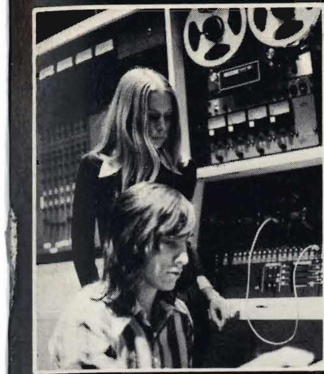

LEARNING MUSIC WITH SYNTHESIZERS

DAVID FRIEND
ALAN R. PEARLMAN
THOMAS D. PIGGOTT

LEARNING MUSIC
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CONTENTS

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Part I covers the basic theory needed for understanding sound synthesis, and many new terms are presented and explained. These terms are the language of synthesizers and should be understood by those wishing to expand their musical experiences through synthesizers.

Part II is a "hands-on" approach to mastering the ARP Odyssey synthesizer. More terminology is presented and explained, and the experiments and exercises always relate to the theory presented in Part I of the text. The practical understanding of the operation of the synthesizer gained through studying this part of the text will quickly enable anyone to learn how to operate nearly any synthesizer.

Part III of the text ties this newly mastered instrument into the framework of traditional musical concepts. The synthesizer is a remarkably versatile and flexible tool for the musician, and this part of the book explains how the synthesizer can be used to demonstrate and reinforce many basic concepts in music and sound.

The material presented in this book will be best understood if the three major parts of this book are studied in the order presented. And although there is no substitute for practical experience with a synthesizer, it is possible to gain a good basic understanding of electronic music production by studying the many detailed diagrams in this book.

The synthesizer represents a major evolutionary step in music and music education. It adapts so well to such a wide variety of applications that its usefulness is limited only by one's imagination. In this book we have offered the tools necessary to begin exploring and enjoying the new capabilities of electronic music and synthesizers.

The authors would like to express their thanks to Dave Fredericks of ARP Instruments, Inc., and to Gary Meisner and Dick Peck of Hal Leonard Publishing Corporation for their invaluable assistance in writing this book, and to Vivian Hutchins, editor, and Margaret Shepherd for art direction.

Newton, Massachusetts
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PART I

Introduction: Electronic Sound Synthesis

Music is an art which uses sound waves as a medium for conveying ideas from a musician to a listener.

A musician uses either the human voice or instruments to create musical sounds. The human voice is naturally expressive; the quality or *timbre* of every sound it makes is extremely flexible and capable of both subtle and dramatic variations from note to note and within individual notes. The expressive capabilities of the human voice make it possible to achieve many artistically useful emotional qualities.

Music made by musical instruments can also be expressive. However, the range of timbres and pitches obtainable from most instruments is inherently limited. These limitations must be overcome by the composer, arranger, and performer to obtain musically pleasing and artistically valid qualities. A symphony orchestra, for instance, overcomes some of these limitations by using many different instruments, which together have a wide range of pitches and timbres.

Solo instruments, of course, create special problems for the composer and performer because of their limited range of sounds. Even complex polyphonic instruments like pianos and organs have a relatively limited range of timbres and provide very little control over the sound once a note has been struck.

Electronic music synthesizers are the most expressive musical instruments yet designed, since they allow for the most complete control of timbre variations. Like the human voice, it is possible for every note played on a synthesizer to have a timbre different from its neighboring notes. And it is possible to change the timbre of a note while it is being played, either over an enormous range of tone colors, or through very subtle changes. And most synthesizers can produce an enormous range of pitches, from lower than the lowest organ pipe to higher than the human ear can perceive sound.

The family of synthesizers includes instruments that are capable of imitating traditional instruments, and also of creating "new sounds" never before obtainable with any instruments. Synthesizers can produce pitched sounds, unpitched sounds (wind, thunder, machinery, etc.), and sequences of sounds too complex or too fast for performers to play on ordinary instruments.

Two basic ideas are involved in all electronic sound synthesis. The first is that acoustical waveforms — virtually any sound you hear — can be *generated* and *modified* by purely electronic means. In short, any sound from the musical sound of a clarinet to the howling of the wind outside your home during a thunderstorm can be produced electronically, given the right kind of equipment. The second idea upon which electronic sound synthesis is based is that this sound-generating and sound-modifying equipment can be *controlled* electronically.

With a synthesizer, you can both invent and imitate sounds. It is a scientific mixer, sifter, and producer of sound waves, and it opens up truly unlimited scope for experimentation. The "super hand" of voltage control carries out your commands instantaneously and with great precision; by the time you have finished this book, you will know *why* and *how* you can achieve the results you want.

Lesson 1: Waveforms

Now that we've established the two basic ideas involved in electronic sound synthesis, let's back up to the first idea and examine waveforms in more detail. Sound is transmitted through several mediums; the one we are most concerned with here is air. For example, assume that you have an A-440 tuning fork. (Your school music department probably has one that you can use.) When you tap this tuning fork on the edge of a desk or a table, you are setting up a series of vibrations that produce the sound you hear. An A-440 fork vibrates at the rate of 440 cycles per second, producing the note — or pitch — A above middle C. Stated another way, this means that in 1/440th of a second, the fork makes one complete vibration back and forth (Figure 1.1).

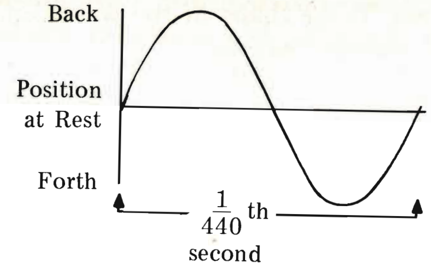


Figure 1.1. Tuning-fork vibration.

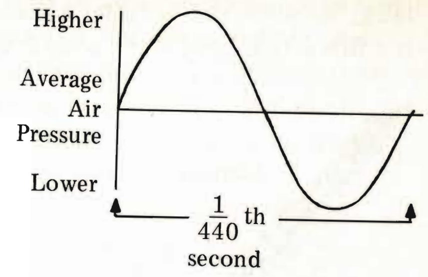


Figure 1.2. Barometric variation.

If you had an exceedingly precise barometer, it would register a variation in air pressure during that same 1/440th of a second (Figure 1.2). The displacement of air which the barometer would measure is simply a series of air waves set up by the vibration of the tuning fork. When these waves reach your eardrum, they set up a vibration corresponding to the one created by the natural, or acoustic, sound source (Figure 1.3).

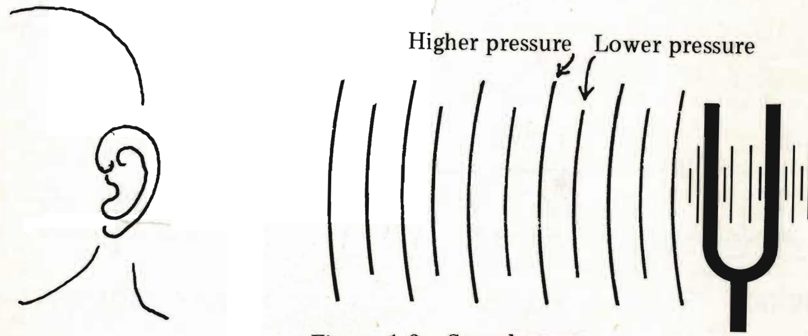


Figure 1.3. Sound waves.

When someone speaks, sings, or plays an instrument, similar air waves are generated, transmitting the sound to your ear. These sound waves can be picked up by a microphone and converted into electrical signals (Figure 1.4). The electrical signal for each individual sound has a particular shape, which can be shown on an oscilloscope. The shape of each signal is called its *waveshape* or *waveform*.

The fact that waveforms have a definite shape is an important concept to remember. The sound of your voice, the sound of a trumpet, and the sound of an automobile horn all produce waveforms, and each of these waveforms has a different shape. Different sounds have differently shaped waveforms (Figure 1.5). This is extremely important in terms of the sound synthesis you'll be doing before long, because the reverse is also true: if the shape of the waveform is modified (changed), the sound will also be changed.

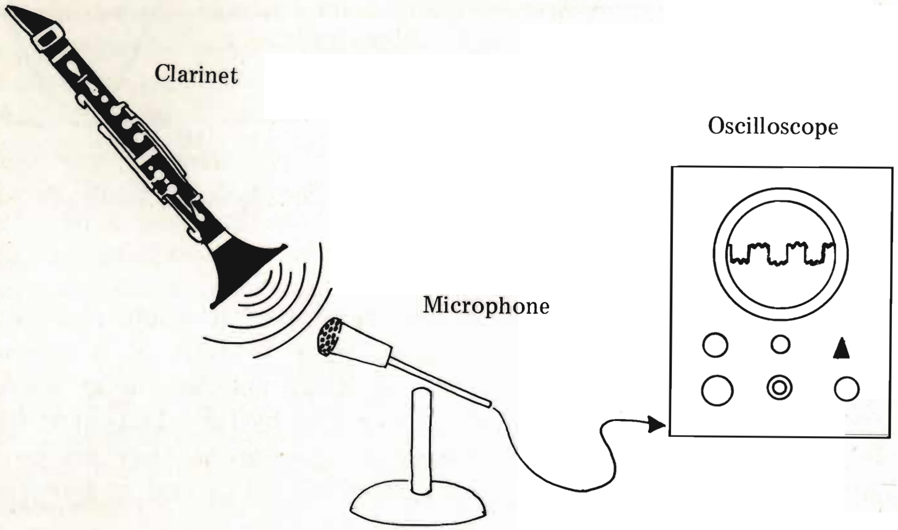


Figure 1.4. Sound-wave transmission.

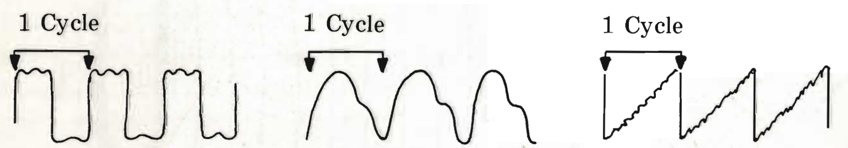


Figure 1.5. Three different waveforms.

Basic Waveforms

The most basic classification of waveforms breaks them down into two groups: *periodic* and *aperiodic*. Periodic waveforms have a repeating pattern, as in the case of the waveforms shown for the clarinet, the flute, and the brass instrument. Note the marking, "one cycle," the following complete cycle, as well as the preceding cycle, look very much alike — there is a repeating pattern. The shape of these waveforms could, and probably would, change somewhat over a longer period of time, but a definite repeating pattern would still be apparent.

Not all waveforms, however, are periodic. Some waveforms occur only once, or at irregular intervals and only upon command. Other

waveforms are so complicated that your chances of finding any kind of repeating pattern are almost nonexistent. In any case, a waveform that does not demonstrate any repeating patterns is called aperiodic (Figure 1.6). Sounds with aperiodic waveforms have no pitch — for instance, those made by thunder, surf, rain, wind, a snare drum, etc.

Periodic Waveforms

For now, we'll be most concerned with the periodic type, since most musical sounds have a periodic waveform. The shape of four common periodic waveforms is shown in Figure 1.7.

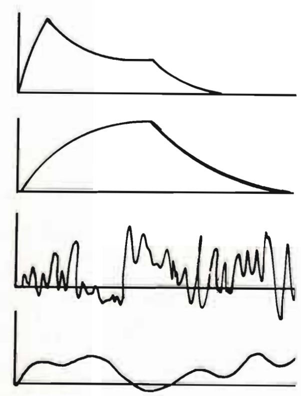


Figure 1.6.
Aperiodic waveforms.

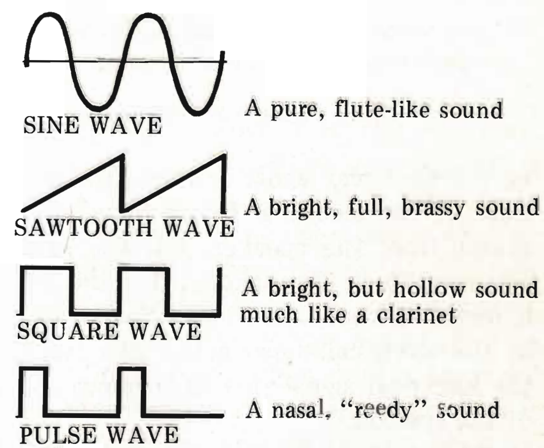


Figure 1.7. Periodic waveforms.

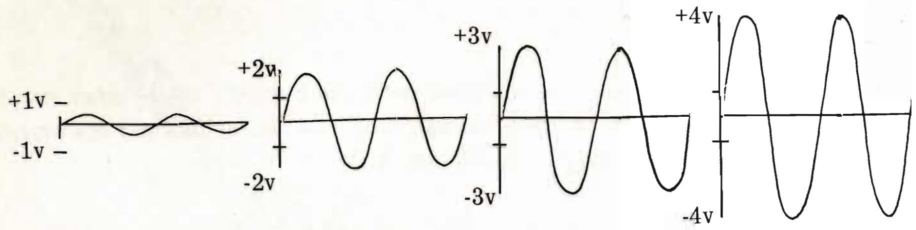
Amplitude and Frequency

We have established that every sound has a waveform with a particular shape. That waveform, however, is not perfectly constant; if it were, all that you would hear would be a monotonous tone, with no change in the volume, pitch, or timbre (tone quality). We can change the volume (amplitude) or pitch (frequency) of a waveform without changing its basic shape.

The amplitude of a waveform is the amount of maximum deviation from its "center." In a loudspeaker, this center is the position of the loudspeaker cone at rest; in an electrical circuit, it might be the condition of zero voltage. The amplitude of the loudspeaker cone's vibration would be measured in inches, or fractions of inches, that the cone moves back and forth. The amplitude of a fluctuating or alternating voltage would be measured in positive and negative volts.

Thus a voltage waveform that reached a peak of +1 volt and then of -1 volt would have an amplitude of 2 volts "peak-to-peak," as shown in Figure 1.8.

The volume control on a record player determines the amplitude



Amplitude 1 volt peak to peak | Amplitude 4 volts peak to peak | Amplitude 6 volts peak to peak | Amplitude 8 volts peak to peak

Figure 1.8. Sine waves of different amplitude.

of the electrical signal delivered to the speaker; the greater the amplitude of the electrical signal delivered to the speaker, the louder the sound from the speaker. For the sake of keeping our terminology accurate, then, we might say that the volume control on a record player is really an *amplitude control* since it really controls only the amplitude of the electrical signal going into the speaker. It is the amplitude of the electrical signal that determines the volume of sound coming out of the speaker.

An important point has now been raised in our discussion of amplitude. You'll save yourself a lot of confusion by thinking of amplitude only in connection with electrical waveforms, and of volume only in connection with sounds. For example, because there are no sounds inside a synthesizer — only voltages — there cannot logically be any volume controls. There are, however, a great many amplitude controls. In fact, until any electronic device — whether it is your stereo or a synthesizer — is connected to a loudspeaker, changes in amplitude can be made that will in no way affect the volume you're hearing. More important, in working with a synthesizer, changing the amplitude of a waveform in many cases will have an audible effect other than a change in volume of whatever sound you are creating. Therefore, while a change in the amplitude of an audible waveform could change the volume of the sound you're hearing, it is more precise and much less confusing to speak of amplitude in connection with waveforms, and of volume only in connection with an actual change in the loudness of a sound.

The *frequency* of a waveform also does not change the basic *waveshape.* We have already established that periodic waveforms have a repeating pattern or cycle. The rate at which that cycle is repeated is the frequency of the waveform. For example, when discussing the tuning fork we said that it was tuned to A-440, producing A above middle C. This means that the waveform frequency required to produce this note is 440 cycles per second. If the frequency were lower, the pitch of the note you hear would be lower. If the frequency were higher, the pitch would be higher. There is an international standard unit of measurement for frequency: this unit is "Hertz" (abbreviated as Hz) and it is defined as one cycle per second. Thus, the A pitch that we have been discussing is produced by a frequency of 440 Hz. The prefix "Kilo-" means "one-thousand," and a frequency of 1,000 cycles per second is called "one KiloHertz," or 1 KHz.

We've established that changing the frequency of the waveform changes the pitch. However, just as with amplitude and volume, it is important to keep the ideas of frequency and pitch separate in your mind. Frequency is a characteristic of all vibrations, whether mechanical or electrical, while pitch is a characteristic of the way human beings *perceive* acoustic vibrations between approximately 20 Hz and 20 KHz. Another way to say this is that every pitch (a sound we can hear associated with a particular frequency) is produced by some frequency, *but many frequencies produce no pitch.* This is because there are many frequencies above and below the range of

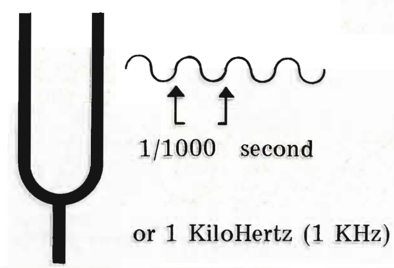


Figure 1.9

human hearing. If you were able to try the experiment with the tuning fork, by tapping it on the side of a desk and listening to the pitch produced, you already know that you can hear the frequency 440 Hz. On the other hand, you would not be able to hear 4.40 Hz — it is far below audibility. Nevertheless, subsonic, or low, frequencies are very useful in synthesizing many sounds, and it is for this reason that frequency must be thought of as being related to, but distinct from, pitch.

One final point concerning amplitude and frequency: *changing the amplitude or frequency of a waveform does not alter the basic shape of the waveform.* In Figure 1.10, illustrating the three sawtooth waves of different frequencies, you'll notice that the basic shape of the waveform does not change at all — it is simply narrower and appears more often in the same period of time as the frequency rises. The essential characteristics of the shape are still present. The same is true of amplitude; if you examine the three sine waves of different amplitudes, you'll see that the general shape of the waveform does not change. It still looks like a sweeping S-curve, with just slightly different dimensions. *Changing the frequency or amplitude of a wave will not count as a change in its shape* (Figure 1.11).

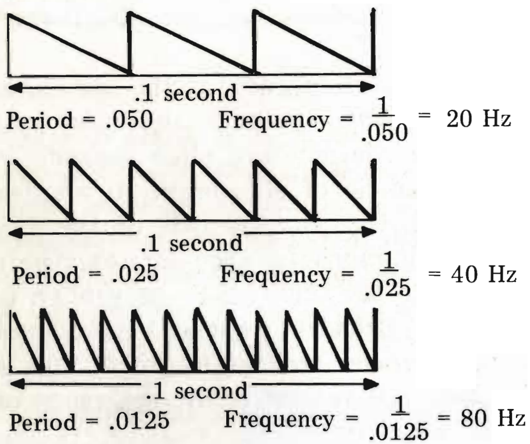


Figure 1.10.
Sawtooth wave of
three different frequencies.

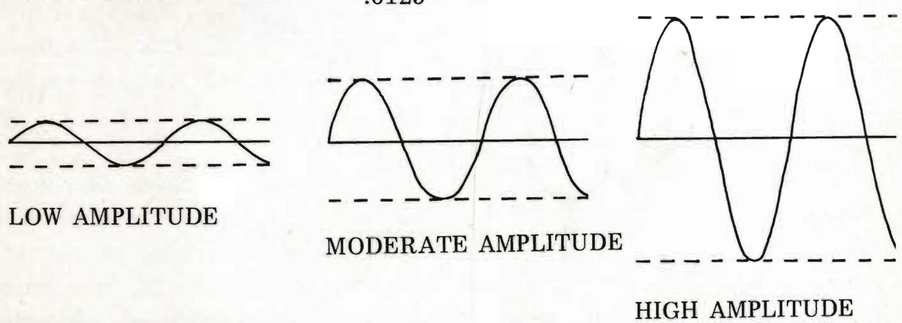


Figure 1.11. Sine waves of three different amplitudes.

Timbre

Changes in the shape of an audio waveform are generally associated with changes in the tone quality, or *timbre*, of the sound produced. Generally speaking, timbre is the subjective quality of a tone which

enables the listener to distinguish between it and other tones which may have the same pitch and/or loudness. For example, trumpets and clarinets produce sounds of different timbres but can be equally loud. Since human perception has limits, changes in the shape of a waveform are possible that might not be perceived by even the most practiced ear. However, any time that a change in timbre is perceived, you can be sure that a corresponding change must be reflected in the shape of the waveform.

Summary of Terms

Based upon what we now know, we can set up the following chart summarizing these approximate relationships:

Perceived subjective changes in:	*Correspond to physical changes in:*
Volume	Amplitude
Pitch	Frequency
Timbre	Waveform or Waveshape

This is worth remembering because, as emphasized earlier, synthesizers work with the qualities listed in the right-hand column, whereas we perceive the results as changes in those subjective qualities listed in the left-hand column. With practice, you will learn to translate easily from the language of volume, pitch, and timbre to the language of amplitude, frequency, and waveshape.

Questions

1. If you use a tape recorder to record a clarinet playing an A-440 Hz and then slow the tape down to half speed, what would the pitch be and would the waveform shape change? Why?
2. If you talk with your face in a pillow, does the pillow alter the waveshape of your normal speaking voice? Why do you think so?
3. Are the following sounds periodic or aperiodic waveforms:
   (a) a car horn;
   (b) striking a garbage-can lid;
   (c) rustling a newspaper;
   (d) a squeaky door hinge?
4. What is the difference between volume and amplitude?
5. If a waveform repeats itself once every 1/100 sec, what is the frequency of the waveform? Is this a low-pitched sound or a high-pitched sound?
6. What happens when a sound gets so low in pitch that you can no longer hear it? What does it sound like then? How low do you think you can hear pitch?

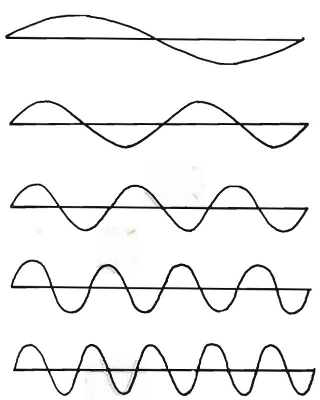
Lesson 2: Overtones and Harmonics

The Fundamental

Given the information that has now been presented, we can ask: "Why do differently shaped waveforms produce different sounds?" The principal reason is because these differently shaped waveforms have different *overtones.* Overtones can best be defined by example and illustration. Assume that you are listening to a particular pitch — low A, for example, at 110 Hz. The acoustic effect produced by a single vibration of this frequency would be described as a pure sound — one with no harmonics, or overtones. A sine wave is a pure sound. However, most natural sounds, such as the vibrating string of a violin, do not consist of just a single, simple vibration. Virtually all musical instruments produce complex, composite sounds, consisting of the main sound — the *fundamental* — plus a number of additional pure sounds of lesser amplitude called overtones.

Harmonic Frequencies

In most musical instruments, the overtones consist of vibrations that are simply multiples of the fundamental frequency. For instance, an A (110 Hz) would have overtones at 220 Hz, 330 Hz, 440 Hz, etc. Overtones like these, that are simple multiples of the fundamental, are called harmonics. It is the relationship of the relative amplitudes of the different harmonics that allows us to distinguish between two sounds with the same pitch. For instance, one of the reasons a tuba sounds different from a bassoon is that the harmonics produced by the two instruments are different in amplitudes with respect to the fundamental. Strings, trumpets, and reeds produce tones which are relatively rich in harmonics, while flutes and French horns produce sounds with relatively few harmonics. Figure 2.1 shows these harmonics with their pitches.



HARMONIC	FREQ	PITCH
Fundamental Sine Wave	110 Hz	Low A
2nd Harmonic	220 Hz	A, one octave above 110 Hz
3rd Harmonic	330 Hz	E above middle C
4th Harmonic	440 Hz	A above middle C
5th Harmonic	550 Hz	C sharp above A-440

Figure 2.1. Harmonic frequencies.

Therefore, the various regular waveforms differ from each other by their harmonic content and this difference accounts for their different sounds. Figure 2.2 illustrates the harmonic content of the sine, square, sawtooth, and pulse waves.

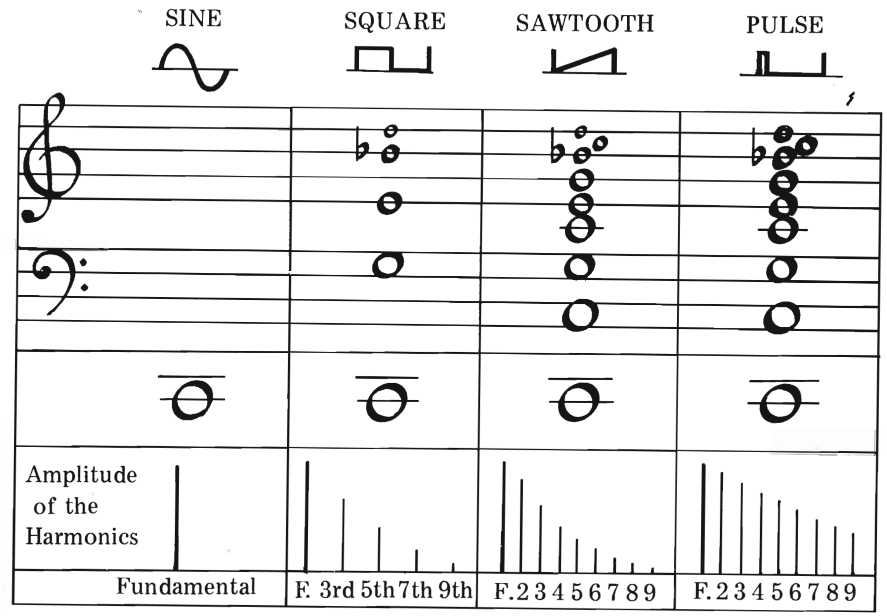


Figure 2.2. Harmonic series.

You'll note that the square wave is made up of the fundamental frequency plus all odd-numbered harmonics, while the sawtooth and narrow pulse are made up of the fundamental plus *all* harmonics. The essential difference between the sawtooth and the narrow pulse waves is that as you listen to the two, you will notice that the pulse wave sounds brighter. This is because the amplitude of the pulse-wave harmonics is greater than the amplitude of the harmonics of the sawtooth wave. Consequently, the greater amplitude of the higher harmonics results in the narrow pulse wave having the brighter sound to your ear.

Other Overtones

Some instruments, especially instruments involving the striking of metal, like gongs, chimes, triangles, and bells, produce overtones that are not exact multiples of the fundamental. Sometimes, as with a Chinese gong, there are so many overtones that it is virtually impossible to tell which is really the fundamental pitch. The frequencies of these overtones can be in very odd relationship to the frequency of the fundamental, like 7/2 or 9/4.

Since the vibrations created by the oscillators (tone generators) on a synthesizer have only simple harmonics, a device called a "ring modulator" is provided to create overtones which are not harmonics, thereby permitting the synthesis of metallic sounds.

Waveshapes

As you might expect, the chart showing the four waveshapes — sine, square, sawtooth, and pulse — does not represent every conceivable waveshape. As Figure 2.3 shows, there are a number of intermediate waveshapes that consist of the fundamental plus fewer harmonics than are found in the square, sawtooth, and narrow pulse waves.

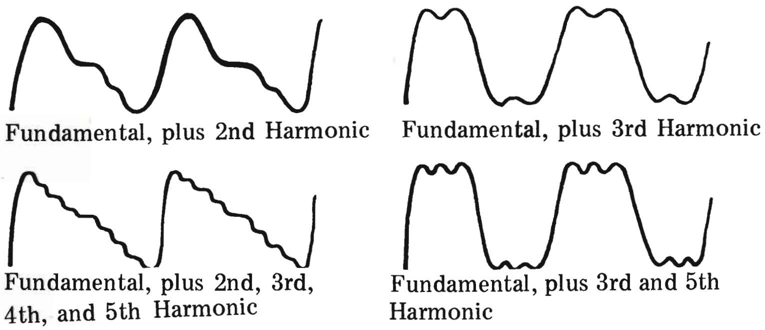


Figure 2.3. Waveshape created by adding harmonics.

