound, an envelope *follower* is used to detect and respond to the loudness curve of an incoming signal.

KEY TO DIAGRAMMING NOMENCLATURE

Schematics of audio-processing modules included in this and other chapters adopt the following visual nomenclature for diagramming components and the flow of a signal source.

The *signal path*, indicated in a left-to-right direction using an arrow, follows the path of a voltage signal through the necessary stages of sound processing required to complete the function of a given module (see Figure 7.16).

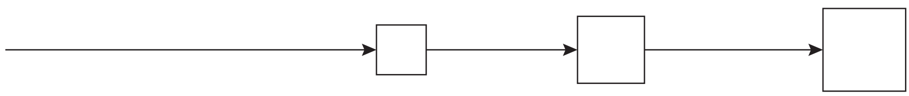


Figure 7.16 Signal path. (after Strange, 1983)

The waveform is indicated by a circle and a particular kind of waveform is illustrated within the circle (see Figure 7.17).

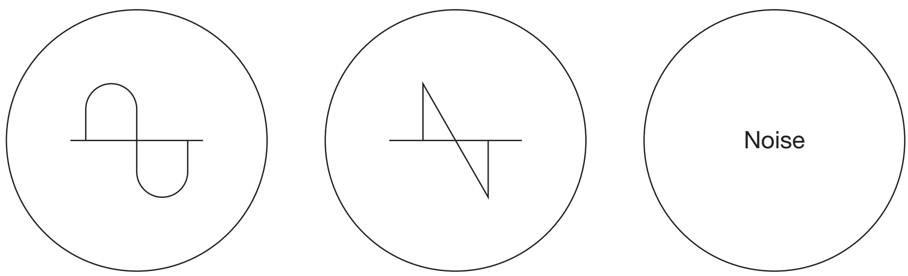


Figure 7.17 Waveform symbols. (after Strange, 1983)

Two lines joined by a dot indicate that a signal is patched to a module (see Figure 7.18).

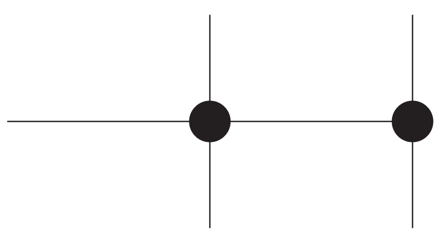


Figure 7.18 Patch symbol. (after Strange, 1983)

The attenuation—adjustment—of a signal prior to its linkage to a module is indicated by the symbol in Figure 7.19.

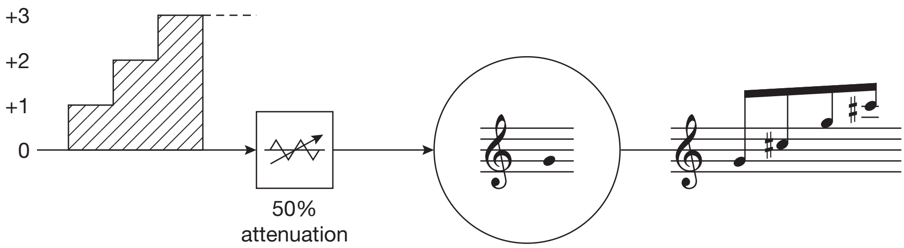


Figure 7.19 Attenuation symbol. (after Strange, 1983)

The symbols for other modules will be clearly labeled. Examples include those in Figure 7.18.

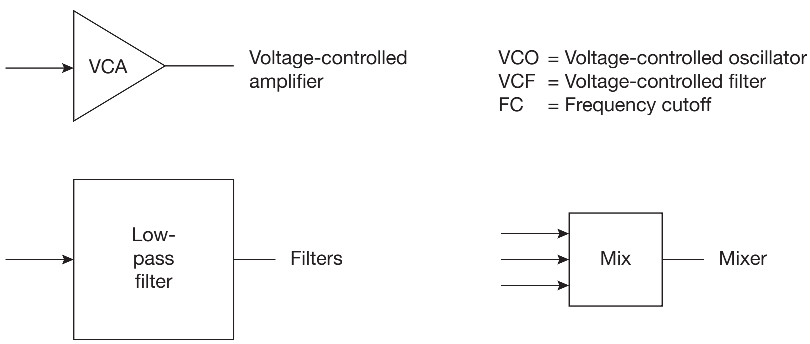


Figure 7.20 Other miscellaneous symbols. (after Strange, 1983)

A simple patch would be illustrated as in Figure 7.21; note that the rightmost arrow represents the final output signal that can be amplified for listening.

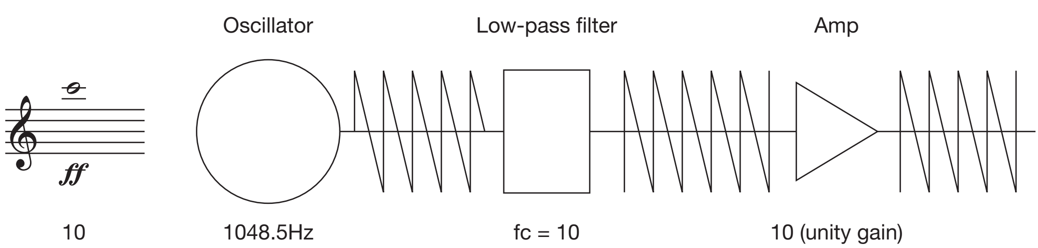


Figure 7.21 Patch diagram. (after Strange, 1983)

- **Sequencer**. One of the most diverse sources of voltage control, and could be used to generate patterns of tones or programmed changes in amplification, filter, mixing, modulation, and sound distribution.
- **MIDI**. MIDI IN/OUT/THRU for controlling one or more keyboards or interfacing a synthesizer with a computer in real time (see Chapter 8, pp. 278–85).