It is useful to note that as more higher harmonics are added, the corners on the corresponding waveforms become sharper and more abrupt. Thus, looking at a waveform that has smooth, rounded features should tell you that this waveform will sound mellow or flutelike (Figure 2.4). On the other hand, a jagged waveform with sharp corners will always produce a bright, buzzing sound. Figure 2.5).

Figure 2.4. Mellow sound has smooth-looking waveform.

Figure 2.5. Bright sound has jagged, angular waveform.

Based upon what you've now read concerning harmonics, you have the basic knowledge necessary to understand the significance of the following point: the sine wave is the simplest waveform, and all other waves are actually combinations of sine waves of related frequencies, called harmonics. Stated another way, complex waveforms can be synthesized by adding together a number of simple waves; logically enough, this process is called *additive synthesis.* Remember this term, because the converse of this concept will appear in the following discussion of *filtering.*

Questions

1. What is the relationship between timbre and harmonics?
2. If you change the harmonics in a sound, will the timbre change? Will the waveform change?
3. Which waveform would sound brighter: a waveform with smooth, flowing features or a waveform with sharp, jagged corners?
4. Which instrument produces more harmonics: a slide whistle or a kazoo?
5. What is meant by "additive synthesis"?

Lesson 3: Filters and Filtering

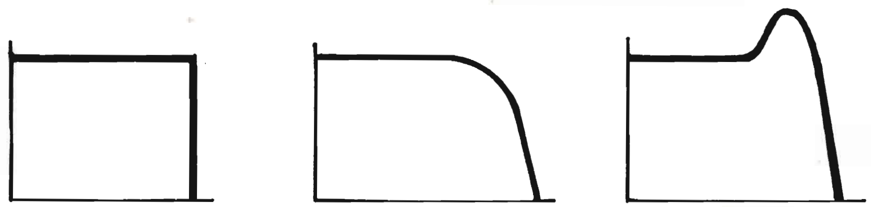
Subtractive Synthesis

Just as complex waveforms may be created by adding a number of simple waves together, it is possible to simplify a complex wave by filtering out certain frequencies. This process is called *subtractive synthesis.* This concept is extremely important, as it will form the basis for much of the sound synthesis you will be doing on the ARP Odyssey and, in fact, with any synthesizer.

The tone controls of a stereo are no more than simple filters capable of removing or attenuating the high (treble control) or low (bass control) frequencies. Synthesizers, on the other hand, may employ as many as four types of somewhat more sophisticated filters. These filters, and their specific functions, are shown below.

An ideal low-pass filter would pass all frequencies up to the cutoff point and then completely eliminate all frequencies above this cutoff point. A graph showing how an ideal low-pass filter works is shown in Figure 3.1 (a). In practice, however, it is impossible to create an ideal low-pass filter. Instead, low-pass filters (such as the one on the Odyssey synthesizer) pass all frequencies up to the cutoff point, and then gradually reject the frequencies above this cutoff point. This gradual attenuation of the higher frequencies is called "rolloff." The response of a low-pass filter with rolloff is shown in Figure 3.1 (b).

Some low-pass filters, like the one in the ARP Odyssey, have an additional characteristic called resonance. Adding resonance to a low-pass filter causes the filter to emphasize a band of frequencies just at the cutoff point. The response of a low-pass filter with resonance is shown in Figure 3.1 (c). Resonance is extremely important in synthesizer filters. Almost every mechanical instrument has its own characteristic resonances, and in order to simulate most natural sounds, the synthesizer must be able to duplicate these resonances.



3.1 (a) Idealized LPF 3.1 (b) LPF with Rolloff 3.1 (c) LPF with Resonance

Low-pass filters with resonance occur commonly in nature. When you talk through a long pipe, for instance, you will notice that your voice is muffled because the pipe is a low-pass filter, but your voice also has a kind of nasal quality which is characteristic of resonance. A pillow would be an example of a low-pass filter without resonance. Your voice is simply muffled by the pillow.

Low-Pass Filter. Each filter name is descriptive. As this name would imply, the low-pass filter passes all frequencies *below* a certain cutoff point.

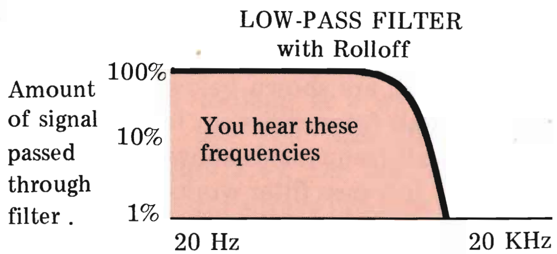


Figure 3.1.
Low-pass filter function.

High-Pass Filter. This filter passes those frequencies *above* the filter cutoff frequency (Figure 3.2).

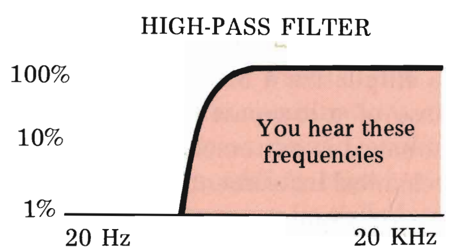


Figure 3.2.
High-pass filter function.

Band-Pass Filter. The band-pass filter passes only a certain band (range) of frequencies, eliminating or attenuating those frequencies both *above and below* the band being passed (Figure 3.3).

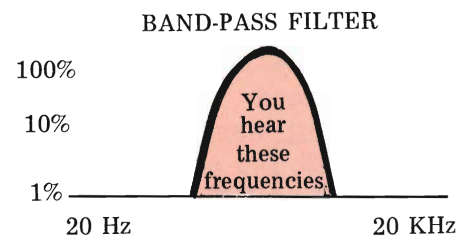


Figure 3.3.
Band-pass filter function.

Band-Reject Filter. This filter passes all frequencies *except* a certain band of frequencies. Thus, the function that this filter performs is exactly opposite to that of the band-pass filter (Figure 3.4).

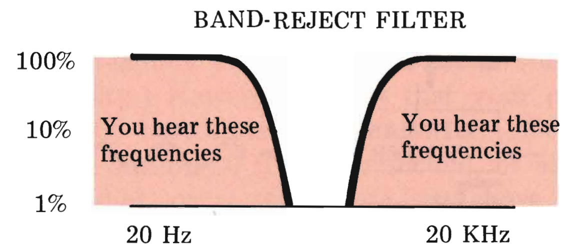


Figure 3.4.
Band-reject filter function.

The most useful filters are the low-pass and the band-pass filters. This is because the bodies of most natural, or acoustic, instruments (the tubing and bell of a trumpet, the wooden box of a violin) are low-pass and band-pass filters. The ARP Odyssey has one voltage-controlled filter which can be used as either a low-pass or a band-pass filter. In addition, it has a separate manually controlled high-pass filter.

Waveform Modification

The important point to remember here is that the filter(s) of a synthesizer are waveform modifiers, capable of changing the shape, and therefore the sound, of the waveform passing through the filter. Basically, this is done by filtering out certain components of the waveform. For example, if all the harmonics of a square, or sawtooth, or pulse wave are filtered out, the result will be a sine wave. Therefore, by filtering — or subtractive synthesis — the shape of the waveform has been reduced to its fundamental component. Passing a square wave through an idealized low-pass filter will change the waveshape. In Figure 3.5 (a) through (d), the cutoff point of the filter is made lower and lower, cutting off more of the harmonics (Figure 3.5). This selective removal of the overtones or harmonics will become extremely important as you begin synthesizing both instrumental and electronic sounds.

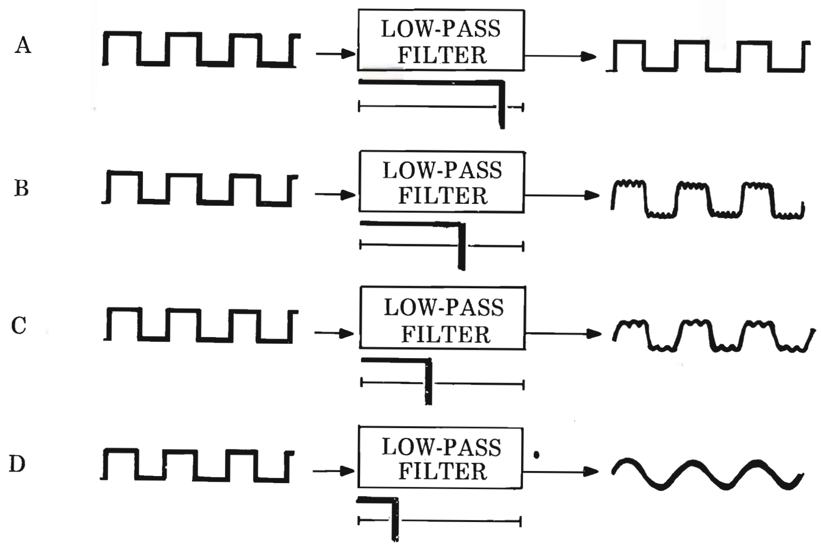


Figure 3.5 Filtering out harmonics with an Ideal Low-pass Filter

Notice that the same waveforms can be created by filtering the harmonics out of a square wave in Figure 3.5 as were created by adding together sine waves in Figure 2.3.

Because the low-pass filters on synthesizers only approximate an ideal filter, we cannot produce the exact waveforms shown in Figure 3.5. To your ear, however, the approximation is very close indeed. The basic principle of subtractive synthesis—shaping a waveform by filtering—will allow us to create an enormous range of musical sounds with the filters available on synthesizers such as the ARP Odyssey.

Questions

1. When you talk through a pillow, your voice becomes muffled. Do you think the pillow is a filter? What kind? Why?
2. Would passing a waveform through a high-pass filter make the sound brighter or muffled?
3. When you listen to music over a telephone, it doesn't seem to have good bass or good treble. What kind of filter is the telephone?
4. Describe the process of subtractive synthesis.

Lesson 4: Low-Frequency Waveforms

Subsonic Frequencies

Up to this point we have been largely concerned with periodic, audio waveforms. A synthesizer, however, can generate and modify both audio frequencies and subsonic, or low, frequencies. You'll remember that we call those frequencies between approximately 20 Hz and 20 KHz "audio frequencies." This is because it is only when physical vibrations are within that range that we perceive them as sounds having a definite pitch. However, it is possible to hear things happening at frequencies lower than 20 Hz — but you'll hear them as separate and repeating sounds, not as continuing tones or noises. For example, you can plainly hear your heartbeat by listening through a stethoscope. Although it would be rather slow for a pulse rate, let's say that your heart is beating 60 times per minute. That's the same as saying it beats once every second, or at a frequency of 1 Hz. Now imagine your heartbeat gradually increasing to 120 per minute, or 2 Hz. (Your heart frequently beats this fast during moderately strenuous exercise, but 2 Hz is still well below the limits of audibility.) However, assume that your pulse rate could run on up to 1,200 heartbeats per minute. At 1,200 per minute, your heart would be beating 20 times per second, or at a frequency of 20 Hz. At this point, or at somewhere just above 20 Hz, you would lose your sense of hearing individual heartbeats, and instead would begin to hear a very low pitch, which would gradually rise with the rising frequency of your heartbeat. As your heartbeat slowed, the opposite would happen: first, you would hear a falling pitch, then a gradual transition to a state of hearing no pitch at all, but rather separate and countable beats, or *events*.

We can use another example: playing a note on a piano, at a rate of one note every second, produces a series of low-frequency events at 1 Hz. These events do have a certain pitch because each event is the occurence of an audio-frequency vibration. A graph of everything that is happening as you play three successive notes would look like Figure 4.1:

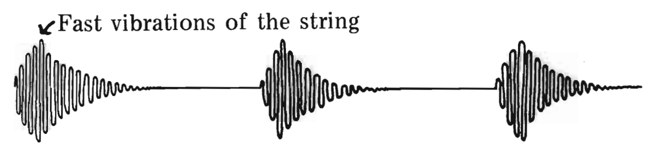


Figure 4.1. Low-frequency events.

The low-frequency waveform involved would look like Figure 4.2:

Figure 4.2. Low-frequency waveform.

And the audio frequency that produces the pitch you hear would look like Figure 4.3.

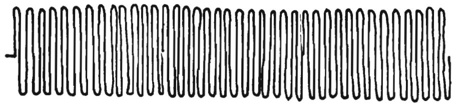


Figure 4.3. Audible part of sound in Figure 4.1.

Note that when an event, or series of events, is represented by a graph, you can derive the shape of the low-frequency waveform by simply connecting the highest points on the audio-, or higher, frequency waveform. Now we can see that *events have shapes,* and the shape of any event is the shape of the low-frequency waveform that can "produce" that event. For example, playing a staccato tune on an organ, where decay is almost instantaneous, would produce a series of different pitches that would look something like Figure 4.4:

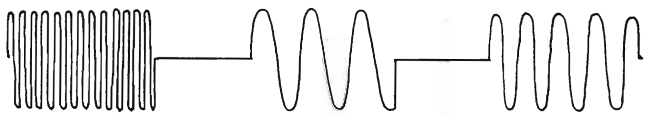


Figure 4.4. Staccato organ notes.

And a series of events would be produced having the shape in Figure 4.5.

Figure 4.5. Event shape of Figure 4.4.

Playing the identical tune on a guitar would produce the same pitches (Figure 4.6):

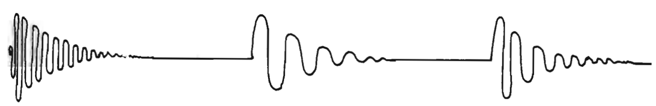


Figure 4.6. Staccato notes on guitar.

But once a guitar's strong vibration trails off, the events would have an altogether different shape (Figure 4.7).

Figure 4.7. Event shape of Figure 4.6.

Envelopes

The shape of an event is called its *envelope* or contour. In the example given above, the notes played on the organ have a different envelope than the same notes played on the guitar. Figure 4.8 shows some other possible envelopes:

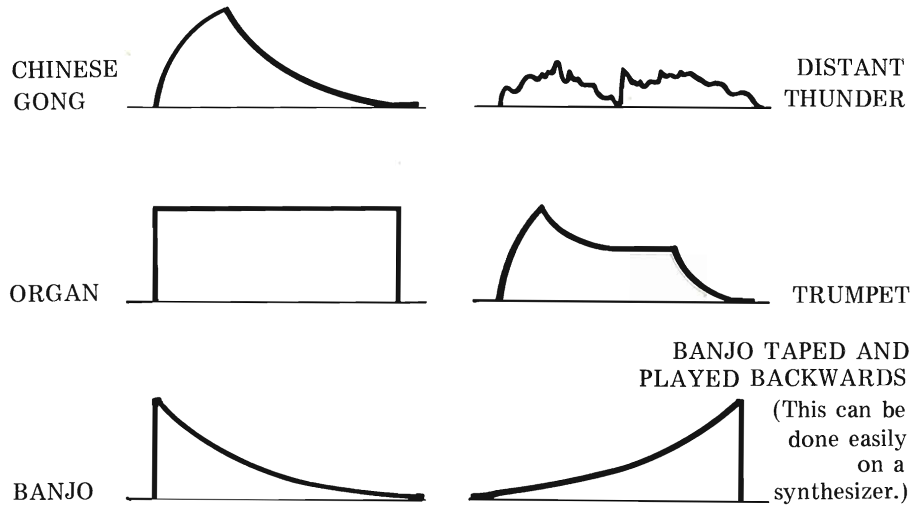


Figure 4.8. Six possible envelopes.

Any low-frequency waveform may be used to produce events and to give them a shape. Usually, however, a synthesizer will have one or more devices designed specifically to generate low-frequency waveforms suitable for giving events a shape. These devices are called *envelope generators.* The output of an envelope generator is aperiodic; instead of appearing over and over again, as a low-frequency periodic waveform would, the low-frequency waveform from an envelope generator appears only when the envelope generator is triggered (for instance, by striking a note on the keyboard).

Questions

1. Honking a car horn would produce an event shaped like Figure 4.5. Draw the shapes or envelopes of the following: a piano note (first a high note, then a low note), a tuba, a drum roll, a harp, a cymbal crash, and surf.

2. If you use a tape recorder to record a clock which ticks once every second and then speed up the tape 100 times faster, what would you hear?

3. If you record a bassoon playing low A-110 Hz, and then slow the tape recorder down 100 times, what would you hear?

You will recall that at the beginning of Part I, two ideas were presented: the first, that acoustical waveforms can be generated and modified by purely electronic means. Based on what you've read here concerning waveforms, amplitude and frequency, timbre, harmonics, and filtering, you are beginning to have a basic understanding of how this is possible. However, the second idea presented at the beginning of this section — that sound-generating and sound-modifying equipment can be controlled electronically — has been discussed only in indirect ways. With the discussion of the envelope generator, we have an excellent example of *voltage control* which now can be discussed in more detail.

Envelope Generators

An envelope generator is generally used to create a signal which can control the Voltage Controlled Filter or the Voltage Controlled Amplifier on a synthesizer, thereby giving the sound produced by the synthesizer the desired shape or "envelope." Let us consider the most common use of the envelope generator—controlling the filter. The voltage controlled filter (abbreviated VCF) on a synthesizer can, of course, be opened and closed manually. (The terms "open" and "closed" refer to raising or lowering the cutoff point of the filter to let more or fewer harmonics pass through).. However, in order to synthesize most natural instrumental sounds—or even more broadly, most musical sounds—a very sophisticated, programmable way of automatically opening and closing the filter is required. This is necessary because the harmonic components of most instrumental tones are not constant from the beginning of the sound to the end of the sound. For example, as you blow into any wind instrument, a certain amount of time is required simply to start the tone. This is called *attack time*. Consider the relatively slow attack time of a tuba for example. This would contrast with the fairly fast attack of a guitar, which sounds a tone almost instantaneously when the strings are struck by the guitarist's pick. During the attack of a tuba, for instance, the sound builds gradually during about 1/5 second after you start blowing. Not only does the sound get louder during this period, but it also gets richer in harmonics and consequently brighter in sound. During the attack of a guitar, all the harmonics seem to sound from the instant the string is plucked, and from that point on they *decay*, the highest harmonics disappearing first.

The envelope generator is hooked up to the filter in a synthesizer so that the attack and decay characteristics of any sound can be programmed on the envelope generator's controls, and the filter will then be "opened" or "closed" automatically by the envelope generator. To synthesize a tuba sound, for instance, we would want to set the controls on the envelope generator so that the sound would build up during the first 1/5 second. When the envelope generator is triggered by pressing a note on the keyboard, the envelope generator produces a voltage which automatically opens the filter during the first 1/5 second. The result is that as the filter opens, it lets more harmonics pass through and creates an attack which sounds very much like the attack on a tuba. You can also simulate the faster attack time of the guitar, bringing all of the harmonics that create the distinctive timbre of the guitar almost immediately. This is done by simply decreasing the attack time so that the filter opens more quickly, thus permitting all of the necessary harmonics to sound very quickly as each note is played.

The point here is that the low-frequency, aperiodic waveform that the envelope generator provides is a *voltage*. This voltage opens the filter for you, in a more precise manner than any type of manual control could ever provide. By adjusting the *attack*, *decay*, *sustain*, and *release* times, this voltage will open, hold open, and then close the filter in a manner characteristic of the sound you are synthesizing. This voltage control of the filter provides programmable control over the timbre of the waveform from the instant that the tone begins to sound to the final moment before it dies out.

All synthesizers must have envelope generators. There are different types of envelope generators. The Odyssey, for instance, has two envelope generators. One of these has four controls, for Attack, Decay, Sustain, and Release, and is called an ADSR type of envelope generator. (See Figure 5.1). The second envelope generator on the ARP Odyssey has only two controls, Attack and Release. This simpler type of envelope generator is called an AR envelope generator. (See Figure 5.2).

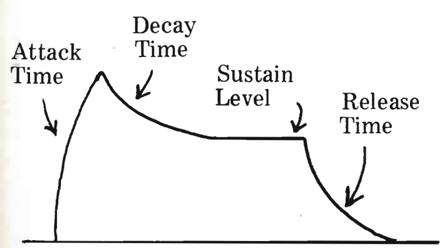


Figure 5.1. Four adjustable parameters on an ADSR envelope generator

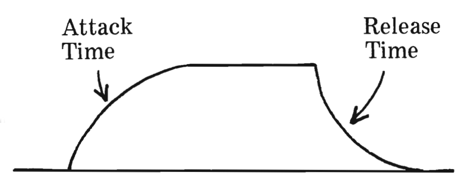


Figure 5.2. Two adjustable parameters on an AR envelope generator

Other Voltage Controllers

The envelope generator is only one example of voltage control as it is employed in synthesizers. You'll notice that each of the *oscillators* (the waveform generators, or sound sources, of the Odyssey), the *filter*, and the *amplifier* of the Odyssey are marked "voltage-controlled." The keyboard is also a voltage controller, providing a voltage that is applied to the oscillator (s) to determine the frequency of the pitch that you'll hear. Certainly you could move the frequency-control slider of the oscillator manually to change its pitch — but could you move it quickly and accurately enough to play even a moderately difficult piece of music at a reasonable tempo, or speed? The answer, of course, is no. Without the voltage control that the keyboard provides, it would be difficult to play even the simplest tune without having some alternative to moving the frequency-control slider up and down by hand. Voltage control, as you'll soon determine for yourself, is an extremely important extension of your ability to control the various functions of the synthesizer.

The whole point of making voltage-controlled circuits on a synthesizer is so that the synthesizer's electronic circuits can "talk" to each other in a language they understand and can respond to — voltages. A human being can turn a knob or flick a switch, but an electronic circuit must use voltages to control other electronic circuits. For instance, let's say that you are listening to a tone generated by an oscillator. If you want to change the pitch of that tone, you can physically move the control labeled "frequency." But if you want to change the pitch of the oscillator by playing on the keyboard, how does the keyboard tell the oscillator to change its pitch? The answer, of course, is that the keyboard produces a voltage which is fed into the oscillator and tells the oscillator to change its pitch. Because the oscillator is designed to react to control voltages from other circuits (like the keyboard), the oscillator is said to be a "Voltage-Controlled Oscillator."

From our discussion, an important concept has emerged — a concept so basic to an understanding of voltage control that it deserves emphasis: *in synthesizing almost any sound you will be employing two types of voltages: (1) an audio signal, and (2) a control voltage.* The audio signal provides the basis for the sound that you'll hear. This signal may be in the form of a sawtooth wave, or a sine wave, or a square wave, or a pulse wave. *Whatever the waveshape, this is the basic raw material for the sound that you will hear.* In short, it is an audio frequency — between approximately 20 Hz and 20 KHz.

Control Voltages

Unless you are creating an extremely simple effect, however, you will also be using control voltages. While you don't normally "hear" the control voltage, you may hear its *effect* upon the audio signal. For example, pressing down one key on the keyboard of the synthesizer and then pressing down a different key will cause the audio frequency the oscillator is producing to change, thus changing the pitch of the sound you're hearing. While you did not "hear" the control voltage that was sent to the oscillator to cause this change, you did hear its effect. Similarly, you may hear a particular instrumental sound that has a pleasant vibrato (a gentle, pulsating effect). In synthesizing this sound, you will actually hear the audio frequency produced by the oscillator; you will not hear the low-frequency sine wave creating the vibrato (this is the control voltage) but you will hear its effect. This concept will become increasingly clear as you begin using the voltage-controlled functions of the synthesizer presented in Part II of this text.

Frequency Modulation and Amplitude Modulation

One final basic concept must be presented before moving on to Part II and the actual operation of the ARP Odyssey. This concept is called "modulation." To modulate a waveform is to change it periodically, following the pattern of another waveform. If the change in the first waveform is a change in frequency, it is called *frequency modulation;* if the change is in amplitude, it is called *amplitude modulation.*

Here is a simple example of frequency modulation. Suppose we begin with a simple oscillation — a sine wave at the audible frequency of 100 Hz. Now suppose that we wanted the frequency (the pitch that you are hearing) of that sine wave to rise gradually to 200 Hz, then fall gradually to 100 Hz and begin this cycle again. To do this, we would frequency-modulate the 100 Hz square wave with

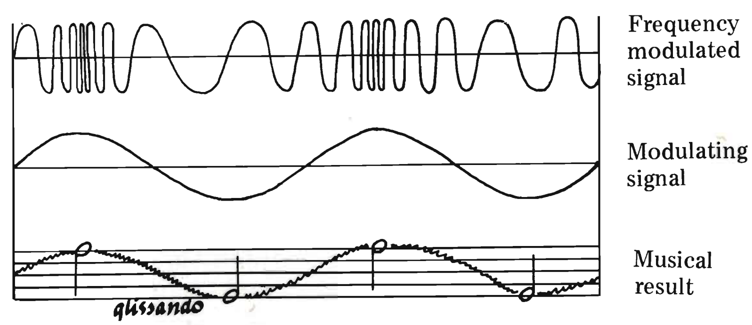


Figure 5.3. Frequency modulation.

a low-frequency sine wave — say, at the rate of 1 Hz. As a result, you would hear the pitch of the sine wave rise gradually from 100 Hz to 200 Hz once every second. Each time it reached 200 Hz, it would begin to fall to 100 Hz and repeat the cycle. A graph of the changes in frequency is shown in the Figure 5.3.

Amplitude modulation, on the other hand, would produce something like Figure 5.4.

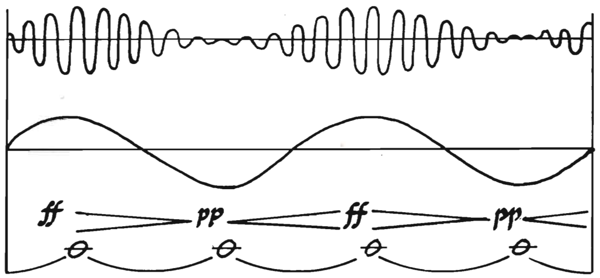


Figure 5.4. Amplitude modulation.

In each illustration you will see that what is happening does conform to the definition established earlier: the first waveform (the sine wave) is changing systematically following the pattern of another waveform (the 1 Hz sine wave).

Questions

1. If an envelope generator can be hooked up to a voltage-controlled filter to produce an attack and decay like a tuba but automatically opening and closing the filter, what would happen if that same envelope generator were hooked up to a voltage-controlled oscillator that was producing a pitch?
2. Explain the concept of "voltage control." How can voltage control be more effective than manual control?
3. If you play a trill on a clarinet, are you modulating the frequency or the amplitude of the waveform? What is the shape of the modulating signal?
4. If you sing a note while patting your mouth with your hand, what kind of modulation do you get?
5. A modern electronic police siren is actually a voltage-controlled oscillator hooked up to a loudspeaker. A siren is an example of what kind of modulation? What do you think the control signal used to create this modulation looks like?

PART II

Section 1: Basic Operational Features

Having now completed Part I, you should have acquired a basic understanding of generating and modifying sounds by purely electronic means. This section is designed to guide you in applying this knowledge in a practical manner through the actual operation of the ARP Odyssey synthesizer.

Block Diagrams and Patches

To begin, study the diagram shown in Figure 1.1.1. The control panel of the Odyssey is organized much like an assembly line. Generally speaking, the sounds you will create will be assembled beginning

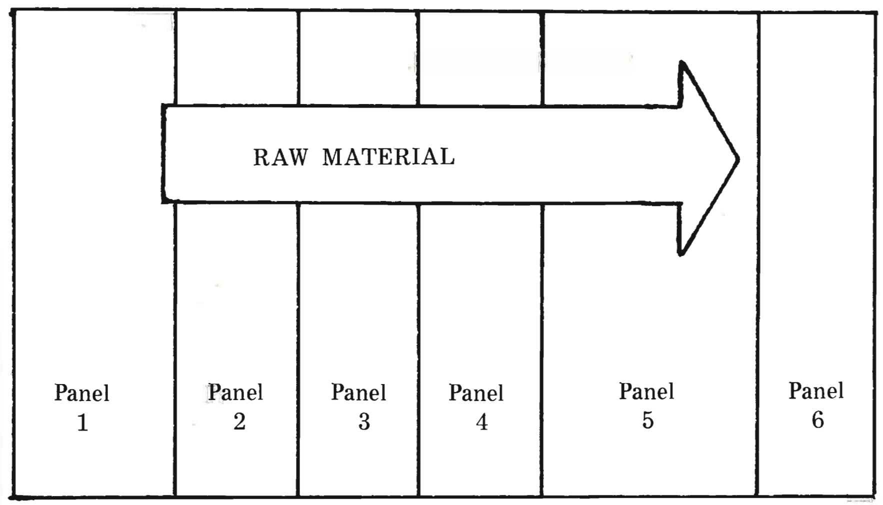


Figure 1.1.1. Control panel organization.

with the raw material — the oscillators and/or noise generator — located on the left side of the panel. Your "product" will then move from the left to the right side of the instrument, passing through the various stages of the assembly line that are required to shape the raw material into the finished sound you are seeking. You will observe this left-to-right movement as you begin experimenting with the Odyssey. Similarly, the method used to record visually a signal's path across the instrument reads left-to-right. These visual representations of the assembly line are called *block diagrams.* Basic block diagrams for a trumpet sound, as produced by a trumpet and by the Odyssey, are shown in Figure 1.1.2. Notice that the audible signal (waveform) path

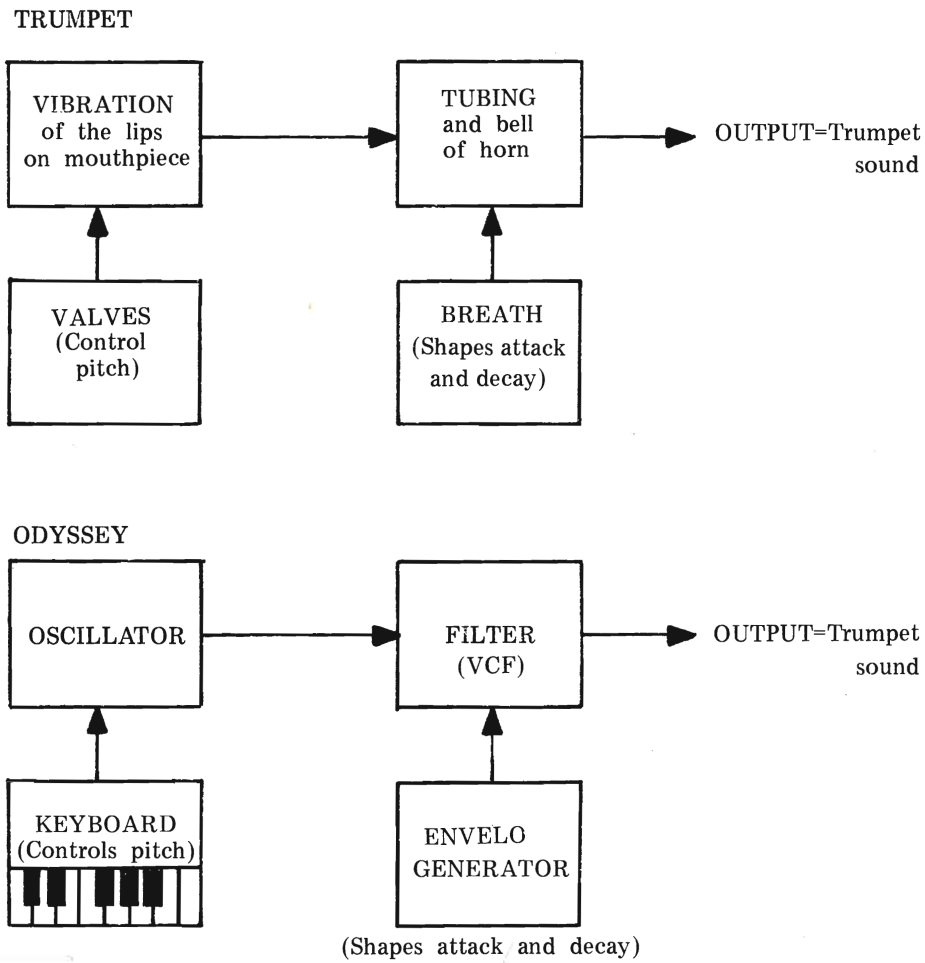


Figure 1.1.2. Block diagrams for conventional trumpet and Odyssey trumpet sound.

runs horizontally, left to right. The control paths run vertically — as, for example, in the case of the keyboard controlling the pitch of the Odyssey's oscillator(s), or in the use of the envelope generator to control the shape of the sound.

Within the Odyssey, of course, the gathering of the raw materials and forming of an assembly line is done electronically. The interconnection of the various functions is known as creating a *patch.* The block diagram shown in Figure 1.2 is a trumpet patch. In a similar manner, it would be possible to diagram any other patch that you might create. It is important that you begin thinking in terms of this sequential left-to-right organization of the Odyssey's controls. Not only will it facilitate working with the synthesizer, but it will also help you in remembering the essential elements of the patches you will be creating. You'll find this useful in freeing yourself from having to record the control positions for every sound you want to remember. Instead, seek to *understand* what functions are being employed — the oscillator(s), the filter(s), the various controllers — so that you can reassemble the sound conceptually rather than by rote.

The Control Panel

In Figure 1.1.3 you can see that the diagramming on the control panel of the Odyssey is designed to assist you in mentally assembling the patches you'll be creating. As in the case of the block diagrams shown in Figure1.1.2 audio signal paths run horizontally, left to right. Control paths run vertically, entering the bottom side of the boxes which label the functions of the Odyssey.

Generally speaking, each function on the Odyssey is set aside in an area of its own, separated from the other sections of the instrument by a line running from the top to bottom of the control panel. We'll call these areas "panels," so that when we speak of the "second panel from the left" you'll understand that the general area being examined or discussed is the Voltage Controlled Oscillator No. 1 panel.

Switches

Now let us examine the basic controls of the Odyssey more closely. Note the row of switches along the bottom of the control panel. These, and similar switches elsewhere on the panel, are two-position switches, like an off/on light switch. Some of these actually *are* off/on switches, while others (the majority of them) are *selector switches.* The selector switches let you choose between two interconnections, depending upon which position the switch is in. Figure1.1.4shows the first selector switch on Panel 5.

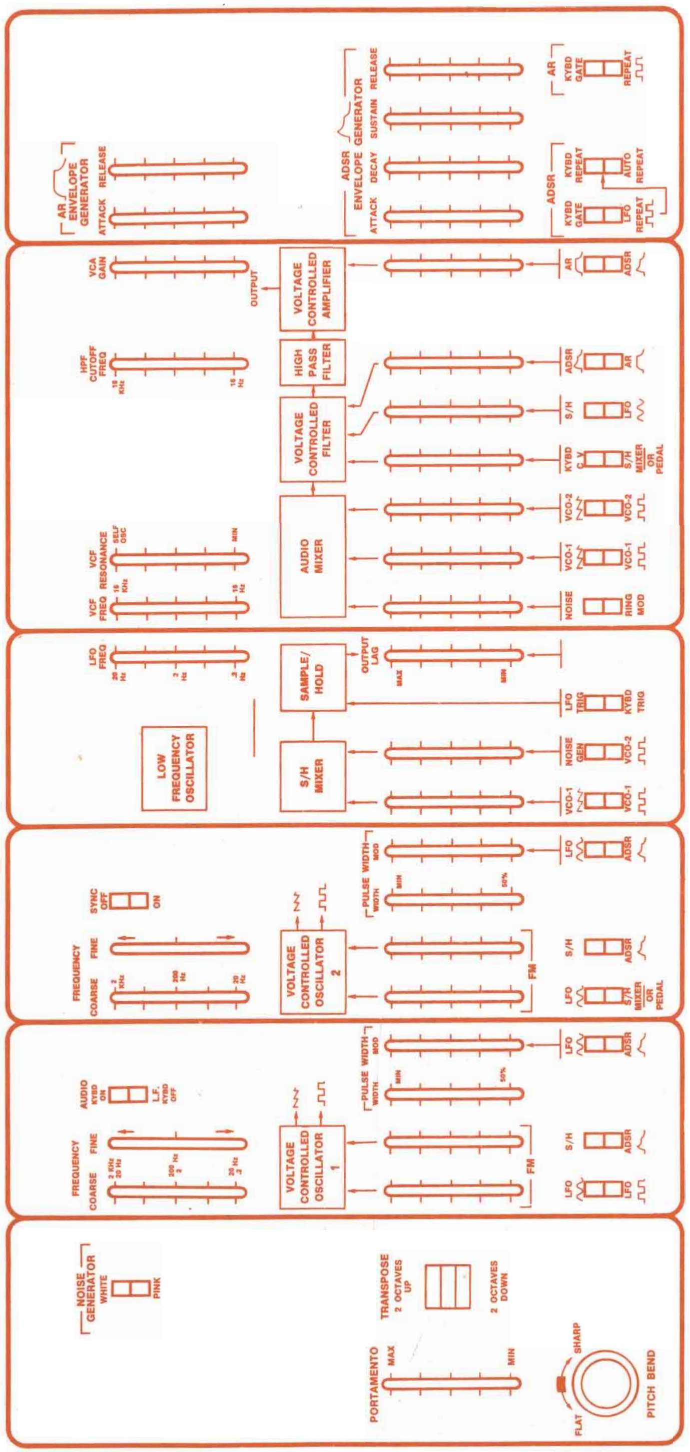


Figure 1.1.3. Odyssey control panel.

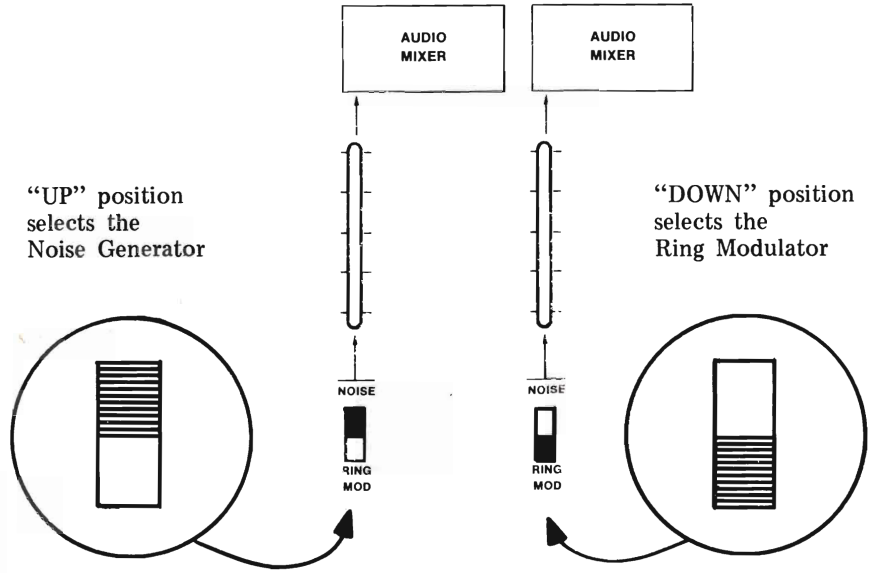


Figure 1.1.4. Selector switch.

Slide Controls and Attenuators

The Odyssey also has a number of slider-type controls. These controls may be divided into two groups: (1) *slide controls,* which govern some *function,* as in the case of the VCO frequency controls, and (2) *attenuators,* which act much like faucets, controlling the *amount* of signal that is allowed to flow along the particular signal or control path you are using (Figure 1.1.5).

The controls on the Odyssey are color coded to associate them with the appropriate circuits that they control or are connected to. The color code, by function, follows:

VCO-1	Blue
VCO-2	Green
LFO	Pink
Noise	White
S/H	Yellow
ADSR & AR	Red
All others	Black

The controls which function as input attenuators (that includes all the controls along the bottom row except the two pulse-width and S/H lag controls on the VCOs and the ADSR) are color coded according to the function connected to each slider when the selector switch beneath the slider is in the "up" position.

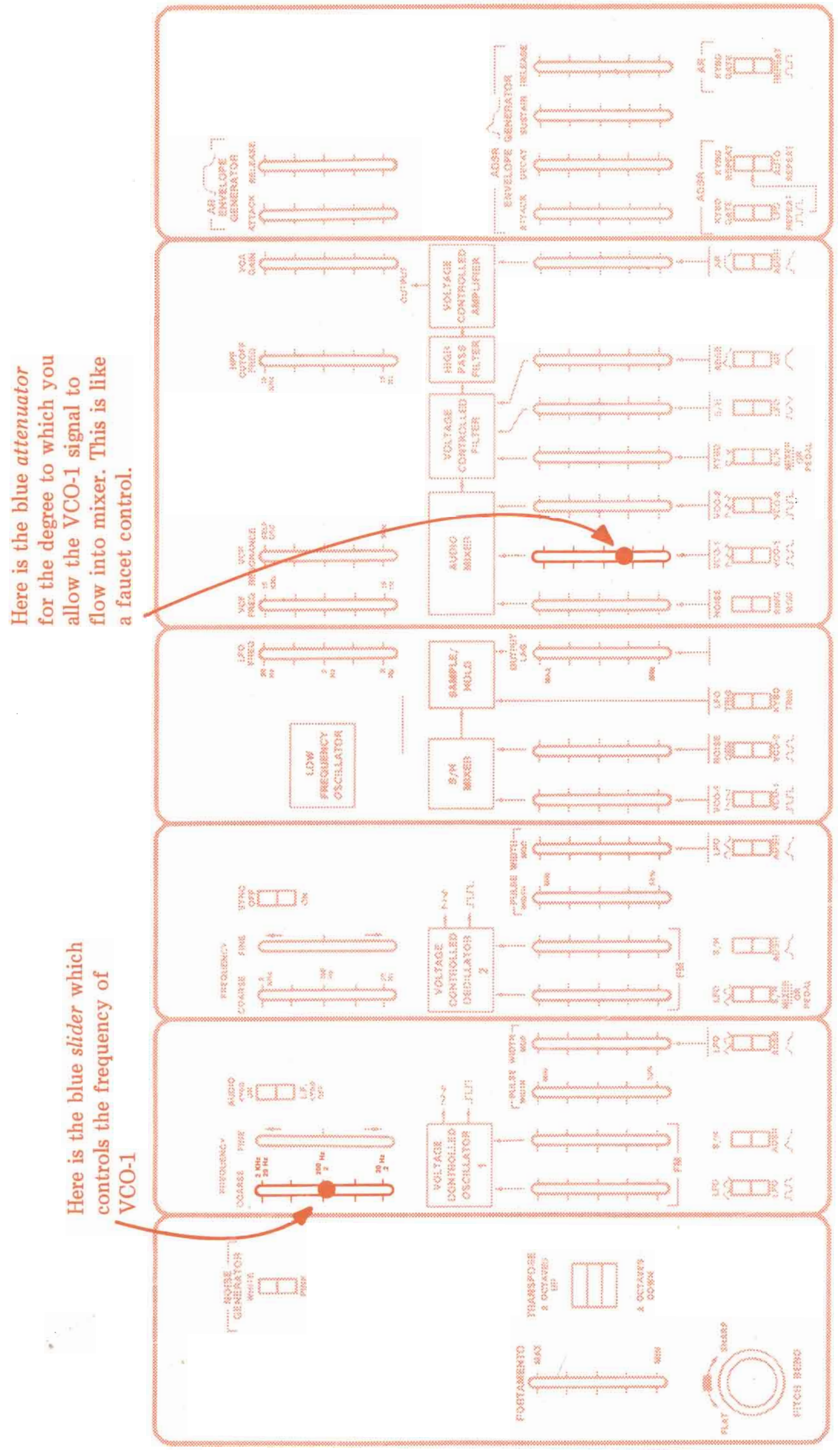
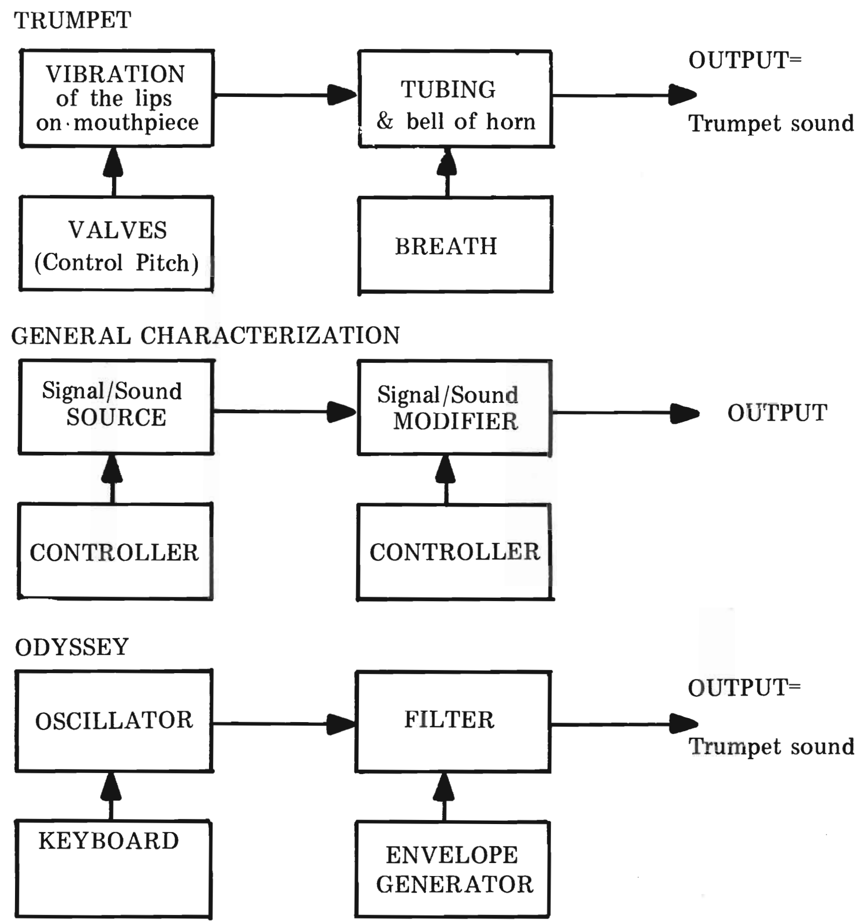


Figure 1.1.5. Slide controls.

You'll find that most slide controls along the bottom row are *attenuators* except for the ADSR, S/H lag, and pulse-width controls. All of the controls along the upper row are simply slide controls governing a particular function.

As you begin to experiment with the switches and sliders in connecting the various components of the Odyssey, you'll quickly realize that the number of patches is actually infinite. Each full patch, however, requires that you employ at least three functions of the Odyssey: (1) a *signal source*, (2) a *signal modifier*, and (3) a *controller*. The block diagram in Figure 1.1.6 again illustrates the trumpet/Odyssey comparison shown earlier (Figure 1.1.2). Note the *signal source*, *modifier*, and *controllers*. The following material will deal with each of these three components — sources, modifiers, and controllers — in detail, allowing you ample opportunity to explore on your own.

Figure 1.1.6. Simple block diagrams.



Section 2: Signal Sources

On the Odyssey, audio signals in their raw form are generated by the *noise generator* and the two *voltage-controlled oscillators* — VCO-1 and VCO-2. The raw signals produced by these signal sources are later processed by other circuits to produce a controllable audio output. Let's examine one of these signal sources now.

Lesson 1: Noise Generator

Using Figure 2.1.1 as a model, set all the controls on your Odyssey as shown. Notice that all of the two-position switches are in the "up" position. The general setup in Figure 2.1.1 will give you a basic patch from which to work. Each time an experiment is completed, return all the controls to this position and you'll be ready to begin again.

Now, turn on the Odyssey (red switch in the upper right-hand corner) and locate the Noise Generator switch in the upper part of the panel to the extreme left (see Figure 2.1.2). Check to make sure the switch is in the "up" position, labeled "White." The first attenuator we will use is the white one labeled "Noise," located in the Audio Mixer section (the second panel from the right in Figure 2.1.2). Remember, the attenuators are like faucets, setting the amount of a signal that is allowed to pass through. As you push the attenuator higher, the sound becomes stronger. Try it. Experiment by moving the attenuator up and down through its entire range of travel.

Noise can be used to synthesize many of the everyday sounds we hear — surf, thunder, rain, motors, and even wind. For a moment, think about the characteristics of the sound of an ocean wave — how it gradually begins, building to a climax with a culminating crash of water, then falling back and fading. Let's attempt to reproduce this sound.

Experiment 1: Noise

Set the Noise Generator switch on "White" and slowly raise the white attenuator, gradually making the "wave" stronger. About halfway up, move the attenuator rapidly to the top, and then bring it back down at a steady rate. Did you create a surf effect? Figure 2.1.4 provides a graphic description of what you accomplished.

White Noise and Pink Noise

You may have asked yourself, "What kind of waveform does the Noise Generator produce?" To answer this question, let's return to the earlier discussion of waveforms, in which they were classified under

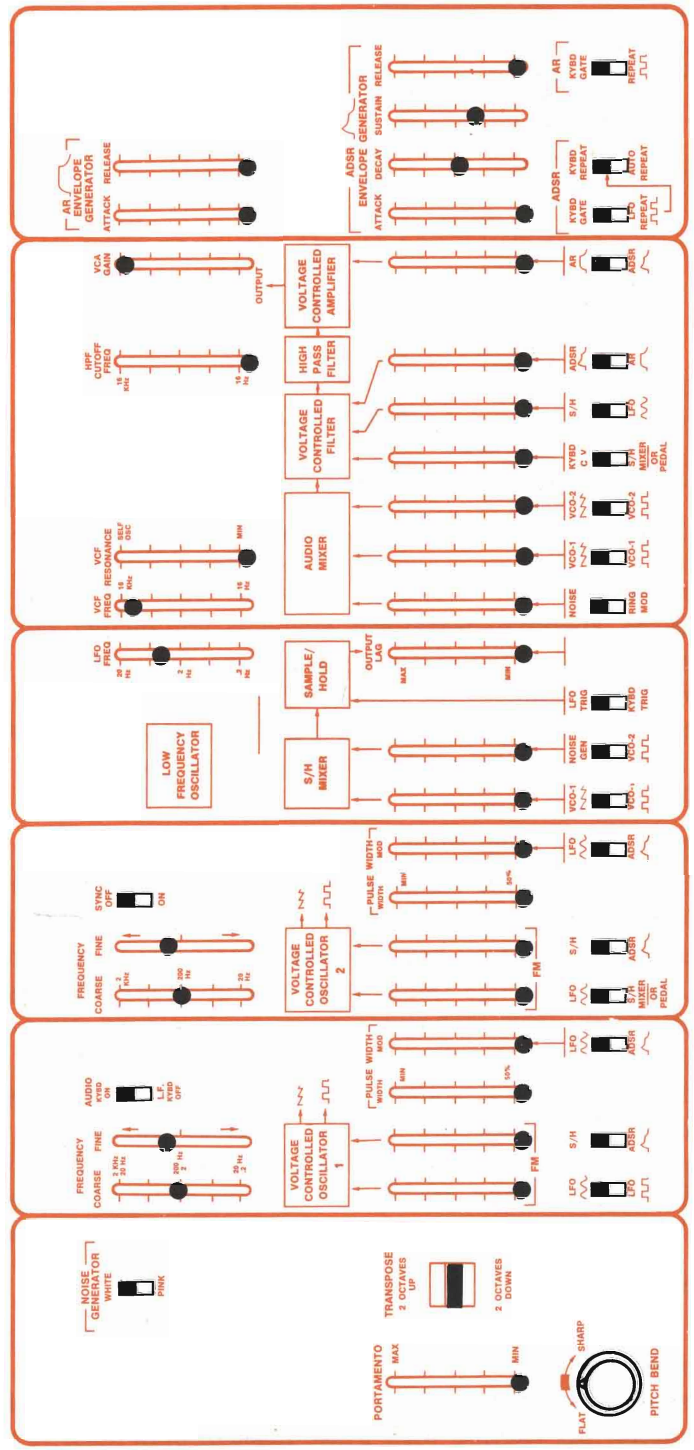
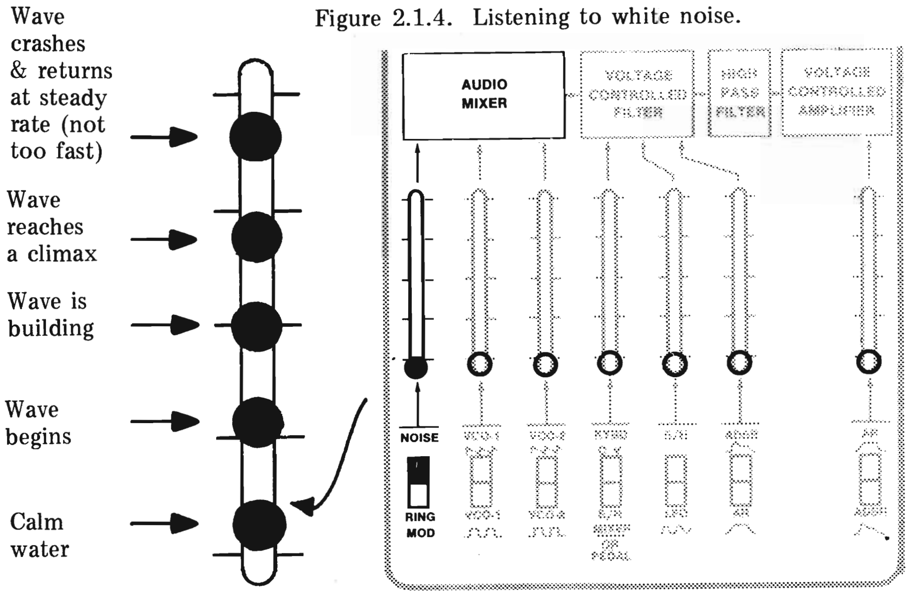


Figure 2.1.1. Control settings for basic patch.

Figure 2.1.4. Listening to white noise.



PANEL 5

two general headings — periodic and aperiodic. The Noise Generator produces an aperiodic waveform which is completely random. You can think of white noise as a waveform in which the chances of finding any particular frequency are equal to the chances of finding any other frequency. This sort of waveform is called *white noise* by analogy to white light, because it contains *all frequencies* just as white light contains all colors.

Experiment 2: Filtering Noise

Return to the Odyssey and move the Noise Generator switch from White to Pink. Listen to the sound carefully. How does it differ from white noise? As you may have guessed, *pink noise* is created by filtering out some of the high frequencies of white noise, thereby allowing the lower frequencies to become more predominant. Human hearing tends to give undue prominence to the higher frequencies in the white noise signal, so that it sounds like steam escaping from a radiator. By filtering white noise, we achieve an effect (pink noise) that has a frequency content in which we *hear* all frequencies, the lows as well as the highs. Pink noise can be said to have equal energy per octave. Don't hesitate to experiment with both white and pink noise.