Synthesizer Configurations

See Figures 7.22 and 7.23.

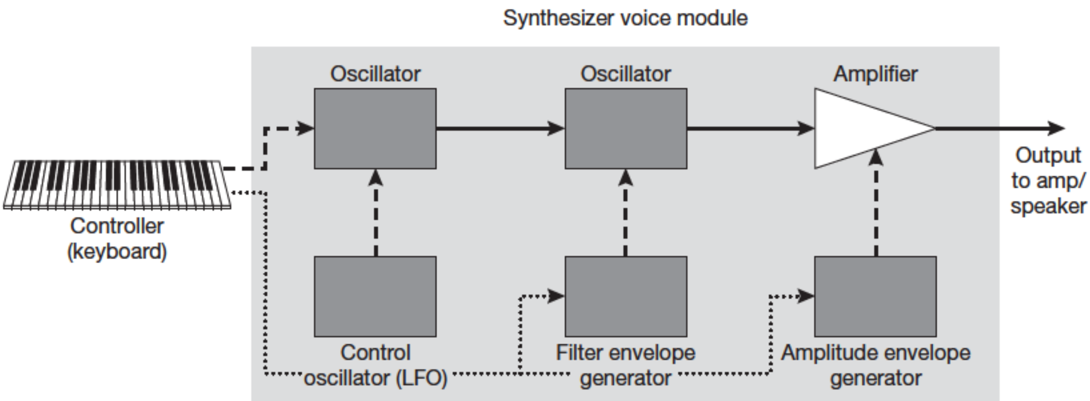


Figure 7.22 **Schematic for a basic analog synthesizer.** (after Crombie, 1982)

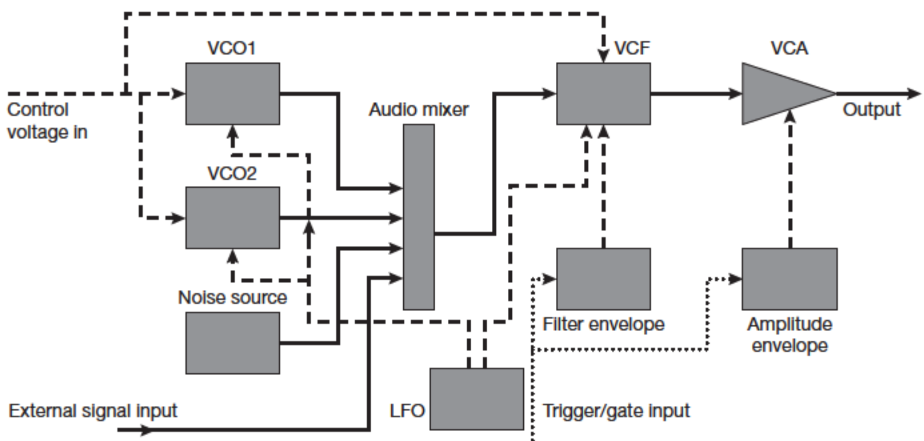


Figure 7.23 **Schematic for a basic synthesizer voice module.** (after Crombie, 1982)

SUMMARY

- Helmholtz showed that the vibrations found in a single musical tone consisted of a fundamental or base tone accompanied by related harmonics above the pitch of the fundamental.
- If a wave vibrates in a regular pattern, it is perceived as a pitched sound, such as those used in music. If the wave does not vibrate in a regular pattern, it is perceived as unpitched sound or noise.
- Components of sound include frequency, amplitude, timbre, duration, and envelope.
- Fourier theory states that any periodic vibration (waveform), however complex, is comprised of, or can be created by combining, a series of simple vibrations whose frequencies are harmonically related and that change in amplitude independently over time.
- Voltage control is a method of applying metered amounts of current to an electronic component to govern how it operates. It was a technique used to control the modules of analog synthesizers.
- A control voltage is discrete from the voltage used to generate an audio signal.
- Common voltage-controlled components of the analog electronic music studio included the voltage-controlled oscillator (VCO), voltage-controlled filter (VCF), voltage-controlled amplifier (VCA), envelope generator (ENV), and low-frequency oscillator (LFO).
- Waveforms can be generated by an electronic circuit called an oscillator, which produces periodic vibrations in the form of an electric current.
- Common waveforms used in music synthesis include sine, sawtooth, triangle, and pulse waves.
- White noise results when all the frequency and amplitude characteristics of a sound occur at random within the audio spectrum. White noise contains equally distributed energy at all frequencies within the audio spectrum.
- Electroacoustic music is broadly defined as music created using electronic and acoustic sound sources.
- Microphones and pickups are two common electroacoustic transducers and are designed to change vibrations in the air or on a solid surface to electric current.
- Acoustic feedback occurs when a sound amplified via a microphone or pickup is re-amplified again and again via the same microphone or pickup. A feedback circuit enables the internal generation of signals by connecting output back to input, prior to their amplification in the listening space.
- Forms of analog signal processing include frequency filtering, envelope shaping, echo, reverberation, loops, delay, and signal modulation such as amplitude modulation (AM), frequency modulation (FM), ring modulation, and pulse width modulation (PWM).
- Analog sound synthesis is commonly done using additive synthesis by combining waveforms, or subtractive synthesis by using filters to eliminate frequency ranges from a given sound.