Questions

1. If there were such a thing as "red" noise, what do you think it would sound like?

2. Name as many sounds as you can that are based on noise. Would you be better off starting with white noise or pink noise to make each of these sounds?

Figure 2.1.3. Block diagram of Figure 2.1.2.

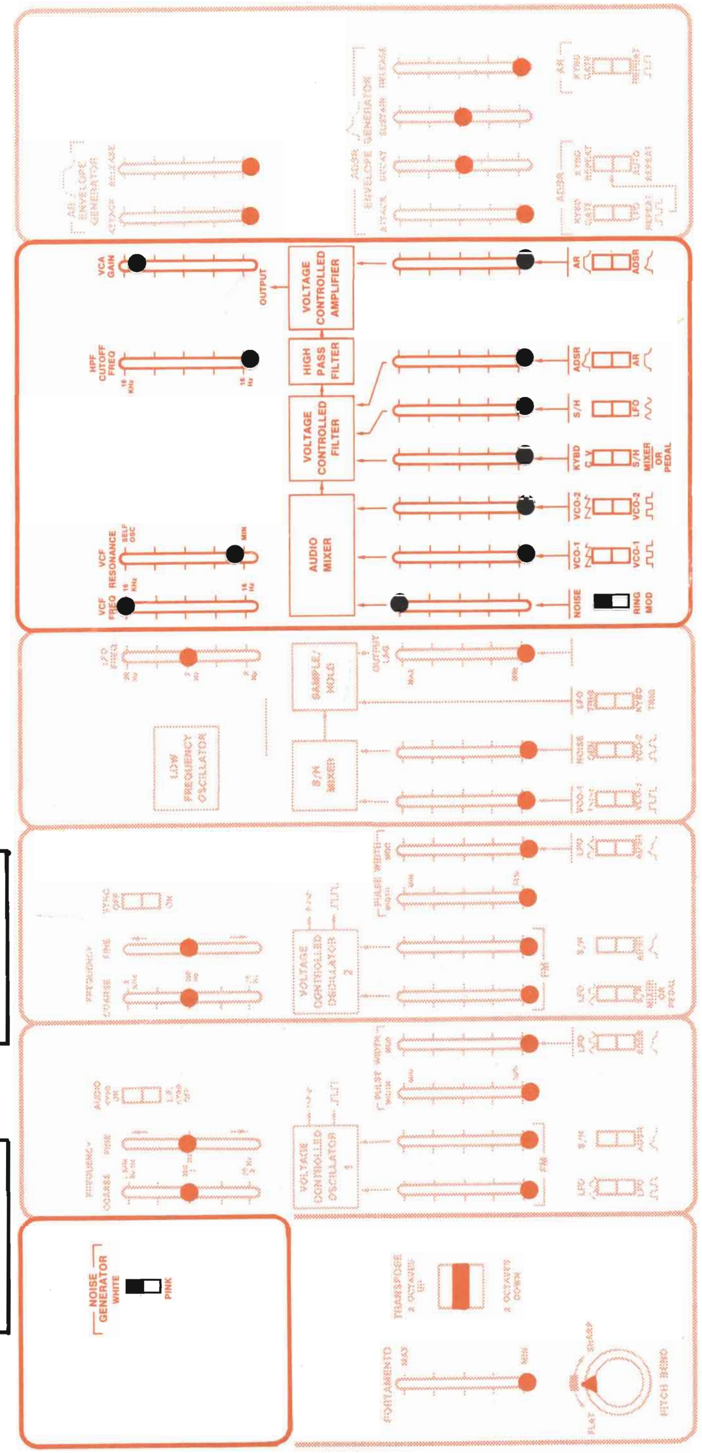


Figure 2.1.2. Control settings for Noise Generator experiments.

Lesson 2: Audio Oscillators

Moving from the noise generator with its aperiodic waveforms, let's now examine two signal sources which generate periodic waveforms. These signal generators are called oscillators. Your Odyssey includes two oscillators labeled "Voltage Controlled Oscillator No. 1" (VCO-1) and "Voltage Controlled Oscillator No. 2" (VCO-2). Focus your attention on VCO-1 (second panel from the left, Figure 2.2.1).

VCO-1

Experiment 1: Waveforms

VCO-1 generates three kinds of waveforms: square, pulse, and sawtooth. The selector switch below the blue VCO-1 attenuator on the Audio Mixer section of the control panel chooses which VCO-1 waveform will be heard. This switch should be on the sawtooth wave, as per the general patch. Gradually open this attenuator until you reach the top, and then close it in the same manner. Next, move the switch down to the square-wave setting and raise and lower the attenuator as you did before. Can you hear the difference in the two waveforms? Raise the attenuator again, and we'll look at some other controls that influence the sound of VCO-1.

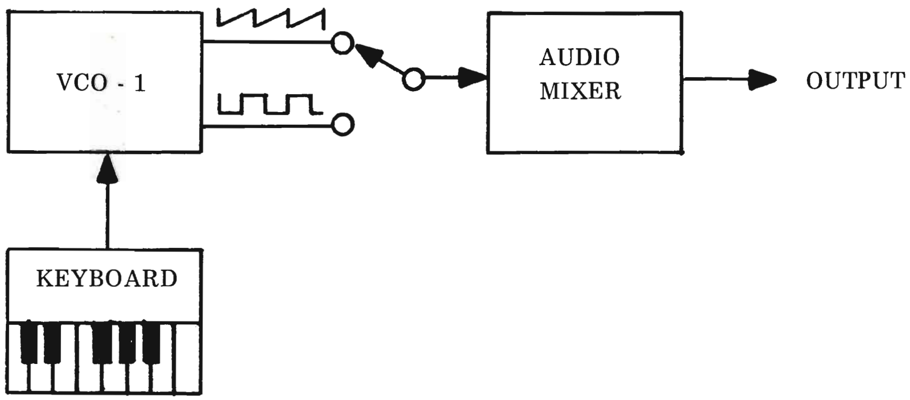


Figure 2.2.2. Block diagram of Figure 2.2.1.

Note: Always remember to return to the general patch shown in Figure 2.1.1 before starting any new material.

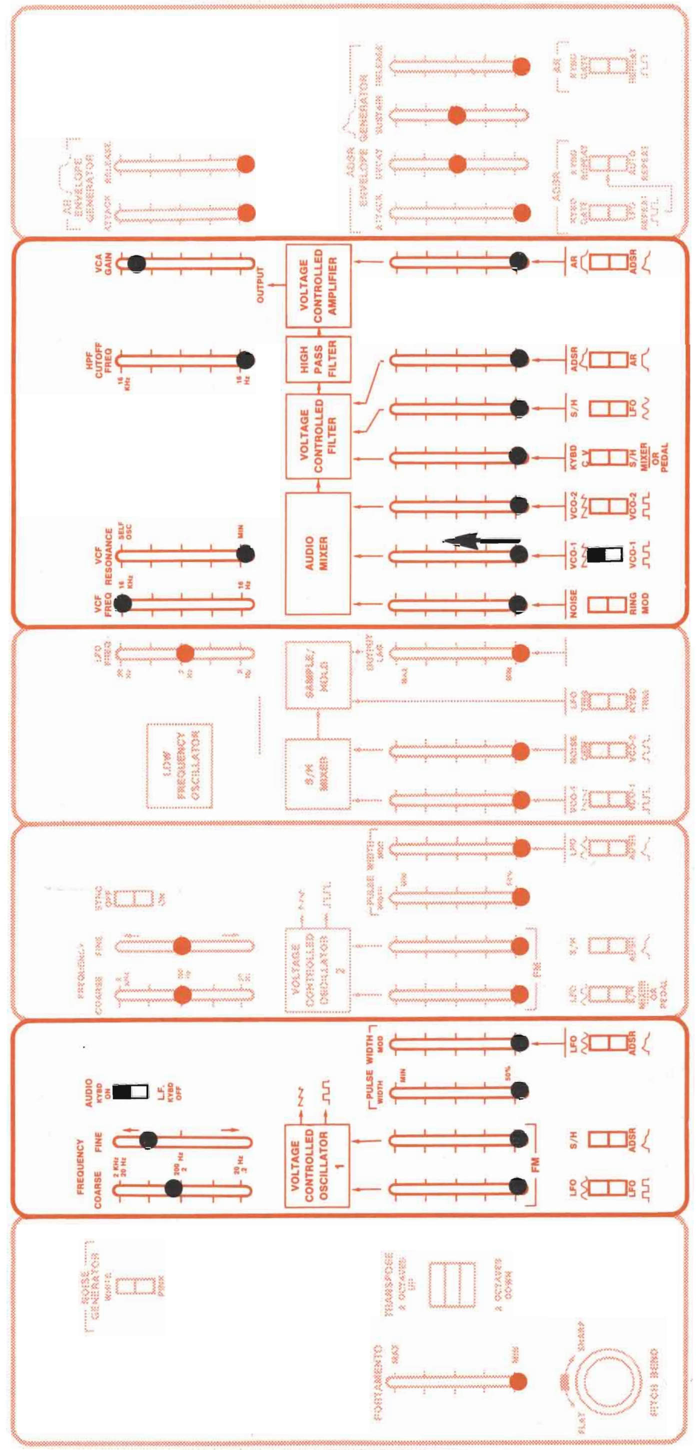
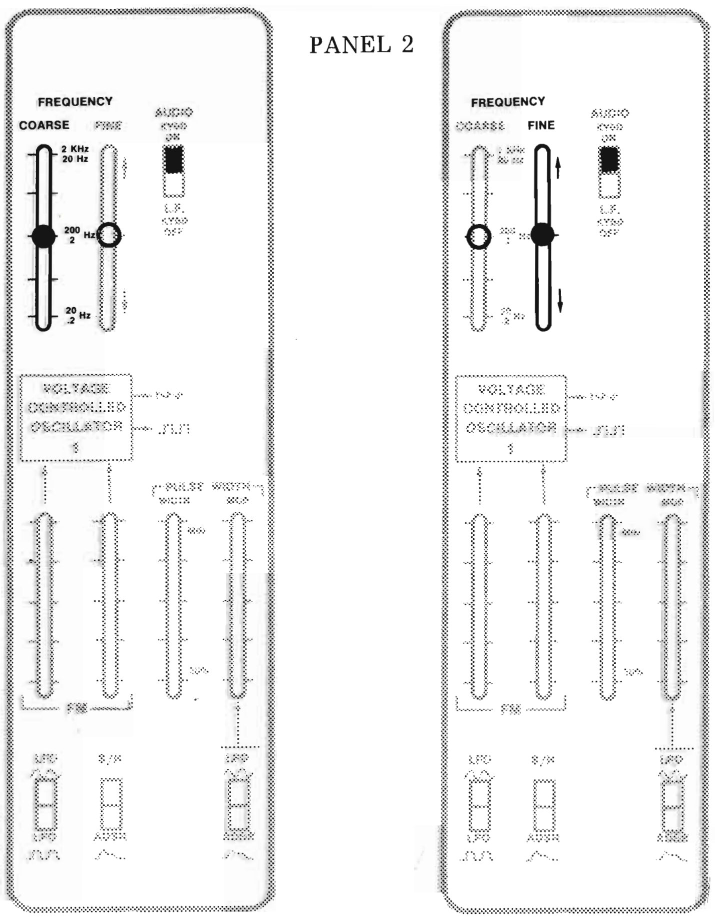


Figure 2.2.1. Sawtooth waveform patch (VCO-1).

Experiment 2: Tuning Controls (VCO-1)

The tuning controls (blue) for VCO-1 are found in the second panel from the left under the work "Frequency." Study Figure 2.2.3. Now, experiment with the coarse-tuning slider. What happens when you raise and lower the slider? Listen carefully while you move the slider through its entire range. Turn your attention to the fine-tuning slider and go through the same steps. Do the sliders have the same basic function? If so, which of the sliders is more sensitive? The answer to these questions is found in the following rule: As you raise the tuning sliders, the pitch (frequency) goes up; lowering the sliders brings the pitch (frequency) down. And "coarse" tune is the most sensitive control.

COARSE TUNING | PANEL 2 | FINE TUNING



How to tune VCO-1. Set the "Fine" control at center. Tune to approximately the desired pitch with the "Coarse" control. Then tune exactly with the "Fine" control.

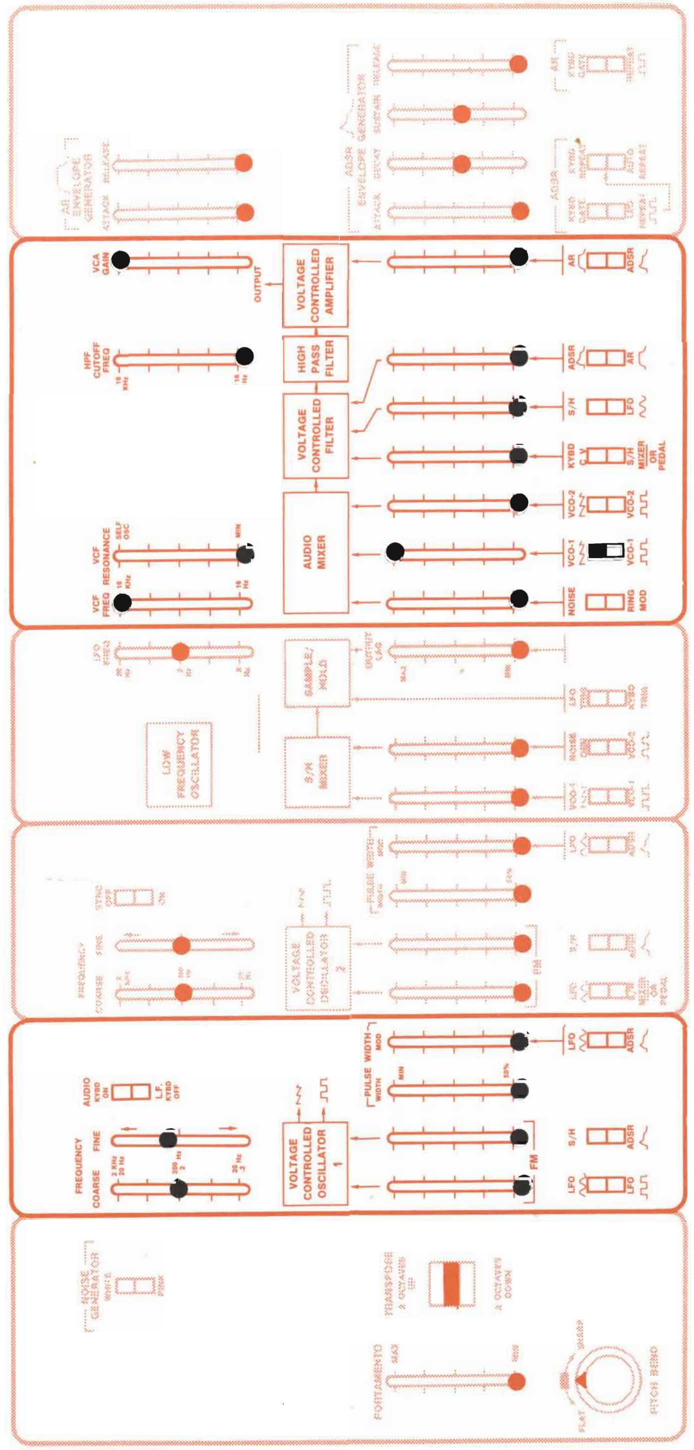


Figure 2.2.3. Tuning (VCO-1).

Experiment 3: LF Range

Move the Audio switch shown in Figure 2.2.5 to the down position (LF KYBD OFF). Move the blue tuning slider all the way up — you'll hear the clicks become faster. The reason we hear separate clicks/events in the low-frequency range is due, of course, to the limited range of frequencies humans perceive as having pitch. We only perceive physical vibrations as pitch between 20 Hz and 20 KHz. Later on, you will learn how to employ the LF range of VCO-1 as a control device. For now, return the selector switch to the Audio Keyboard position (up).

Figure 2.2.4. Block diagram for Figure 2.2.5.

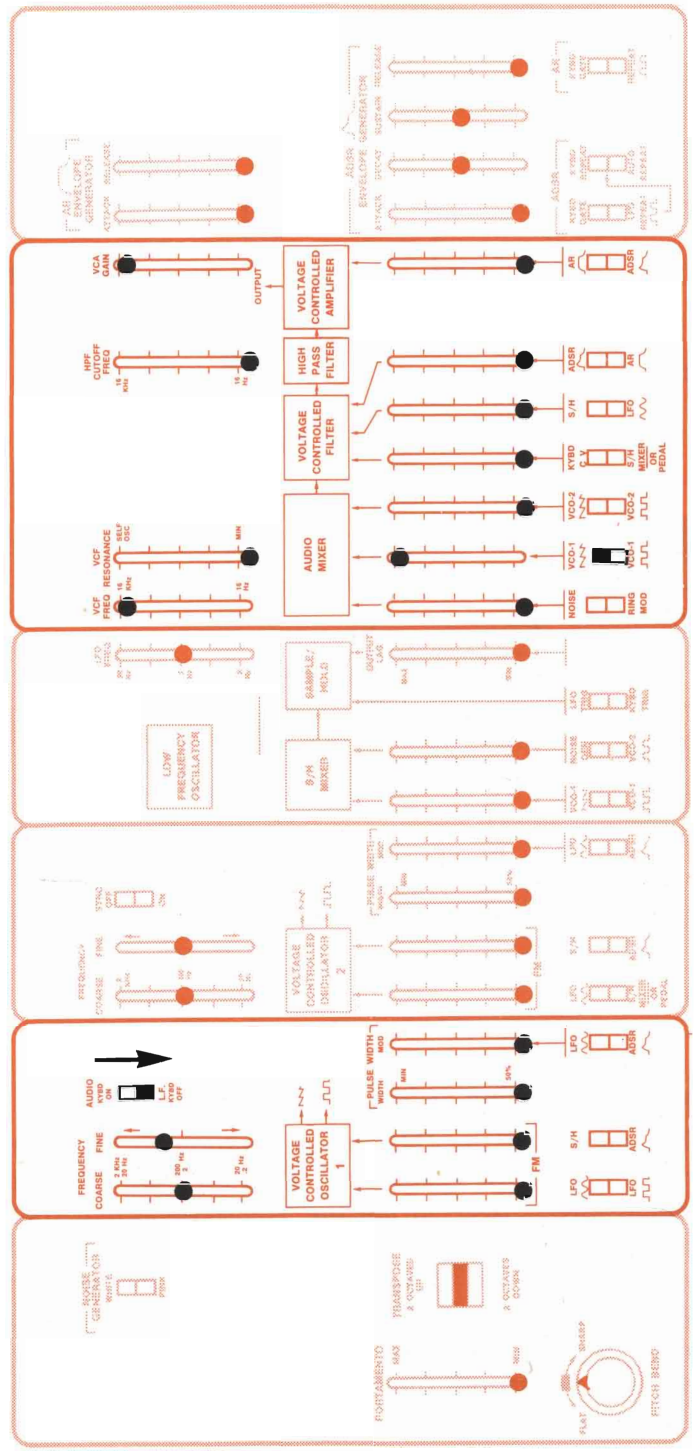


Figure 2.2.5. Low-frequency events.

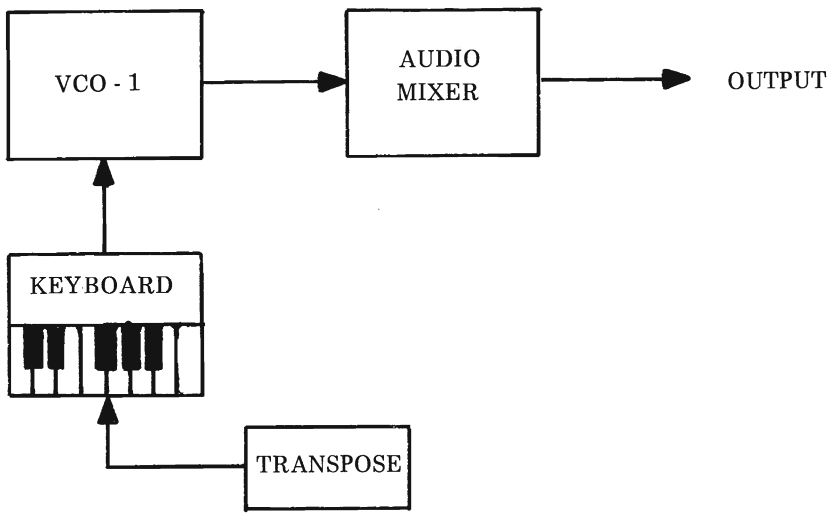


Figure 2.2.6. Block diagram of Figure 2.2.7.

Experiment 4: Transpose Switch and Keyboard

Set the controls of the Odyssey to conform to the general patch in Figure 2.1.1 with one exception — raise the blue VCO-1 attenuator in the Audio Mixer. Also, locate the Transpose switch found in the first panel on the left. Make sure the switch is in the middle position as shown in Figure 2.2.7. Now we'll be using the keyboard of the Odyssey for the first time. This keyboard looks like a section of the keyboard of a piano. A more technical explanation of how the keyboard controls the various functions of the Odyssey will be presented later in this section.

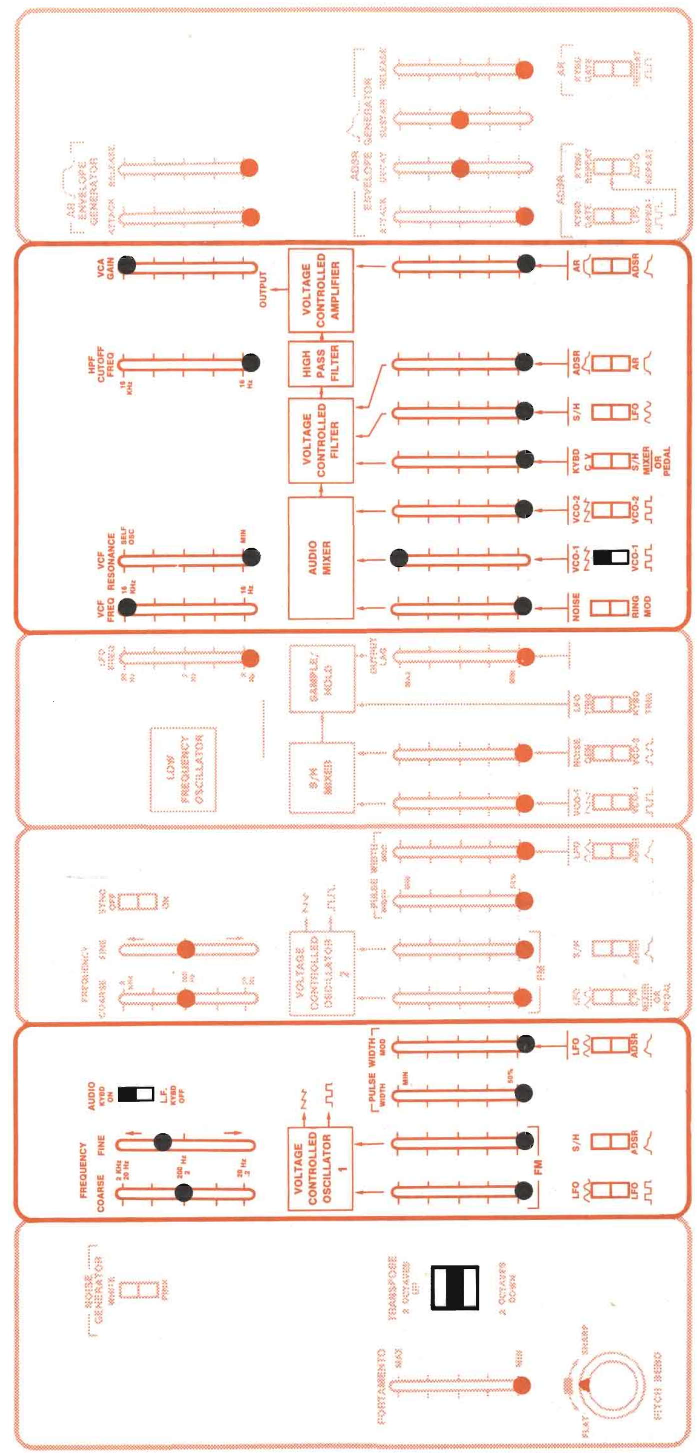
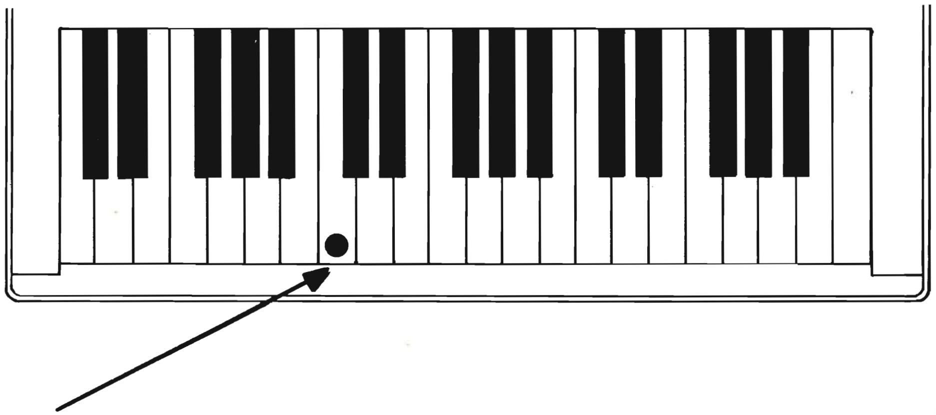


Figure 2.2.7. Transpose switch experiments.



Hold down "C" and try all three positions of the Transpose switch.

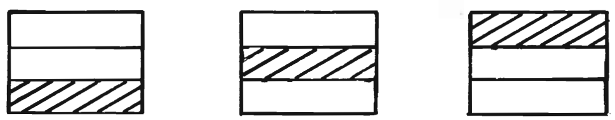


Figure 2.2.8. Transpose switch positions.

Press and hold down one key on the keyboard, as shown in Figure 2.2.8. While holding the key down,move the Transpose switch to the "down" position. What happened to the pitch? Do the same thing, only move the Transpose switch to the "up" position. Did you again perceive a change in pitch? Figure 2.2.9 uses musical notation to show what happened. As you can see, the Transpose switch shifts the entire keyboard control up or down by two octaves. In this way the keyboard range is extended to a full seven octaves.

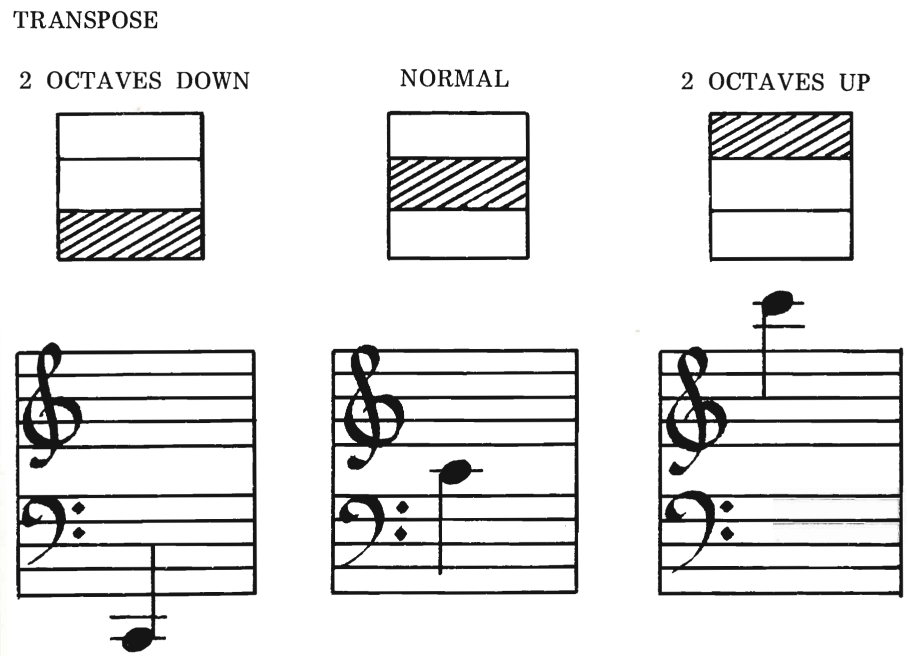


Figure 2.2.9. Transpose switch musical effect.

Next, press individual keys on the keyboard. Listen to what happens as you play keys going to the right, and then as you play keys going to the left. To the left, the pitch goes down. To the right, the pitch goes up.

The keyboard is actually an extremely precise voltage controller which determines the pitch of the oscillators by providing a control signal which is fed to the oscillators. The keyboard control voltage will be discussed in greater detail in this section under "Controllers."

Lesson 3: Pulse Width

What Pulse Width Is

A pulse wave is a waveform that has only rectangular corners. A pulse wave can look like this (a) like this (b) or like this (c) . A pulse wave is described by the relative widths of the high and low portions of the waveform. If, as in the first example, the pulse wave is in the "high" part of its cycle for only a very short time, it is said to be a "narrow pulse wave" and will have a very bright, nasal sound. A wider pulse wave, (b) above, has a fuller sound. Example (c) is a special case of the pulse wave — the square wave. It is called a square wave because the "high" part of the wave is exactly the same length as the "low" part of the wave. The square wave has its own distinct clarinetlike sound. In fact every different width of pulse wave has it's own unique timbre.

One interesting fact about pulse waves is that two pulse waves that are upside-down copies of one another will sound exactly the same. For instance, this pulse wave (d) . will sound exactly the same as this pulse wave (e) . Consequently, on a synthesizer it is only necessary to be able to create half of all the possible pulse widths, since the other half (where the high part is longer than the low part) sounds the same.

Both of the VCOs on the Odyssey are equipped with controls labeled "Width." When a "Width" control is in the "down" position, the pulse width is exactly 50%, making it a square wave. As the "Width" slider is raised, the pulse gets narrower and narrower until it looks like waveform (a).

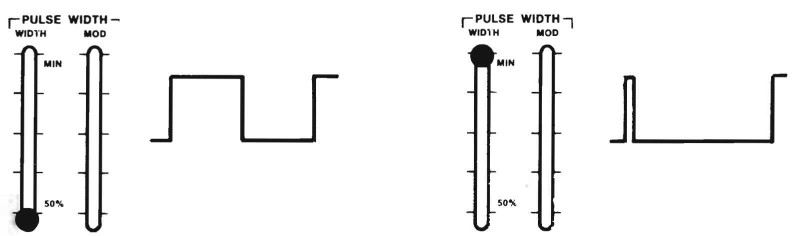


Figure 2.3.1. The pulse "width" control varies the shape of the pulse waveform.

Experiment 1: Changing Pulse Width

Set the controls as pictured in Figure 2.3.2; then raise and lower the blue Pulse-Width slider through its entire range. As the slider moves, listen carefully to the changes in timbre/tone color. Here is what happens when the Pulse-Width slider is raised from its lowest position:

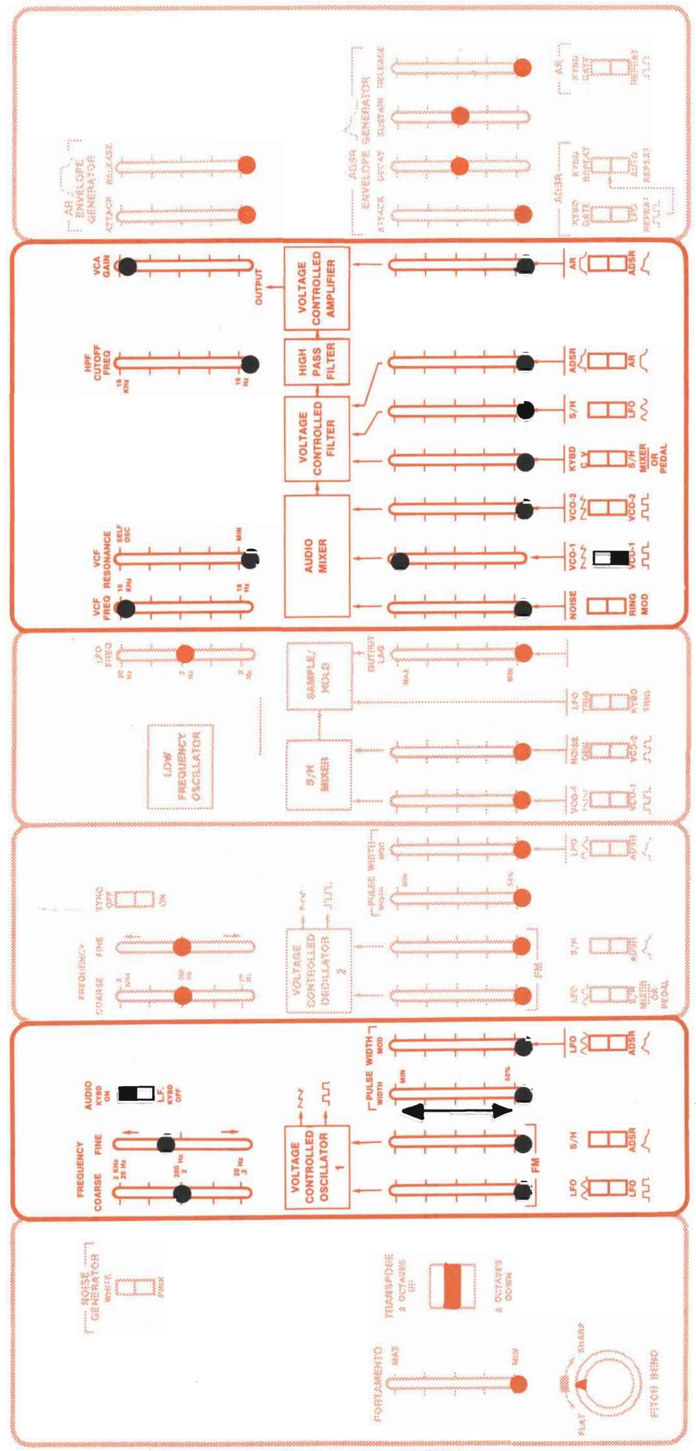
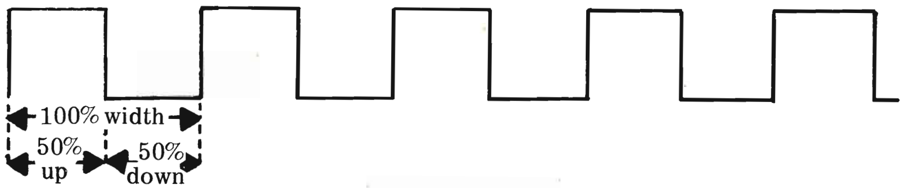


Figure 2.3.2. Basic patch for pulse-width experiments.

Figure 2.3.3. Diagrams for pulse waveforms at 50%, 25%, and 2%.

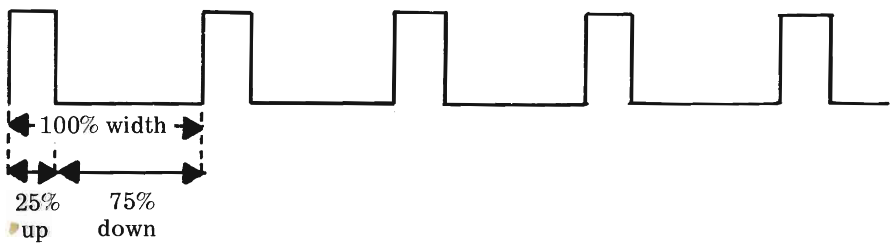
When the waveform
is even like this, it's
called a square wave.

A 50% Pulse Wave



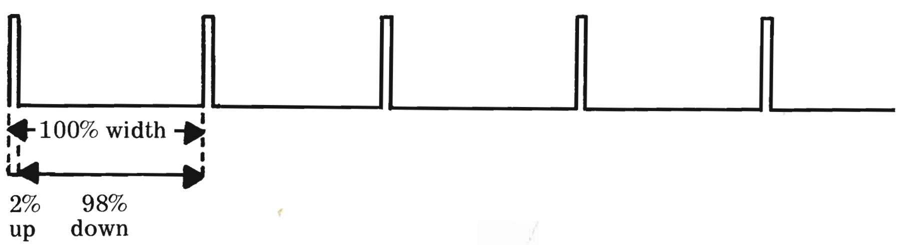
About halfway up,
the waveform looks
like this.

A 25% Pulse Wave



When the slider is
all the way up, the
waveform looks
like this.

A 2% Pulse Wave



Return the slider to the 50% position before continuing.

Experiment 2: Pulse-Width Modulation

Pulse-width modulation simply means changing the pulse width, either manually as you did in Experiment 1, or by using a control voltage, as we will now do. Locate the pink Pulse-Width Modulation slider shown in Figure 2.3.4, and set the controls accordingly. With the pink Pulse-Width Modulation slider raised, the rate of change will now be determined by the pink LFO Freq slide control in Panel 4 (Figure 2.3.4). Set the LFO to a very low frequency and you'll hear the tone getting very nasal, then very hollow. Listen to various LFO settings. This, of course, is an example of voltage control; the LFO is controlling the timbre of the output of VCO-1 automatically. You can do the same thing by raising and lowering the Pulse-Width slider manually. Try it.

Experiment 3: Pulse-Width Modulation (VCO-2)

Set the controls under the Audio Mixer as shown on Figure 2.3.6 and perform the entire preceding experiment using VCO-2 instead of VCO-1. When you've completed the experiment, return the Pulse-Width controls to the closed position.

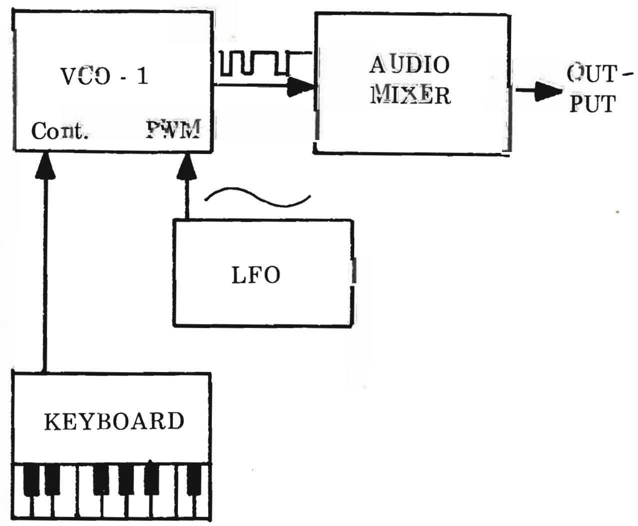


Figure 2.3.5. Block diagram of Figure 2.3.4.

PANEL 5

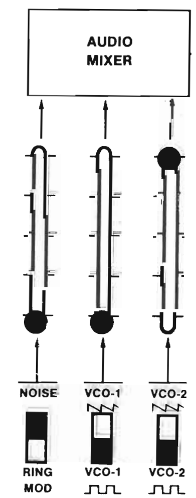


Figure 2.3.6.
Pulse-width modulation
experiment (VCO-2).

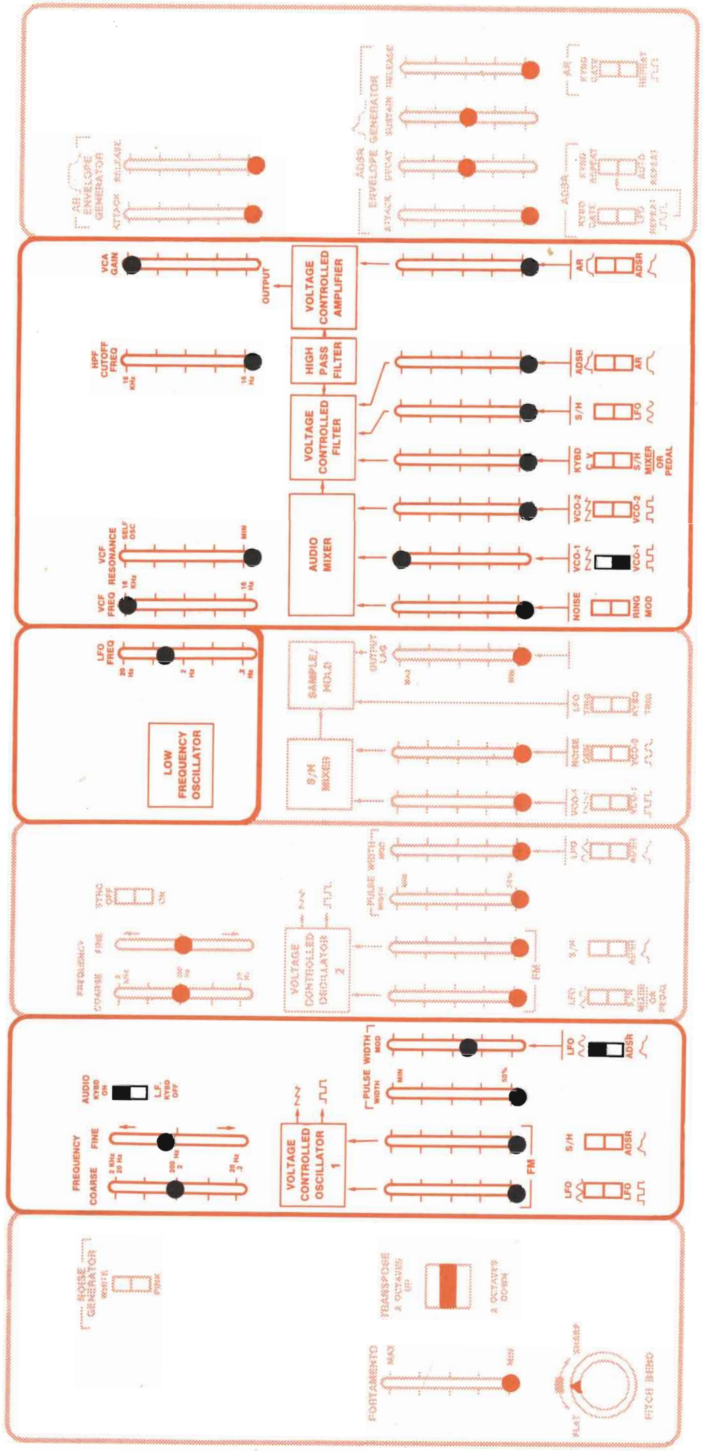


Figure 2.3.4. Basic patch for pulse-width modulation.

Lesson 4: Frequency Modulation

Remember that systematically changing the frequency of a waveform by employing the pattern of a second waveform is called *frequency modulation.* Refresh your memory on the concept of frequency modulation by turning back to Part I of the text. Set the Odyssey controls as shown in Figure 2.4.1 on the next page and you're ready to try some examples.

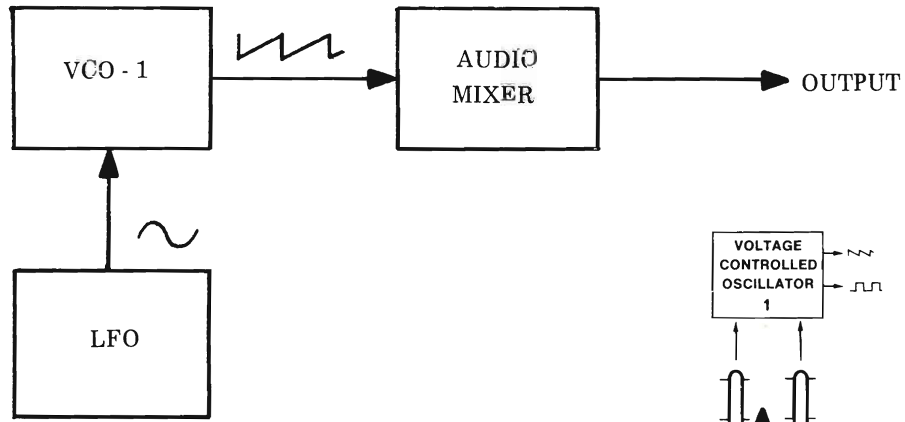


Figure 2.4.2. Block diagram of Figure 2.4.1.

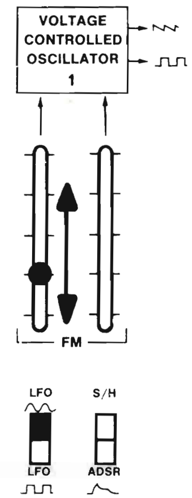


Figure 2.4.3. LFO switch (VCO-1).

Experiment 1: Vibrato and Trill

Locate the pink FM (frequency modulation) attenuator (second panel from the left). For now, just use the first (left) slider, as shown in Figure 2.4.3, and experiment with it in several positions. You should hear a *vibrato.* Directly under the slider, you'll find a two-position LFO switch; move it down and you'll hear a *trill.* As you raise the pink slider, the two pitches of the trill become farther apart; lowering the slider brings the pitches closer together. Experiment with the blue Pulse-Width slider, the pink LFO Freq slider, and the VCO-1 selector in Panel 5, to create additional effects.

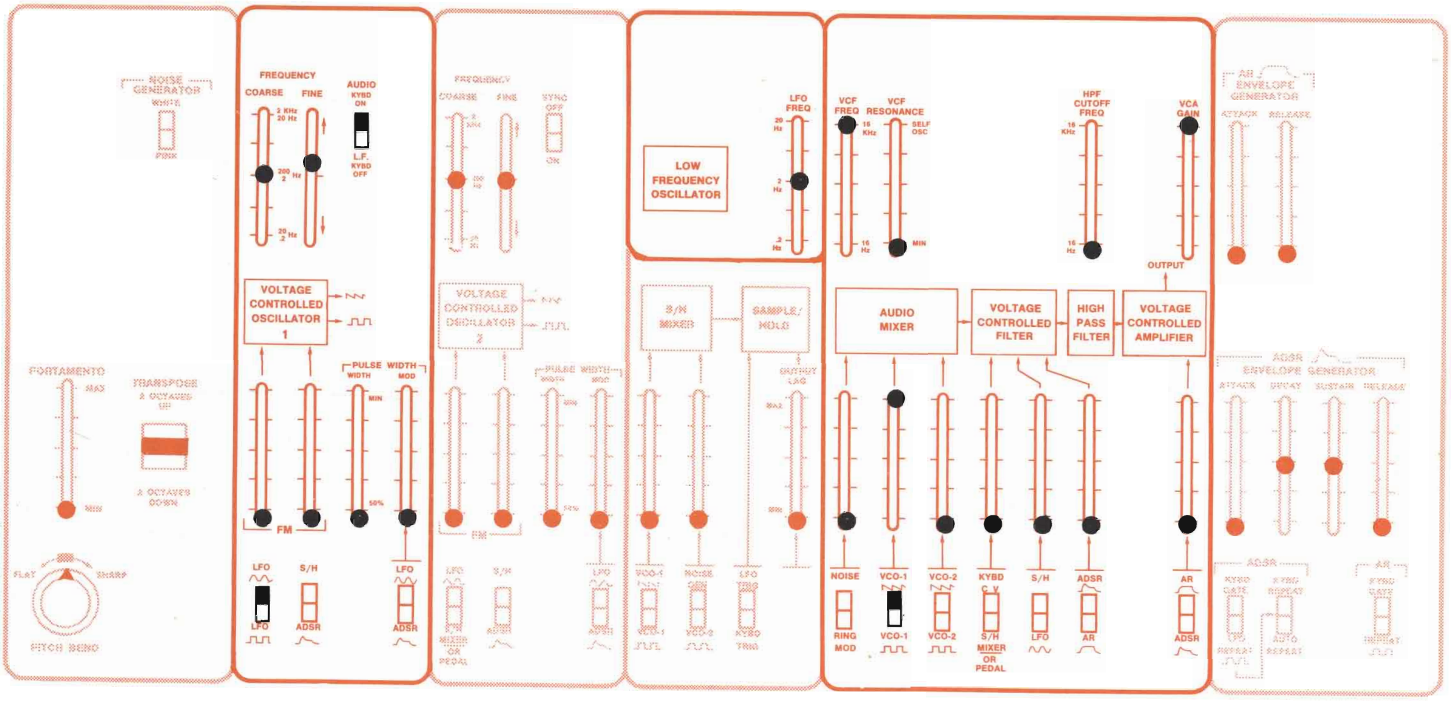


Figure 2.4.1. Basic patch for frequency-modulation experiment (VCO-1).

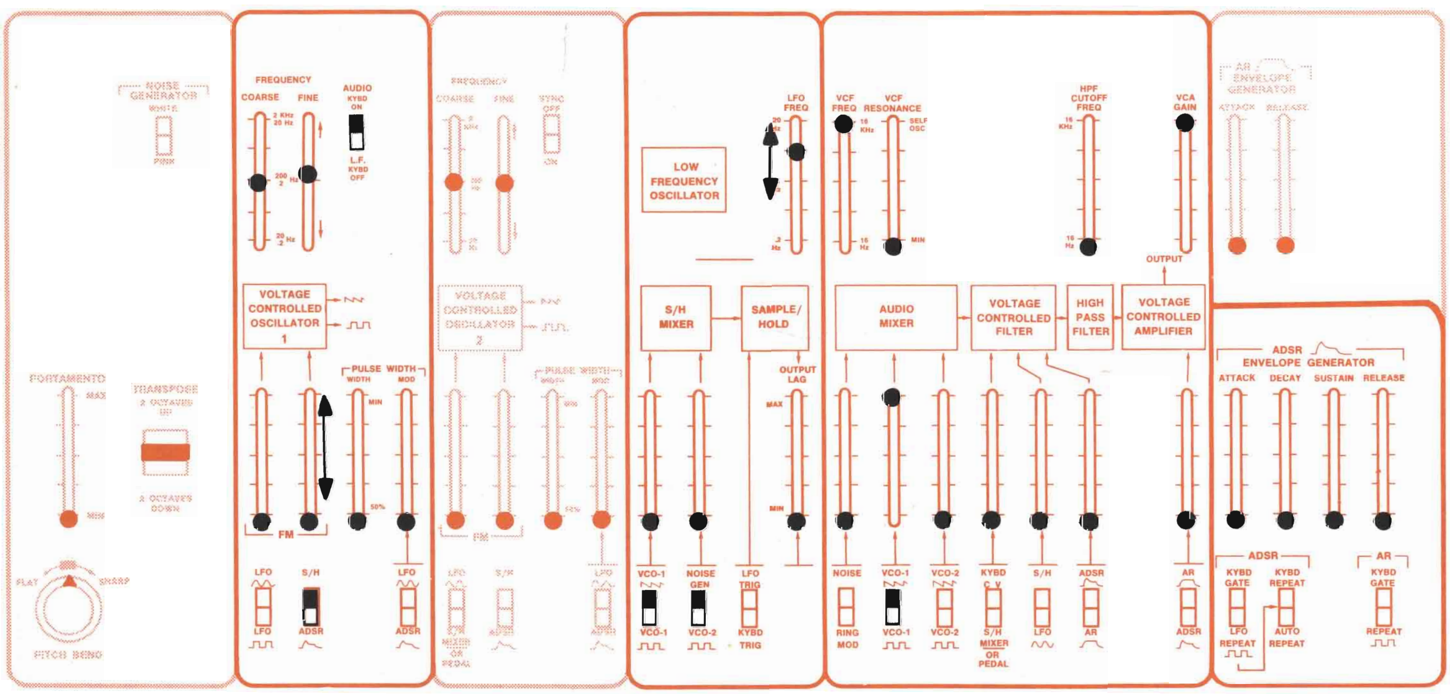


Figure 2.4.4. S/H circuit used to modulate pitch of VCO-1.

Experiment 2: Voltage Control

The second frequency-modulation attenuator in Panel 2 (yellow) permits you to voltage-control the pitch (frequency) of VCO-1 and VCO-2 with either the *Sample and Hold* function (Panel 4) or the *ADSR* envelope generator (Panel 6). Be sure that the controls of the Odyssey are set as shown in Figure 2.4.4. If they have been set correctly, you should hear the sound of the VCO-1 sawtooth wave. Now raise the second slider (the white one, second from the left) on the S/H Mixer panel (Panel 4). After doing this, raise the frequency-modulation slider (yellow, on VCO-1) located above the S/H-ADSR selector switch. The result should be a series of rapidly changing pitches, the speed of which can be controlled by the LFO Freq slider. Vary the speed (frequency) of the LFO. Then perform the same experiment using VCO-2, setting the controls under the audio mixer as shown on Figure 2.4.5.

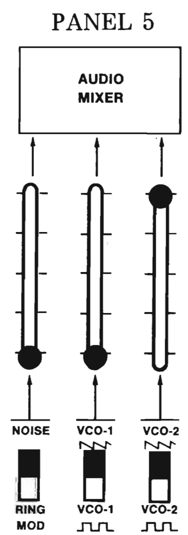


Figure 2.4.5. Audio Mixer controls for using VCO-2.

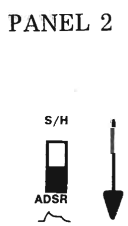


Figure 2.4.6. S/H-ADSR selector switch (VCO-1).

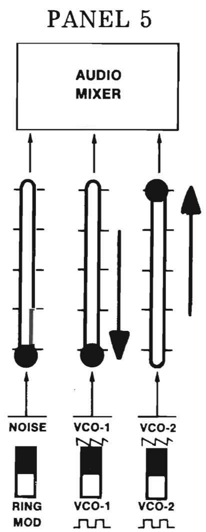


Figure 2.4.7. Same as Figure 2.4.5.

Experiment 3:

After experimenting with the Sample and Hold function, return to the settings shown in Figure 2.4.4 and push the two-position selector switch in Panel 2 down from S/H to ADSR (Figure 2.4.6). The frequency (pitch) of VCO-1 may now be controlled by the ADSR envelope generator. Raise the yellow slider on VCO-1 again and vary

the attack, decay, sustain, and release settings to find out how the pitch of the oscillator is affected. Playing any note on the keyboard will trigger the envelope generator.

Experiment 4: Voltage Control (VCO-2)

Change the controls under Audio Mixer so that you can listen to the output of VCO-2 (Figure 2.4.7). Repeat Experiment 3, now using the controls on VCO-2.

Each of the above procedures provides a method of voltage-controlling the sound sources (oscillators). The precise means by which each function — the Sample and Hold and the ADSR — exerts its control will be discussed later under "Controllers."

Lesson 5: Synchronization

Synchronization of the two oscillators (VCO-1 and VCO-2) is enabled by the selector switch shown in Figure 2.5.1. When this switch is "on" (down position), the audio signal from VCO-2 is *forced* to conform to the frequency of VCO-1. For a demonstration of synchronization, set the controls to match Figure 2.5.2.

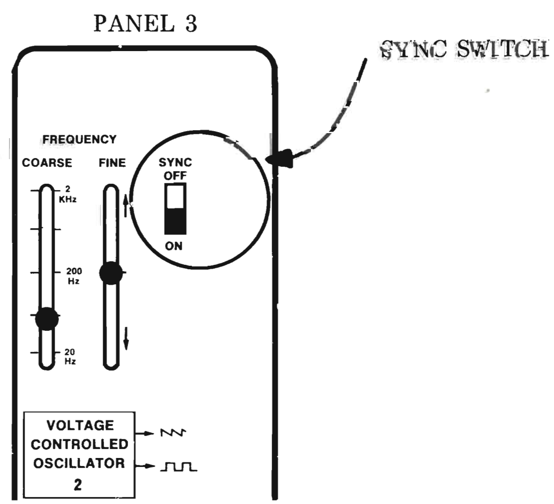


Figure 2.5.1. Synchronization switch.

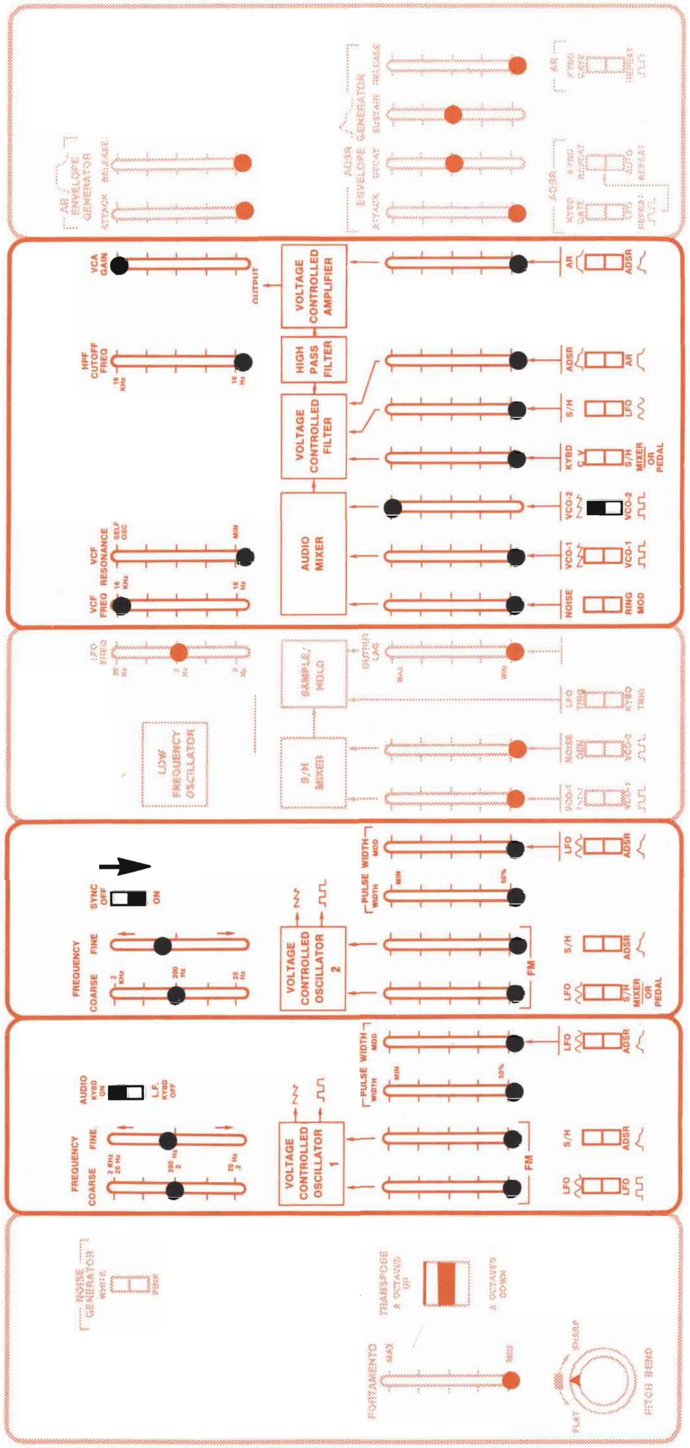


Figure 2.5.2. Patch for synchronization of VCO-1 and VCO-2.

With the Sync switch on, slowly move the green coarse-tuning slider of VCO-2 through its entire range. Notice the change in timbre. Move the pitch of VCO-1. Notice that the pitch of VCO-2 changes with the pitch of VCO-1. At certain settings of VCO-2 pitch, the pitch of VCO-1 disappears, and you only hear a high-pitched harmonic of VCO-1. The disappearance of the pitch of VCO-1 results because you tuned VCO-2 to an exact harmonic of VCO-1. Incidentally, if the frequency setting of VCO-2 is below that of VCO-1, the volume becomes softer and the timbre does *not* change. As the green VCO-2 slider is raised higher than the blue VCO-1, the timbre of VCO-2 does change. Experiment again, changing the frequency of VCO-1 while leaving VCO-2 in the middle of its range. The basis for some exciting sounds will be found in this operation.

Experiment 1: How Synchronization Works

Let's look at some diagrams which will show graphically the synchronization of VCO-2 to VCO-1. Start by listening to the sawtooth outputs of both audio oscillators with the Sync switch in the "off" position (Figure 2.5.3). Set VCO-2 (green) so that it has a slightly higher pitch than VCO-1. Figure 2.5.4 shows the difference in waveforms due to the higher pitch (frequency) of VCO-2. Now turn the Sync on. What happens to the VCO-2 waveform is shown in Figure 2.5.5.

When VCO-2 sounds with the Sync switch on, VCO-1 interrupts or "resets" the signal from VCO-2 when VCO-1 begins each new cycle. Figure 2.5.5 may help to explain. Again, move the coarse-tuning

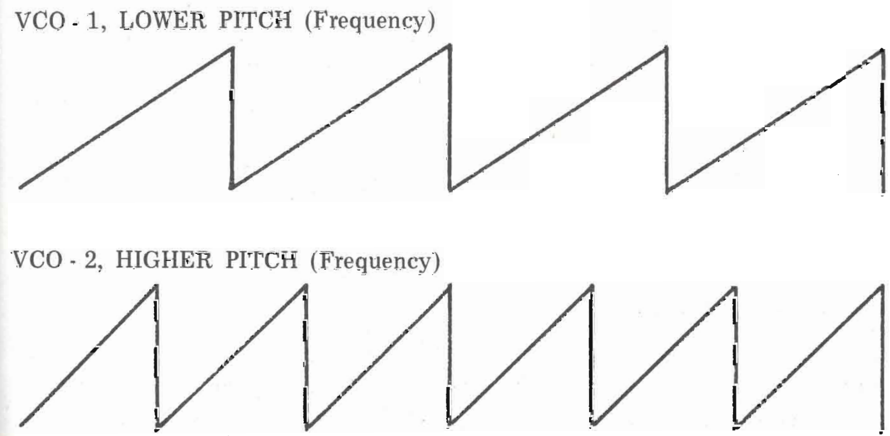


Figure 2.5.4. Sawtooth waveforms out of synchronization. Sync switch, "off."

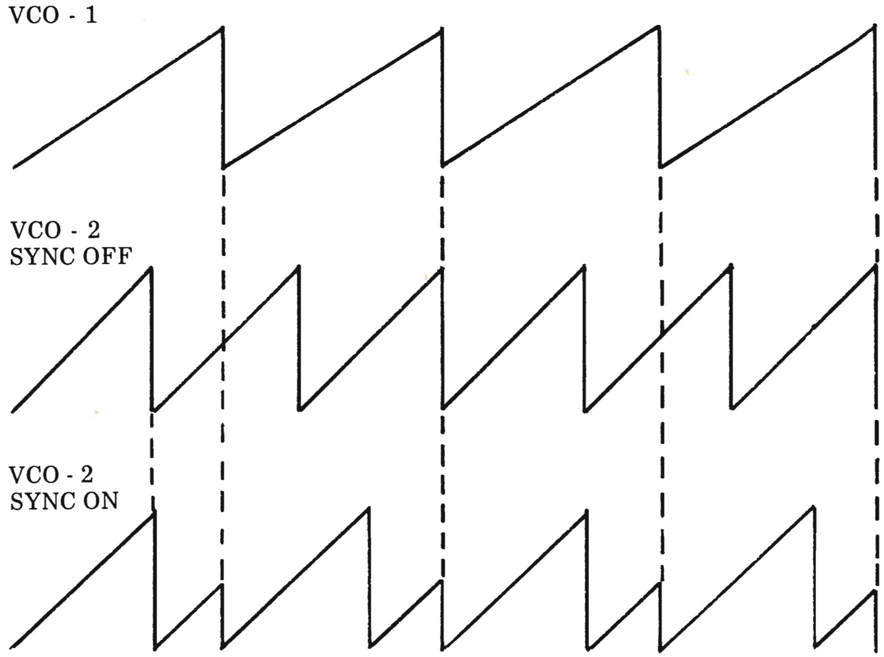


Figure 2.5.5. Comparison of sawtooth waveforms out of and in synchronization.

slider on VCO-2 through its entire range with the Sync switch on, going from low to high. Listen carefully as the fundamental of VCO-1 gets weaker. Raising the slider will enable you to center in on the harmonics of VCO-1 with the fundamental eventually fading completely, then returning as the VCO-2 slider is moved higher. Don't hesitate to use the frequency-modulation and pulse-width controls to see the various effects which you can create, always being sure to picture mentally what the waveform changes would look like on paper.

Lesson 6: Tuning the Oscillators

Open the blue VCO-1 and green VCO-2 attenuators into the Audio Mixer as shown in Figure 2.6.1.

Experiment 1: Unison

Position the blue coarse-frequency control slider of VCO-1 to a medium pitch. Then manipulate the coarse-tuning slider of VCO-2 until it sounds *close* to the pitch of VCO-1. You'll perceive a continuous series of rapid beats or waverings as you near the pitch of VCO-1. Use the fine-tuning control, continuing to make adjustments until

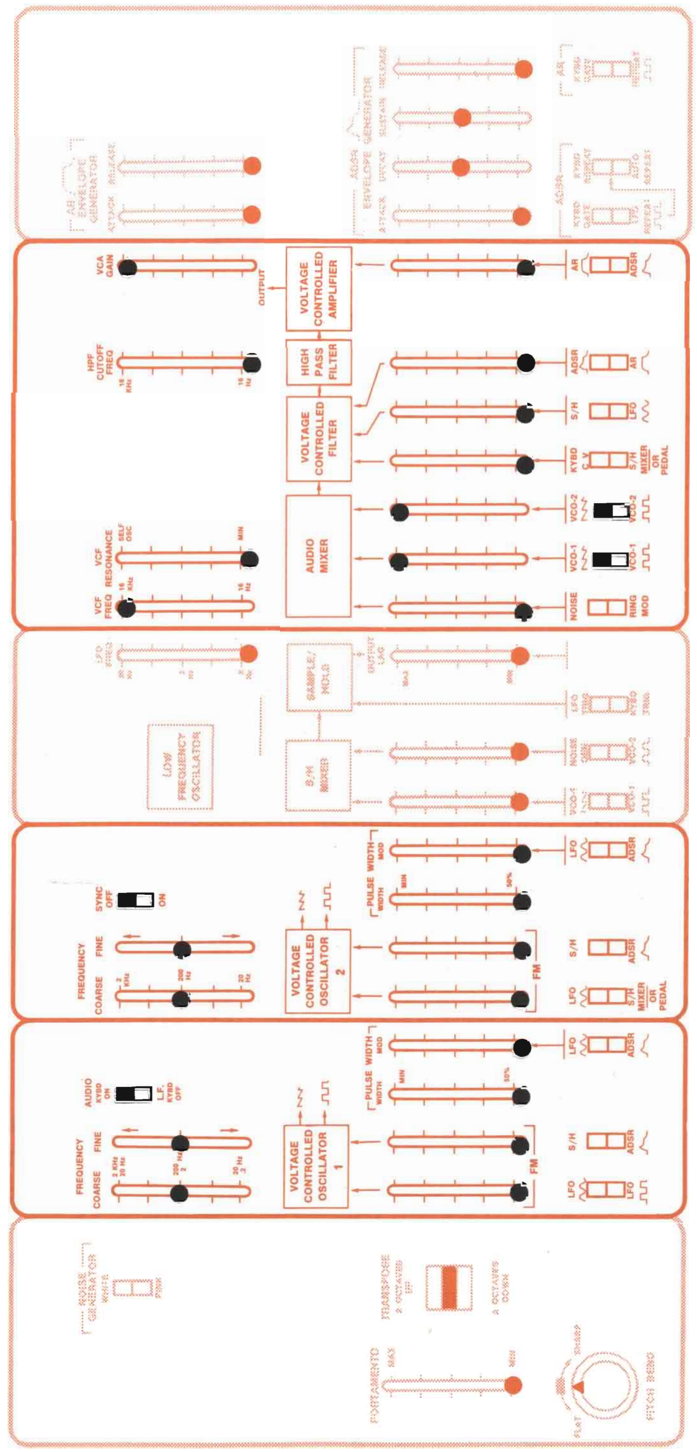


Figure 2.6.1. Frequency (pitch) tuning controls.

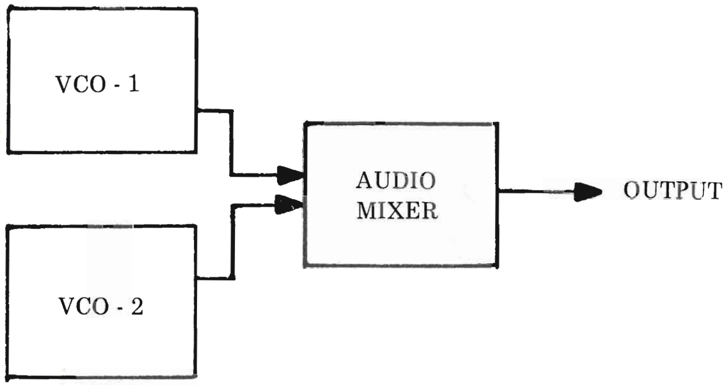


Figure 2.6.2. Block diagram of Figure 2.6.1.

the beats slow down and finally stop. When the beats terminate, the oscillators are tuned to exactly the same pitch — they are now "in unison." Following the same procedure, tune the oscillators to many different pitches, both low and high.

Experiment 2: Interval Tuning

If you wish to try a more challenging task, tune the oscillators to various intervals such as octaves, fifths, and thirds. (You'll find this may take a bit more practice.)

Section 3: Signal Modifiers

Now that you have examined various sources of raw-signal production, it's time to explore those components of the Odyssey which fashion the raw material/signals into an audibly useful form. These components, called signal modifiers, are: (1) the voltage-controlled filter (VCF), (2) high-pass filter (HPF), (3) voltage-controlled amplifier (VCA), and (4) the ring modulator — all on Panel 5. One of the four signal modifiers — the VCF — has the dual capacity of both a signal source and a signal modifier. This dual function will be discussed in Lesson 3. Let's begin by learning about the first and most important of the Odyssey's signal-modifying devices.

Lesson 1: Voltage-Controlled Filter (VCF)

Find the VCF components shown in Figure 3.1.1 — three attenuators (black, yellow, and red) located below the words "Voltage Controlled Filter" and two sliders (black) above and to the left labeled VCF Freq and VCF Resonance. The Voltage-Controlled Filter (VCF) is a low-pass band-pass filter which can be controlled in two ways: either by a voltage from another circuit or manually, using the "VCF Freq" and "Resonance" controls. Keep in mind that filter names such as "low-pass" are descriptive in nature.

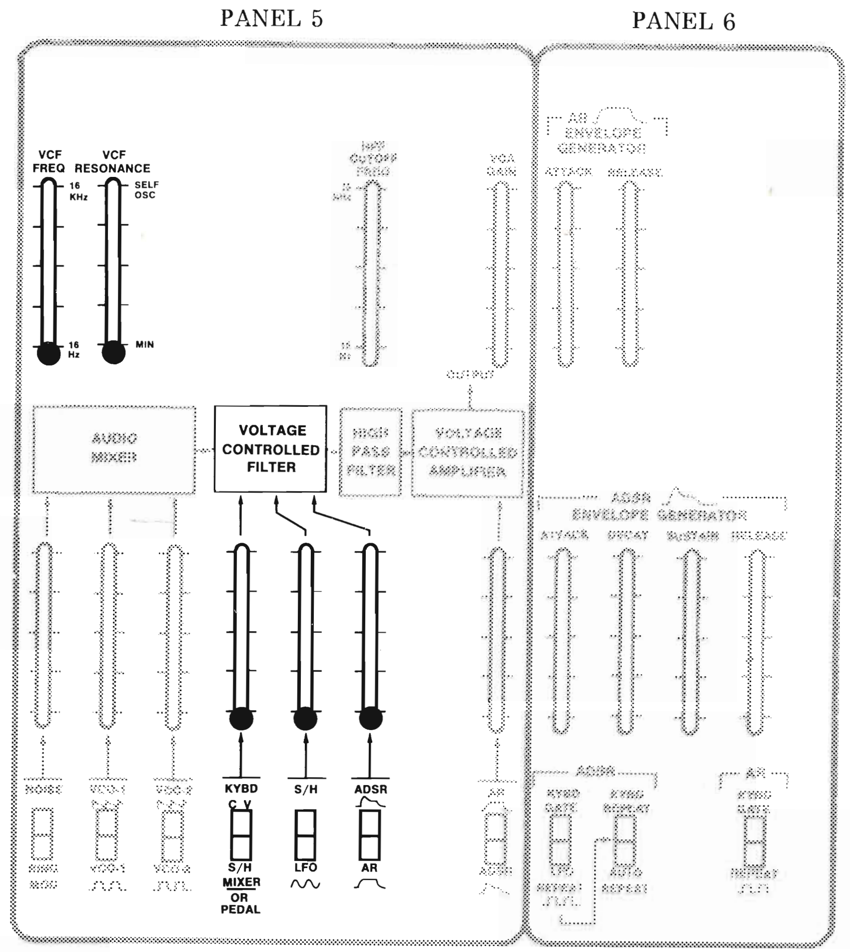


Figure 3.1.1. Location of Voltage Controlled Filter controls.

Raise the white noise slider shown in Figure 3.1.2 (Audio Mixer) and experiment with different positions of the black VCF frequency slider. Next, move this slider to the top of its range. Then gradually lower it all the way through the entire VCF Freq range. You'll hear the high frequencies of the noise signal weaken and disappear, then the middle frequencies, and finally the low frequencies will also vanish.

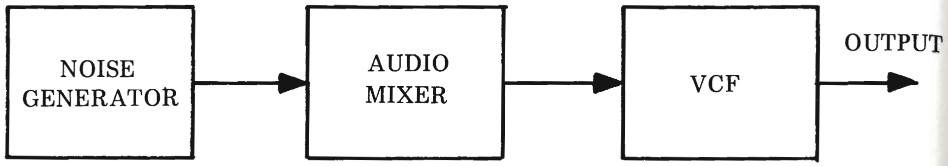


Figure 3.1.3. Block diagram of Figure 3.1.2.

Experiment 2: VCF Resonance Slider

The VCF Resonance slider is located directly to the right of the VCF Freq (Figure 3.1.4). As the Resonance slider is advanced, the VCF changes from a low-pass filter into a band-pass filter. As the Resonance slider is raised, you'll hear the low frequencies fade as the band of frequencies which is allowed to pass through the filter becomes narrower. Use noise as the signal source, and experiment with various positions of the Resonance slider.

Experiment 3:

Repeat Experiment 1, but with the Resonance slider halfway up (Figure 3.1.5).

PANEL 5

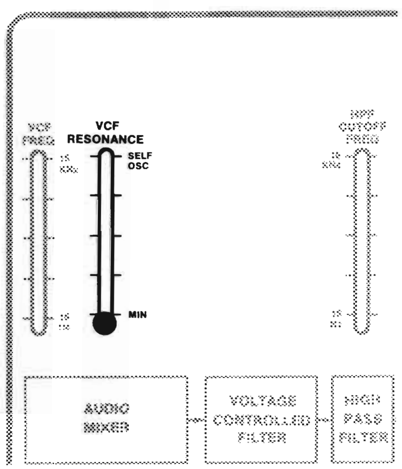


Figure 3.1.4. VCF Resonance control.

PANEL 5

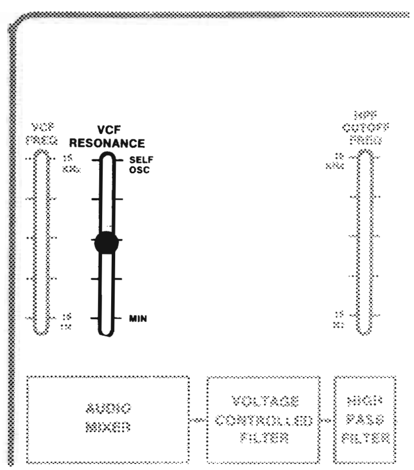


Figure 3.1.5. VCF Resonance control.

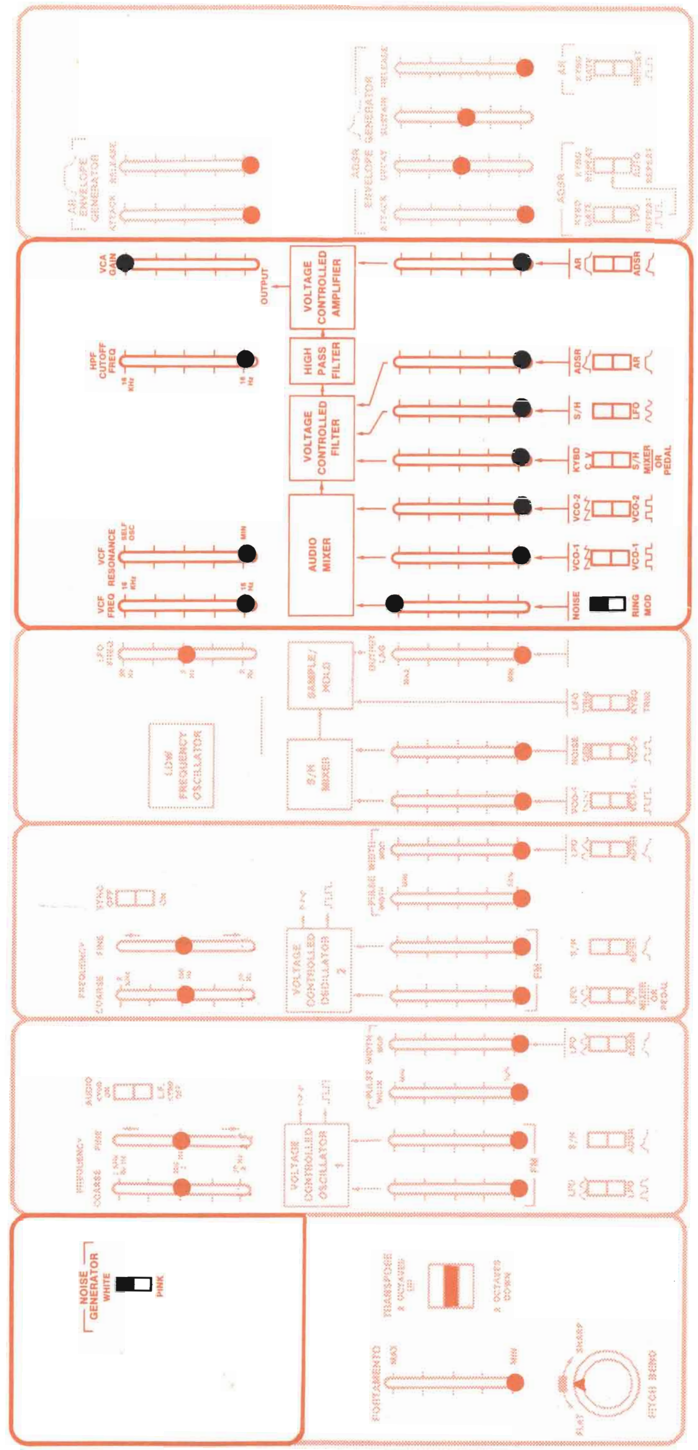


Figure 3.1.2. Basic patch for VCF frequency experiments.

Experiment 4: VCF Resonance Control of VCO-1 and VCO-2

Perform the preceding experiment using the sawtooth wave VCO-1 as the signal source. Listen as the pass-band — those frequencies allowed to pass — sweeps through the harmonics of VCO-1 as you raise and lower the VCF Freq control. Experiment with different settings of the Resonance control.

Experiment 5: ADSR Control of VCF

First locate the ADSR Envelope Generator controls in the lower half of the last panel on the right (Figure 3.1.6). Carefully set the controls as pictured, and you'll be ready for an example of how the VCF may be voltage-controlled by the ADSR Envelope Generator.

Begin by moving the red ADSR attack slider to various positions. The change in sound is due to the "automatic" voltage control of the VCF by the ADSR Envelope Generator. Return the attack slider control to its lowest position and experiment with the decay, sustain, and release sliders. The speed of repetition of the sound is controlled by the pink LFO Freq slider in Panel 4.

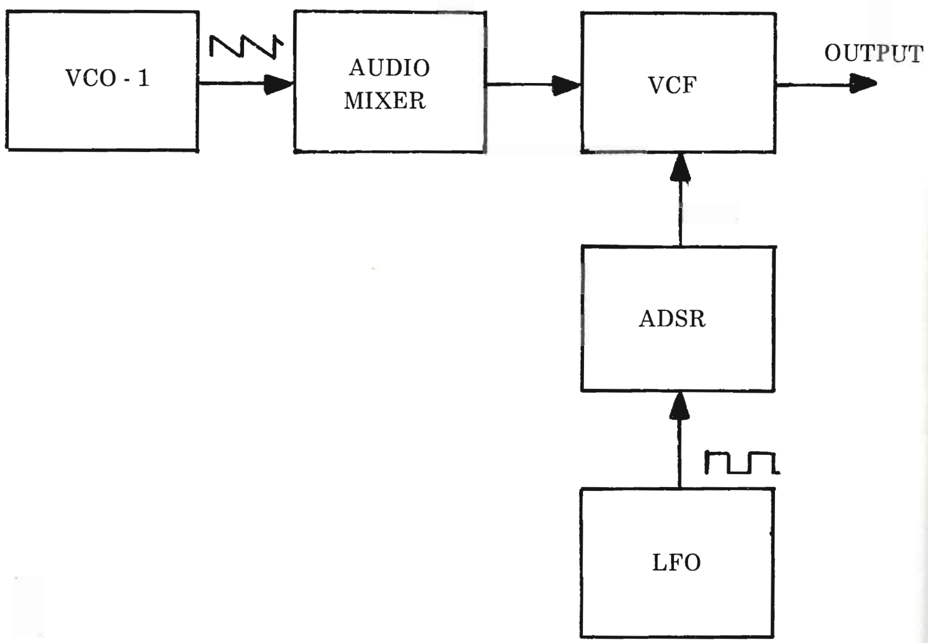


Figure 3.1.7. Block diagram of Figure 3.1.6.

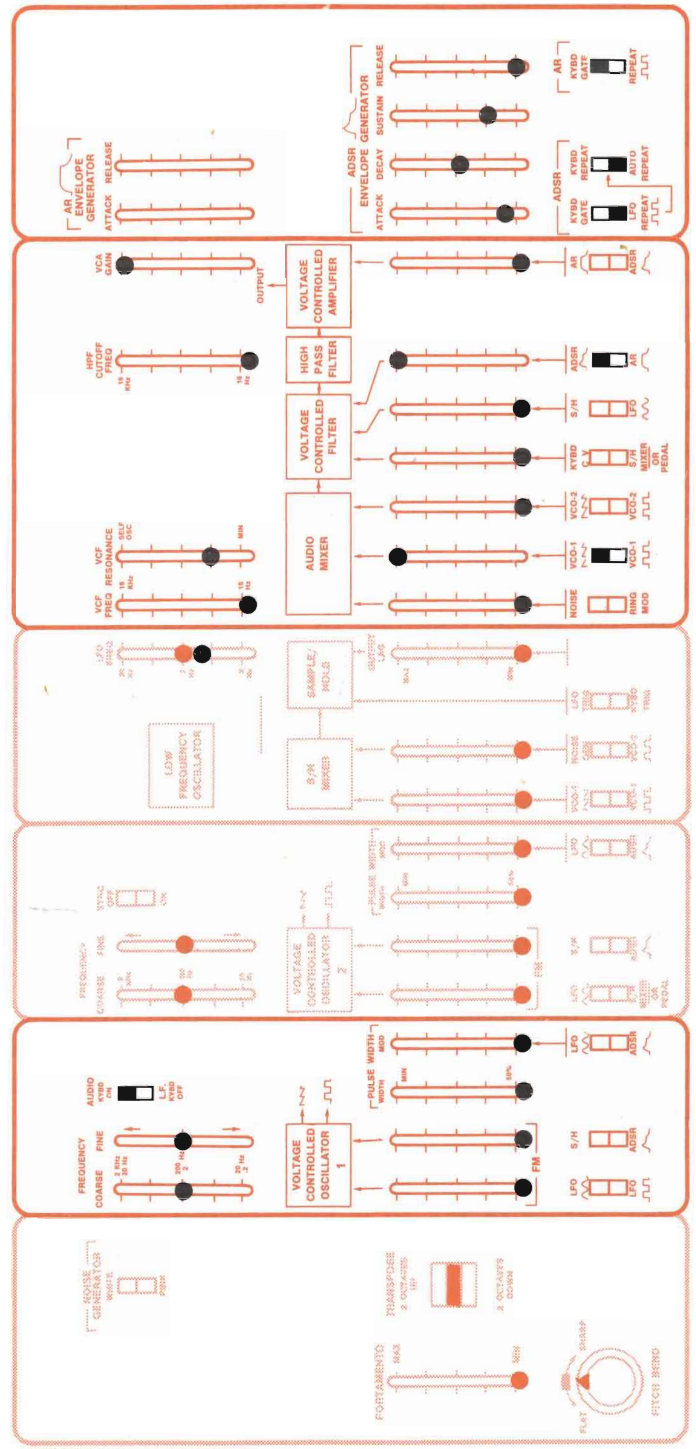


Figure 3.1.6. Basic patch for ADSR Envelope Generator experiments.

Try different settings of the red attenuator which controls the amount of ADSR signal to pass into the filter. This attenuator is located just under the box labeled "High Pass Filter" and just over the ADSR/AR switch. Notice that this attenuator determines the "brightness" of the sound. The reason it does this is because the voltage which is produced by the ADSR Envelope Generator is being used to "open" the VCF, which in turn lets the high-frequency (bright) sounds pass through. The attenuator will reduce the amount of ADSR signal which passes to the filter's control input and will determine how far the filter can "open." The lower the setting of this attenuator, the less ADSR signal is allowed to pass; consequently, the less the VCF will respond to the output of the ADSR.

Before going on to Experiment 6, be sure to review the information presented in Part I on the function of the envelope generator. We will discuss both the AR and ADSR generators again under "Controllers"; however, it is important to understand here that by manipulating the controls of the ADSR you are shaping an aperiodic waveform that is opening and closing the VCF in accordance with the settings of the ADSR controls. By voltage-controlling the filter in this manner, you have a precise and extremely sensitive means by which you can control the *timbre* of any sound.

Experiment 6: Filter Tremolo

Tremolo is a form of timbre modulation. Set the controls to the positions illustrated in Figure 3.1.8 and you'll hear an example of tremolo. The sine wave output of the LFO is being used to open and close the VCF automatically, as if you were moving the pink VCF Freq control back and forth by hand. Try changing the setting of the LFO Freq control and notice how the speed of the tremolo changes. Vary the position of the yellow slider over the S/H-LFO switch and notice how this attenuator varies the depth of the tremolo, in much the same way the red slider next to it controlled the amount of signal from the ADSR that reached the filter in the previous experiment. Listen to each of the three sound sources one at a time by raising the white, blue, and green sliders under the Audio Mixer box. Change the setting of the VCF Freq control and notice how this control still determines the overall brightness of the sound.

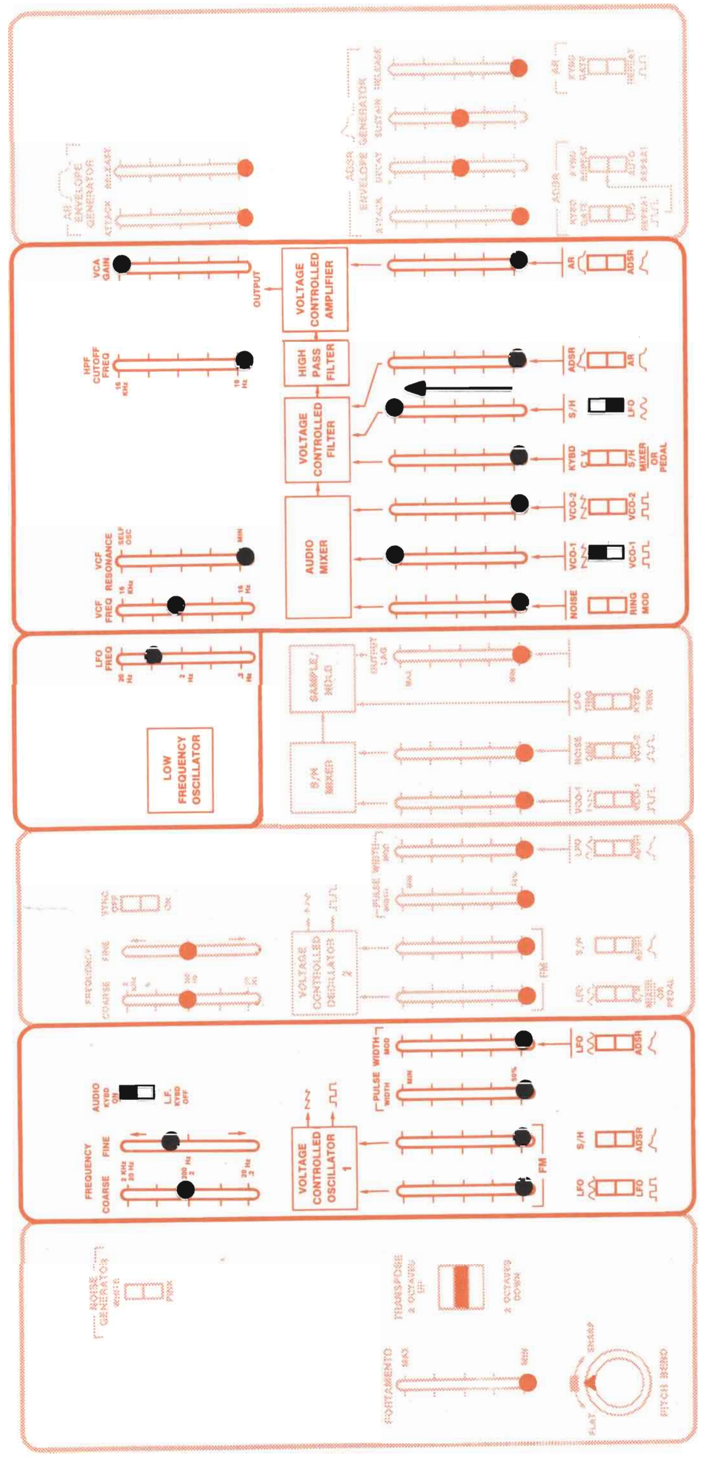


Figure 3.1.8. Basic patch for tremolo experiments.

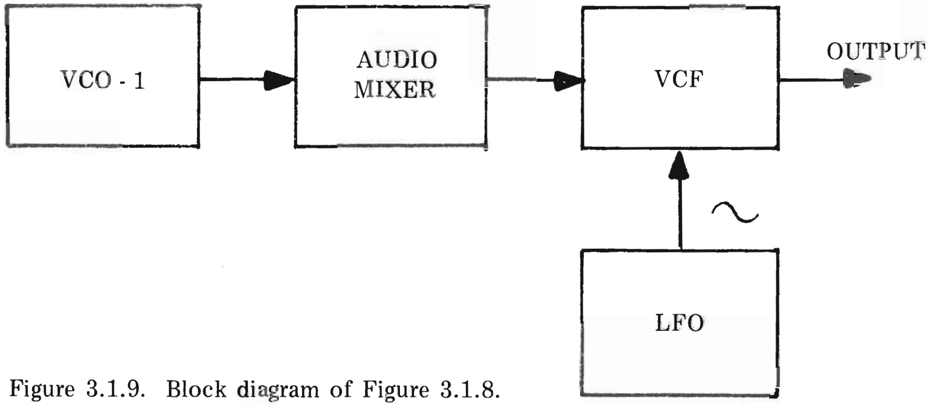


Figure 3.1.9. Block diagram of Figure 3.1.8.

Experiment 7: Voltage-Controlled Filtering of the Noise-Generator Signal

In the beginning of this section you created the sound of surf by manually controlling the output of the noise generator. Now you can use a voltage control to produce the same sound. It will be surprisingly more realistic this time. For a demonstration, set the controls according to Figure 3.1.10. Note the position of the LFO Freq slider. After listening to these sounds, do you agree that voltage control has many advantages over manual control?

Lesson 2: High-Pass Filter

Another component which functions as a signal modifier is the High-Pass Filter (HPF). Figure 3.2.1 shows the location to the left of the VCA Gain slider in Panel 5, labeled "HPF Cutoff Freq."

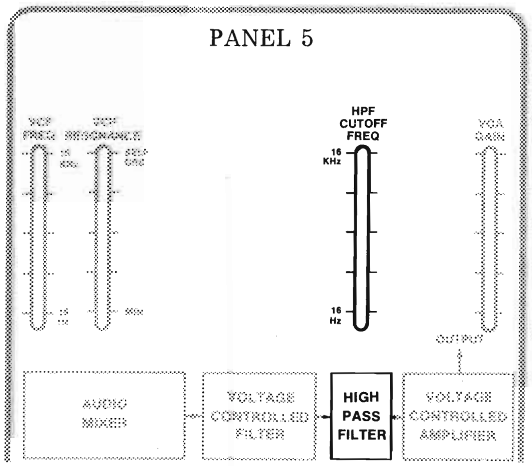


Figure 3.2.1. Location of High-Pass Filter.

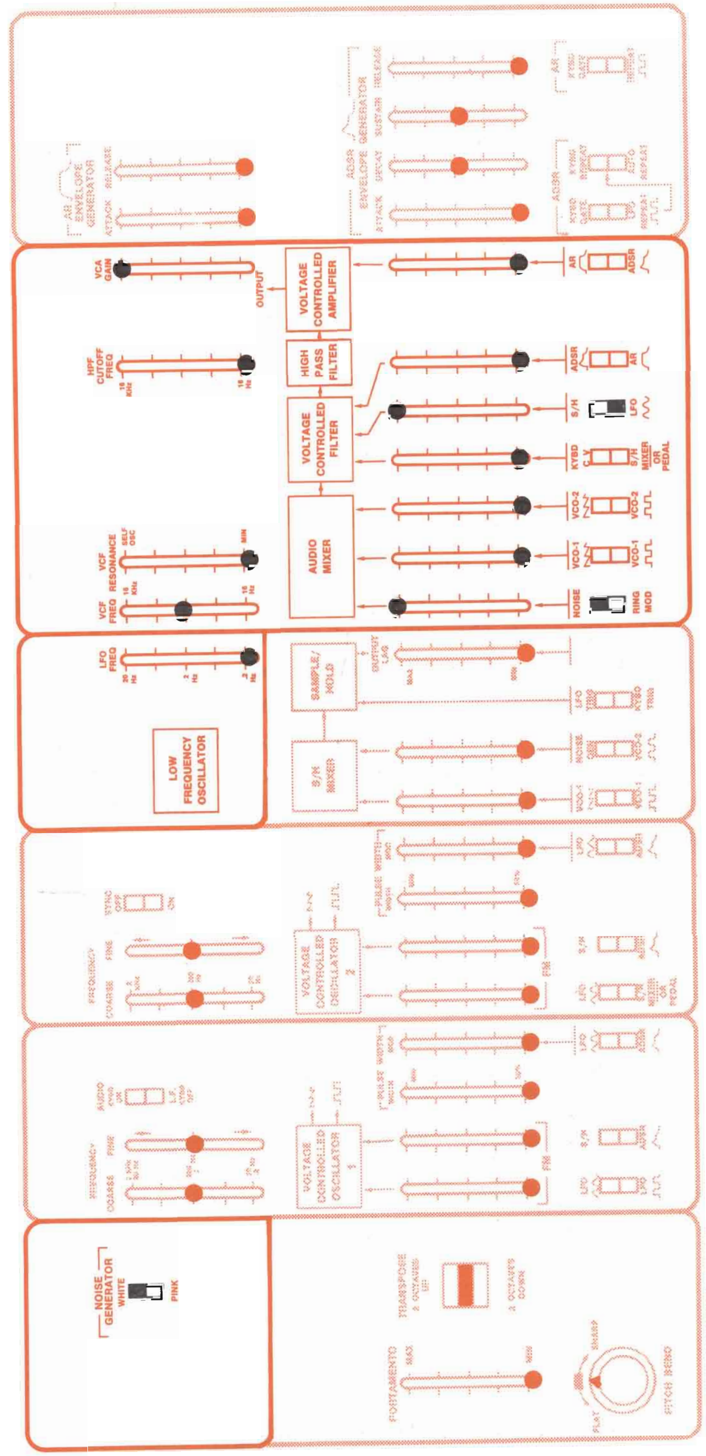


Figure 3.1.10. Voltage-controlled filtering of Noise Generator signal.

The high-pass filter does exactly what its name implies — it lets the high frequencies pass through while filtering out the low frequencies. This particular filter is *not* voltage-controlled, and can only be adjusted manually with the black slider. The HPF is useful in removing the "boomy" quality from lower pitches, and also aids in the texturing of certain instrumental sounds. Set the controls as illustrated in Figure 3.2.2.

Experiment 1: Filtering VCO-1 with HPF

Move the HPF Cutoff Freq slider through its entire range. Notice how the low frequencies weaken and finally disappear as the slider reaches the top.

Experiment 2: Filtering White and Pink Noise

Use the same patch shown in Figure 3.2.2; this time, however, lower the blue slider over the VCO-1 selector switch (under the Audio Mixer box) and raise the white slider to hear the noise generator. Again raise and lower the HPF Cutoff Freq slider to observe the effect of the HPF on noise. Filter both white and pink noise. Remember that pink noise is produced by moving the selector switch of the noise generator to the "down" position. Of course, the HPF has the same effect on both periodic and aperiodic waveforms: it can remove low frequencies.

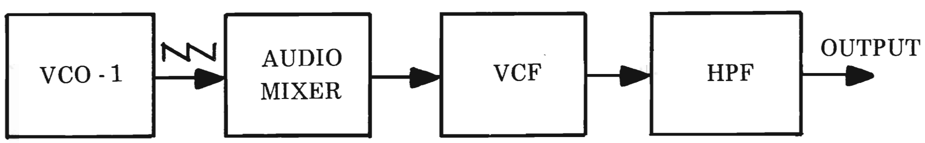


Figure 3.2.3. Block diagram of Figure 3.2.2.

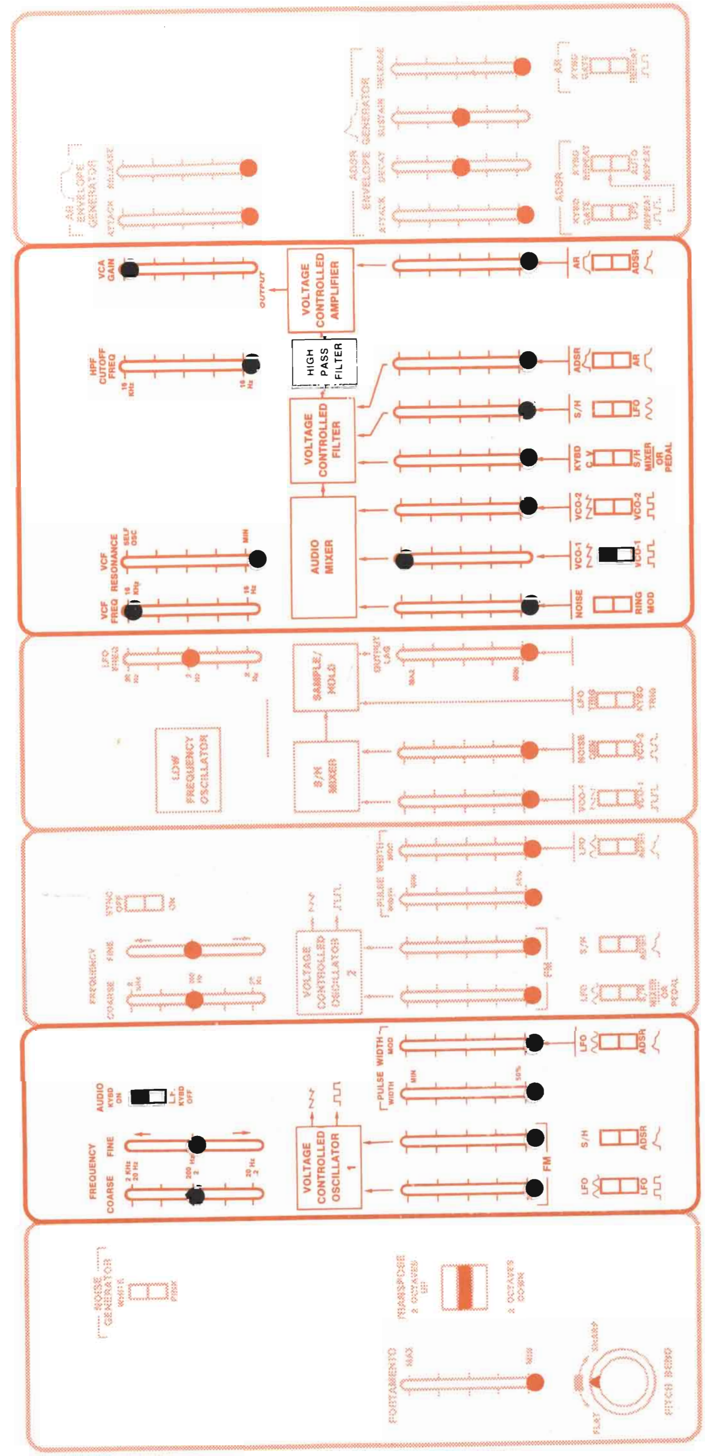


Figure 3.2.2. Basic patch for high-pass filtering experiments.

Lesson 3: VCF as a Tone Generator

At the start of this section on signal modifiers you learned that one component, the VCF, had the dual capability of both signal modifier and signal source. To turn the filter into a sound source, raise the Resonance slider all the way up (Figure 3.3.1). Focus your attention on the words Self Osc located to the right of the Resonance slider.

PANEL 5

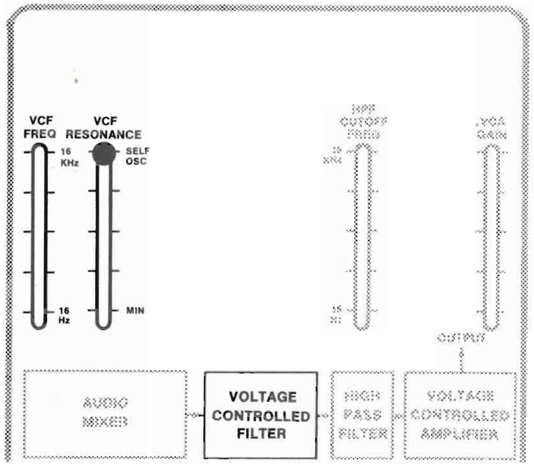


Figure 3.3.1. Using the VCF as a VCO.

Experiment 1: VCF Resonance in Self-Oscillation

Match the controls of the Odyssey to those of Figure 3.3.2. In this position, the VCF is no longer a signal modifier; it's acting as an oscillator producing a raw signal. As you can see, all the attenuators in the Audio Mixer are down, yet a sound is being produced. The waveform you are hearing is a pure sine wave. Raising the black attenuator over KYBD CV switch allows the keyboard control voltage (KYBD CV) to alter the frequency of this output. Note that the pitch of the tone produced by the self-oscillating filter can be changed both by the keyboard and by the VCF Freq control. There is really little difference between the operation of the filter when used as an oscillator (when the Resonance control is all the way up to Self-Oscillate) and either of the VCOs. You can even add vibrato to the pitch from the filter by raising the yellow slider which feeds the LFO into the filter. Be sure that the S/H-LFO switch is set to the LFO position if you want to try this.

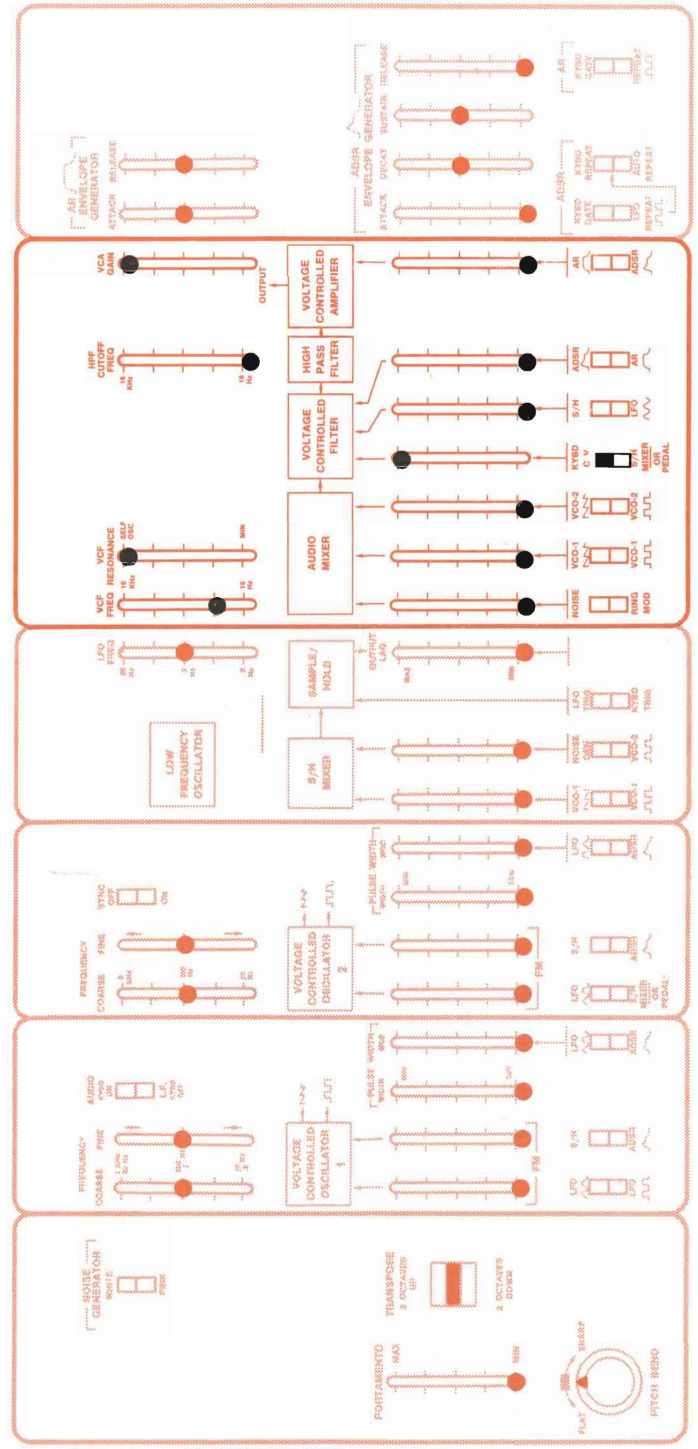


Figure 3.3.2. Basic patch for VCF as a signal generator.

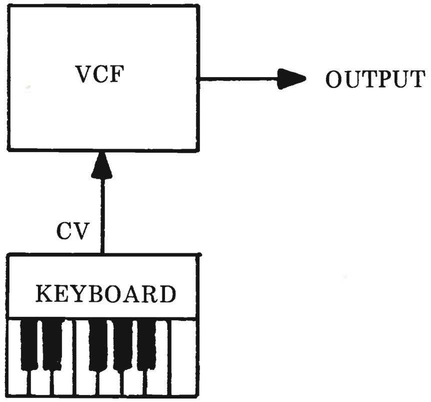


Figure 3.3.3. Block diagram of Figure 3.3.2.

Experiment 2: Playing Microtonal Scales on the VCF

The black slider under the VCF box allows the keyboard control voltage to control the filter, as we demonstrated in the previous experiment. Lowering this slider allows you to attenuate the keyboard control voltage before it reaches the filter and thereby permits you to reduce the pitch change created by the keyboard. Set this slider approximately halfway up, as shown in Figure 3.3.4. Note that the

PANEL 5

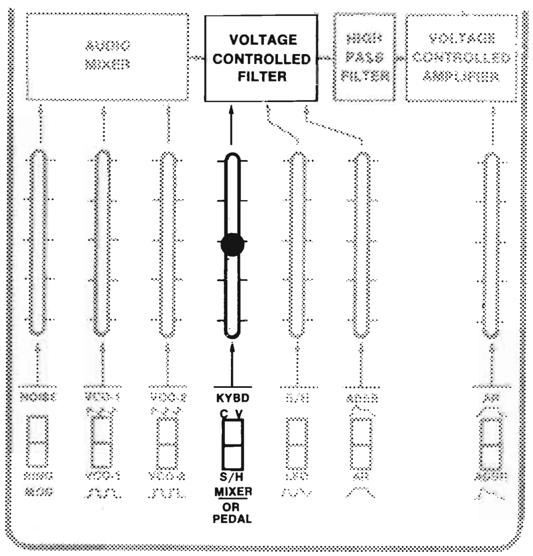


Figure 3.3.4. Creating microtonal scales.

range of the keyboard has been cut in half and playing an octave on the keyboard results in a pitch change of only half an octave. Similarly, every semitone (half tone) on the keyboard will now equal a quarter tone. Such a scale tuning is called a "quarter-tone scale." As you bring this attenuator down further, the pitches between the keys will become closer. Try setting the slider so that 3 octaves on the keyboard produces a pitch change of one octave.

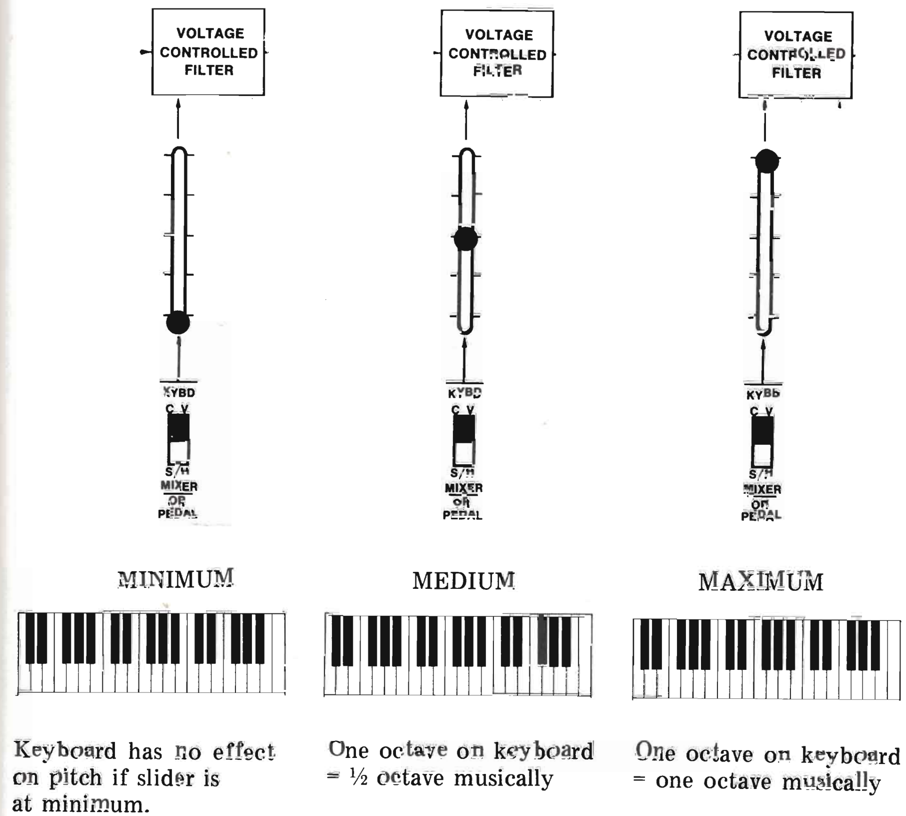


Figure 3.3.5. Different microtonal scales.

Lesson 4: The Ring Modulator

Another signal modifier which you'll learn to use is the Ring Modulator. The following diagram (Figure 3.4.1) will help you to understand this important component better. Study this "assembly line" illustration for a moment. First of all, the Ring Modulator has no controls of its own, and, as you can see from Figure 3.4.1, utilizes the outputs of VCO-1 and VCO-2. From the outputs of these two signal sources, the Ring Modulator produces a single complex output signal which contains all the sums and differences of the two oscillator frequencies. The output (raw sound) created by the Ring Modulator is entirely dependent upon the tuning of VCO-1 and VCO-2, and to a lesser degree upon the pulse-width settings of each signal source. If the two inputs are at two different frequencies not related by simple harmonic ratios, the Ring Modulator output will have a very complex characteristic since it contains high-frequency components which are not harmonics of either fundamental frequency. Locate the Ring Modulator switch as shown in Figure 3.4.2. Be sure the switch is in the "down" position. Set the other controls as shown.

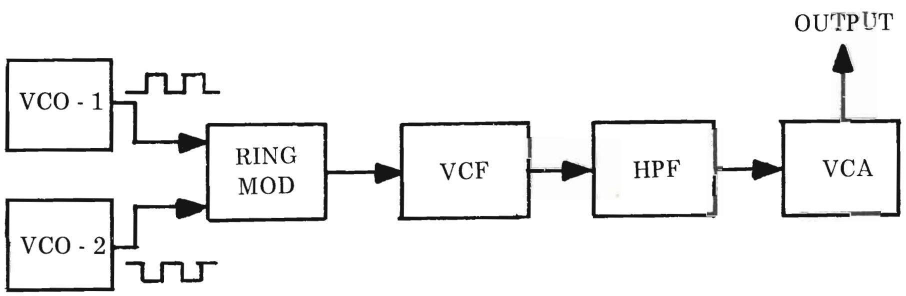


Figure 3.4.1. Block diagram of Ring Modulator function.

Experiment 1: Tuning for Effect

Raise the white Ring Mod attenuator located in the Audio Mixer section. With the Ring Mod attenuator in its highest position, begin moving the tuning sliders of the VCOs to different positions. As the tuning controls are changed, you'll hear the sound change. Change the position of the Pulse-Width sliders on VCO-1 and VCO-2. Observe the effect.

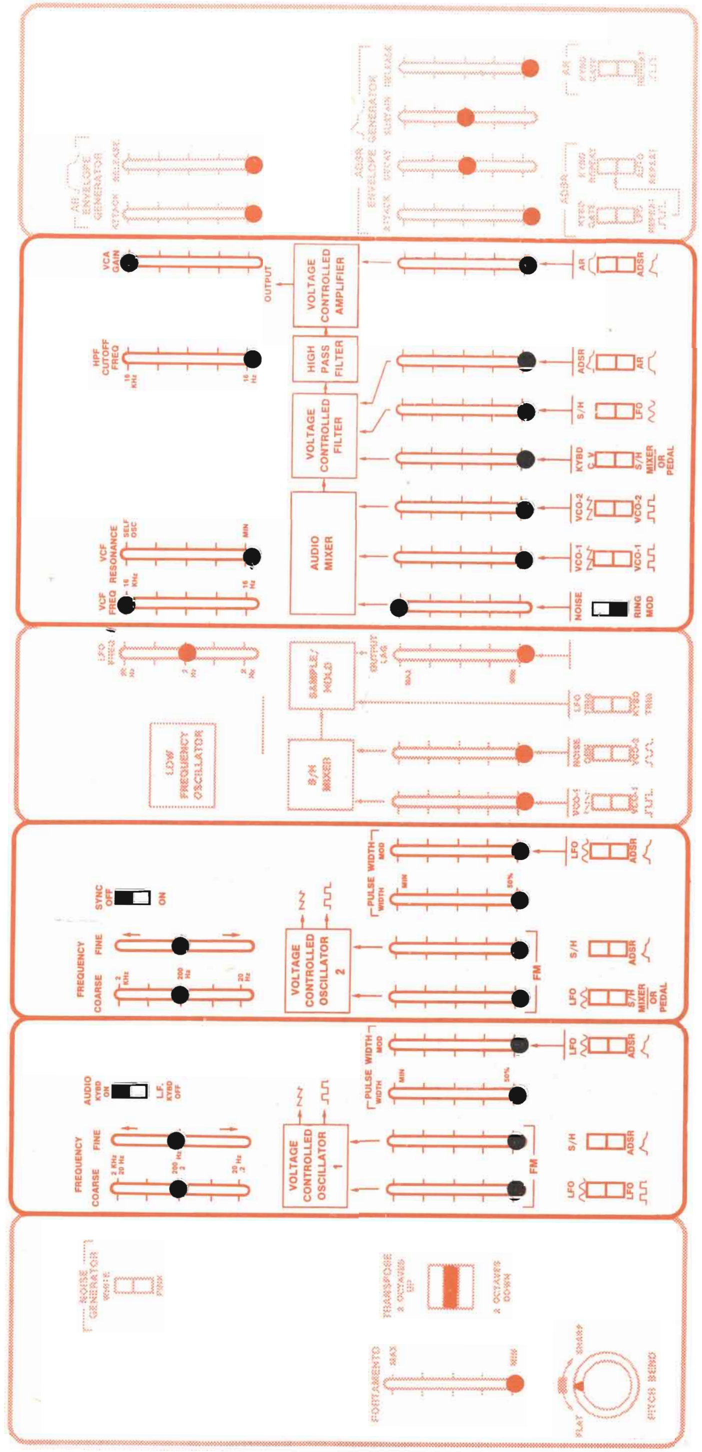


Figure 3.4.2. Basic patch for Ring Modulator experiments.

Experiment 2: Other Controls that Affect the Ring Modulator

Let's attempt to create a gonglike sound by employing the Ring Modulator and ADSR Envelope Generator. Again find the ADSR controls in the bottom half of Panel 6. Figure 3.4.3 illustrates. Set all the controls as shown. Try the following and observe the effect:

1. Change the positions of the pink and yellow VCO-1 and VCO-2 tuning controls and Pulse Width controls.
2. Change the position of the red attenuator under the High Pass Filter box. Note that this control affects the brightness of the sound. Recall that we experimented with this control in our discussion of the VCF.
3. Change the position of the VCF Freq control and note that it too influences the brightness of the sound.
4. Change the setting of the VCF Resonance control and listen to the effect that this control has on the sound. Remember what effect this control had on sounds in previous experiments.
5. Experiment with different settings of the four red ADSR Envelope Generator controls.
6. Try changing the positions of the pulse-width and frequency-modulation sliders on the VCOs to create additional sounds.
7. When the sync switch is in the "on" position, VCO-2 will be "locked" to a harmonic of VCO-1 even if VCO-2 is deliberately detuned when the sync switch is in the "sync off" position. Try using the sync switch to change rapidly from a gong tone to a bright "harmonic" tone.

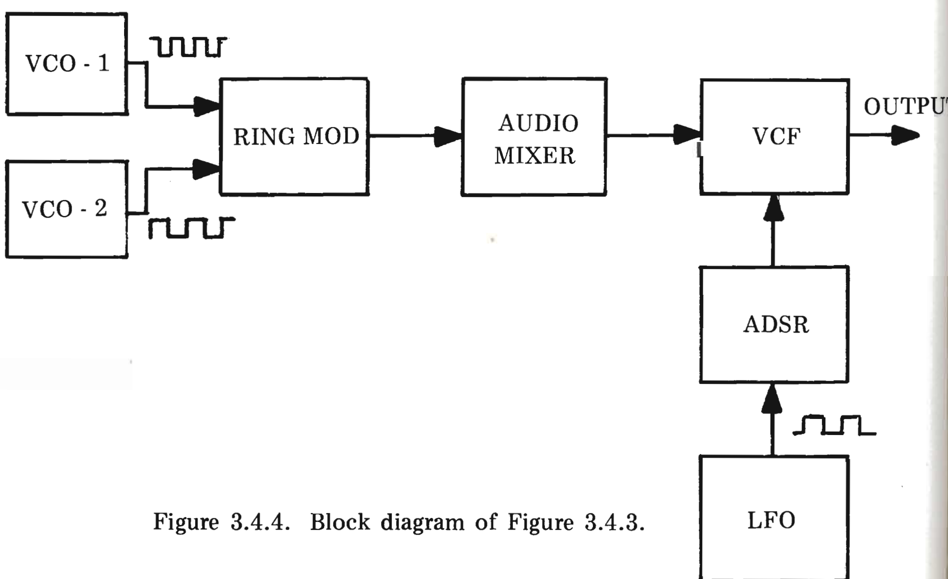


Figure 3.4.4. Block diagram of Figure 3.4.3.

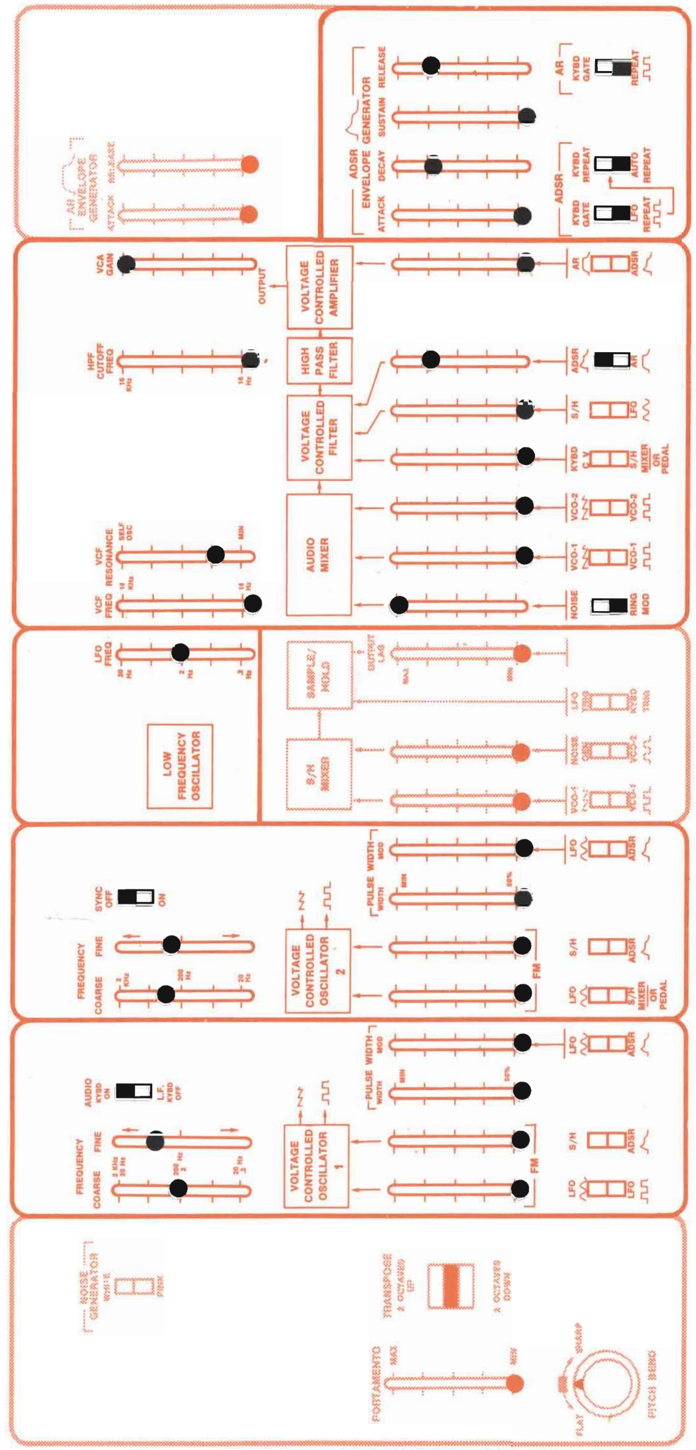


Figure 3.4.3. Using the Ring Modulator.

Lesson 5: The Voltage-Controlled Amplifier

The last signal modifier to be discussed is the Voltage-Controlled Amplifier (VCA). This component determines the final volume or amplitude of the output before it leaves the Odyssey. Find the VCA Gain slider and its companion voltage-control attenuator (red) located directly below it in Panel 5 (Figure 3.5.1).

The function of the VCA is to control the amount of signal passing from the other signal modifiers to the output. It is essentially a volume control that can be operated either manually (using the VCA Gain control) or automatically by signals from the AR Envelope Generator or the ADSR Envelope Generator fed in through the red attenuator located under the Voltage Controlled Amplifier box.

Raising the VCA Gain control will have the effect of permitting any signal passing through the other signal modifiers to reach the output of the synthesizer. The higher up the VCA Gain control is set, the greater the amplitude this signal will have at the output of the synthesizer, and, consequently, the louder the sound that will come out of the speaker you are using.

If a continuous signal exists at the input of the VCA which you wish to turn on and off when you are playing on the keyboard, for instance, you would normally leave the VCA Gain control all the way down and use the signal from the Envelope Generators to open and close the VCA. In this case, the red attenuator under the VCA box will control the volume of the over-all output of the Odyssey. The switch below the red slider will select control of the VCA by either the AR or ADSR Envelope Generators. Position the controls as pictured in Figure 3.5.2 and you'll be ready to demonstrate these principles.

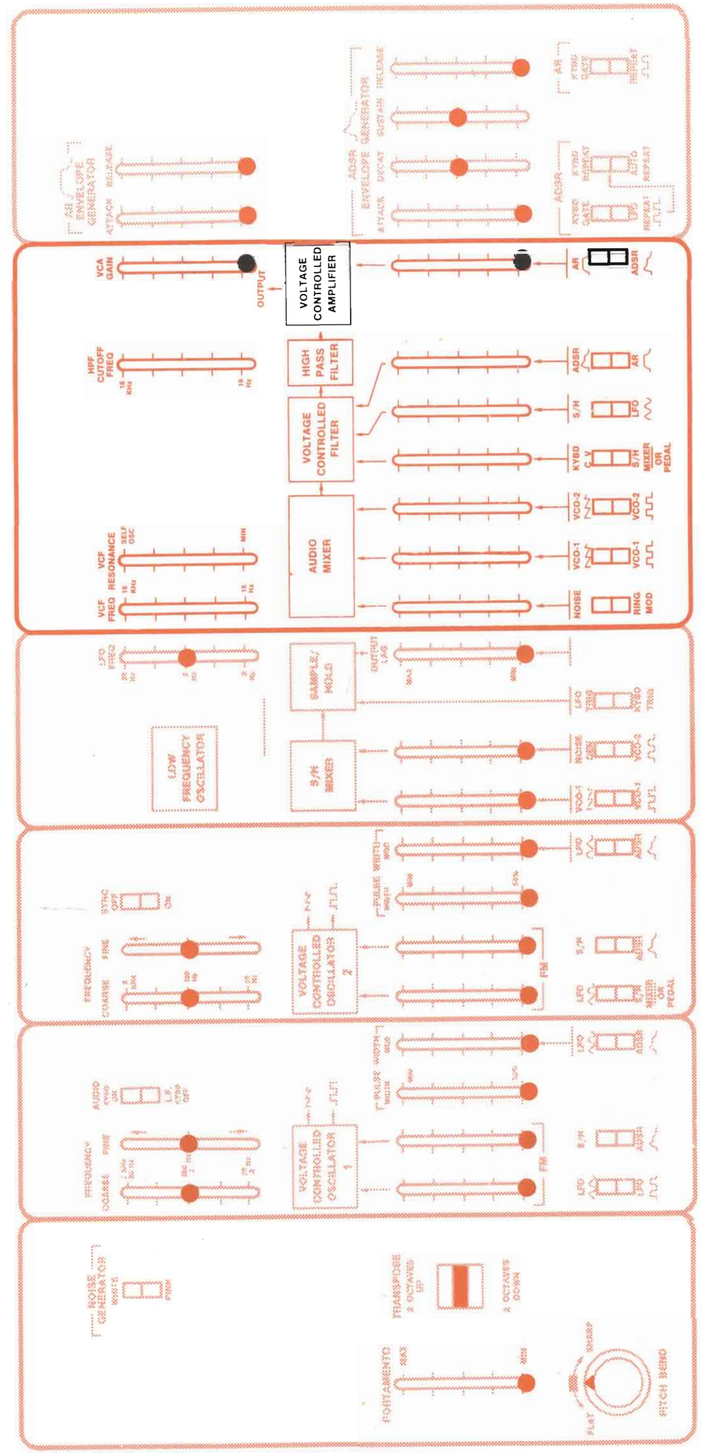


Figure 3.5.1. Voltage Controlled Amplifier.

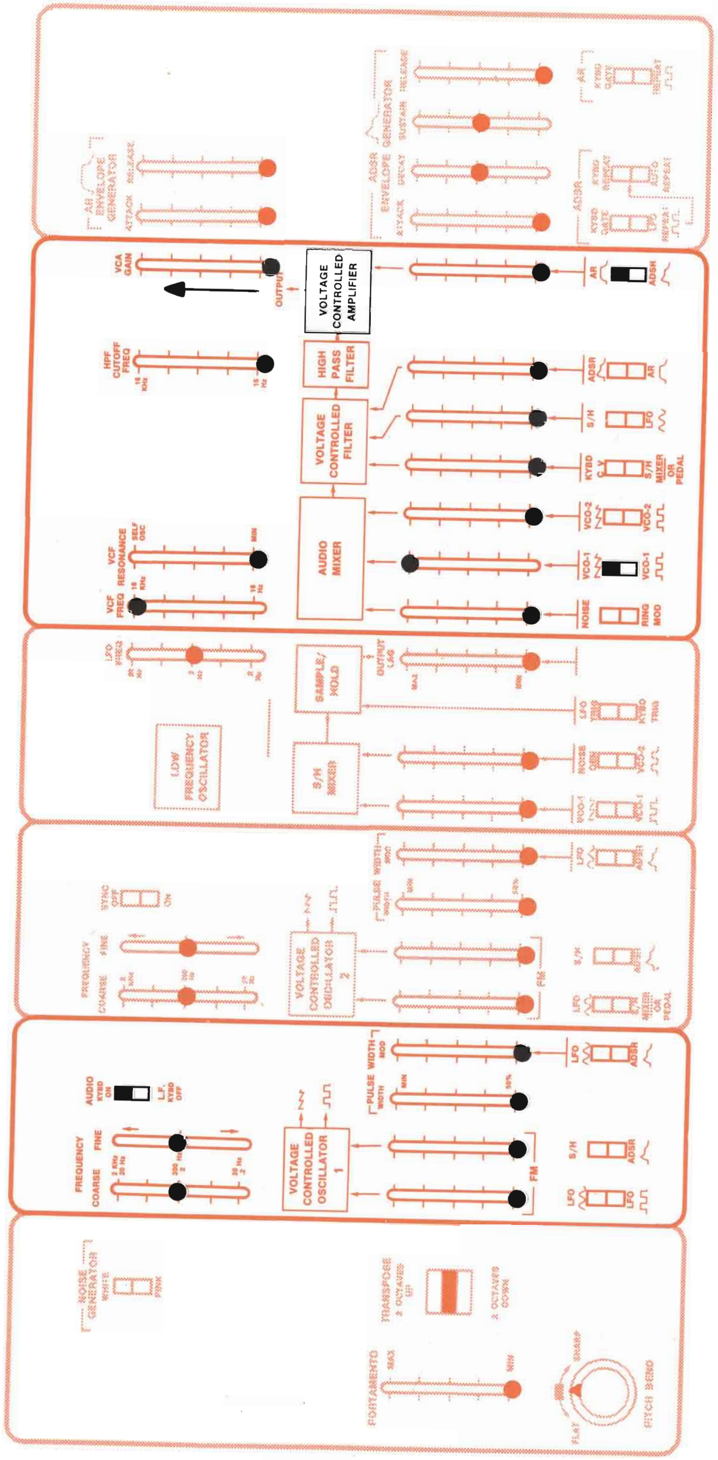


Figure 3.5.2. VCA experiment.

Experiment 1: VCA Gain

With the controls set as shown in Figure 3.5.2, you should not be hearing any sound. Gradually raise the VCA Gain attenuator to its highest position and the volume will increase accordingly. Lower the slider again. Note that you are only changing the amplitude (volume, in this instance) of the audio waveform you're hearing. You are not altering the harmonic content of the sound, as you would if you closed the VCF Freq slider. Prove this to yourself by doing just that. Raise the VCA Gain slider. Now, as you raise the VCF Freq slider, you'll hear the complexity of the audio signal increase. By the time you reach the top of the slider's range, you will hear the full harmonic content of the waveform. Closing it again will have the reverse effect.

The point here is that the use of the VCA, its functions and capabilities, cannot be freely interchanged for the use of the VCF. Each has its own particular functions, which when used together will permit you to create effects that neither could produce alone. The two following experiments demonstrate this.

Experiment 2: ADSR-Controlling the VCA

Set the controls of the Odyssey as shown in Figure 3.5.3. By opening the VCA Gain slider and closing the ADSR control attenuator into the VCA, you can prove that with the filter in oscillation you would have no control over the attack and/or duration of the audio signal produced. Then return to the position of Figure 3.5.3. Using the Keyboard Control Voltage to determine the pitch and the ADSR to control the VCA (providing an envelope to control the amplitude of the waveform), you can now "play" the filter with complete control.

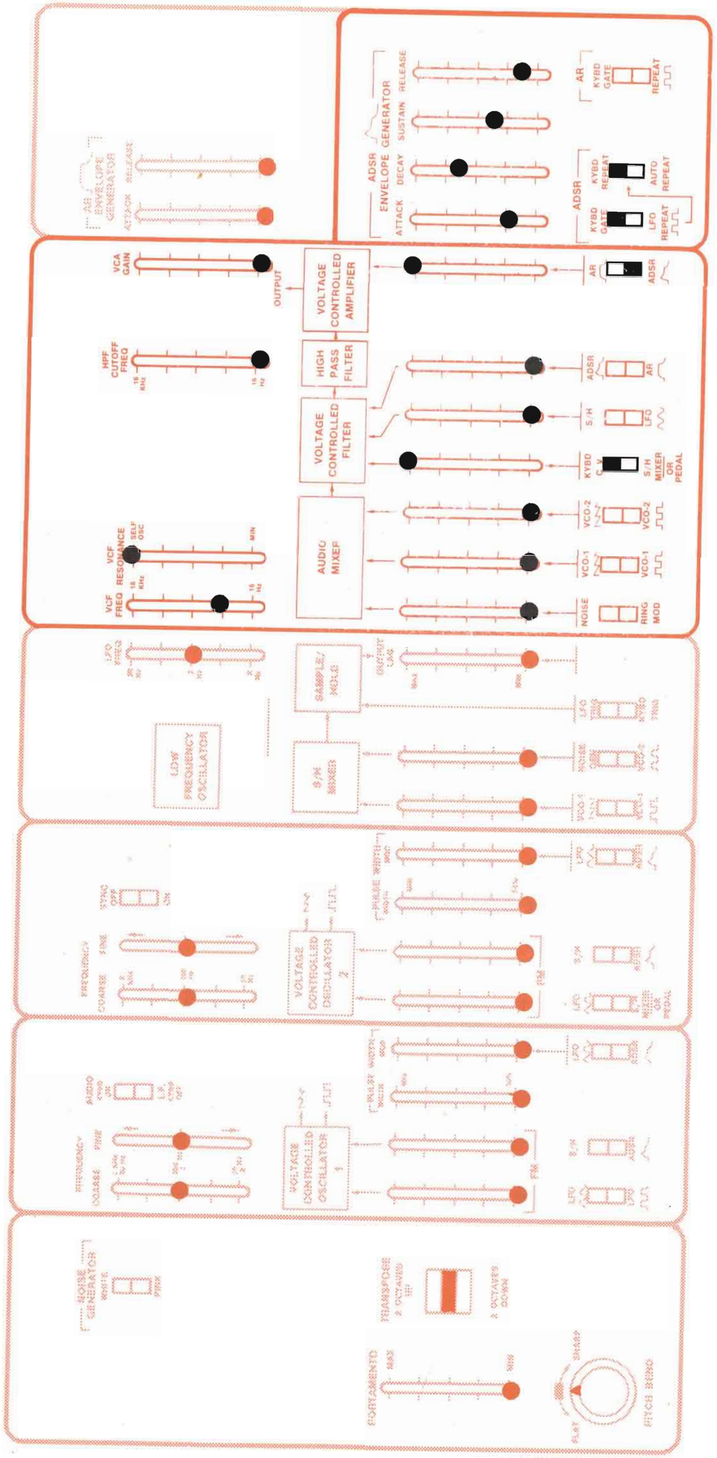


Figure 3.5.3. Basic patch for ADSR control of VCA.

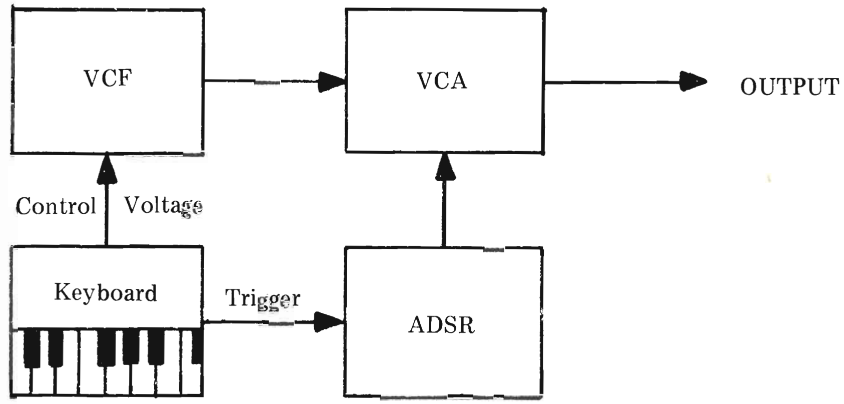


Figure 3.5.4. Block diagram of Figure 3.5.3.

Experiment 3: ADSR-Controlling the VCA and the VCF

The ADSR control of the VCA will be equally useful even when the filter is not in oscillation. There are times when you will want to leave the filter partially open, without hearing the continuous "bleed" of the frequencies not being filtered (Figure 3.5.5, next page).

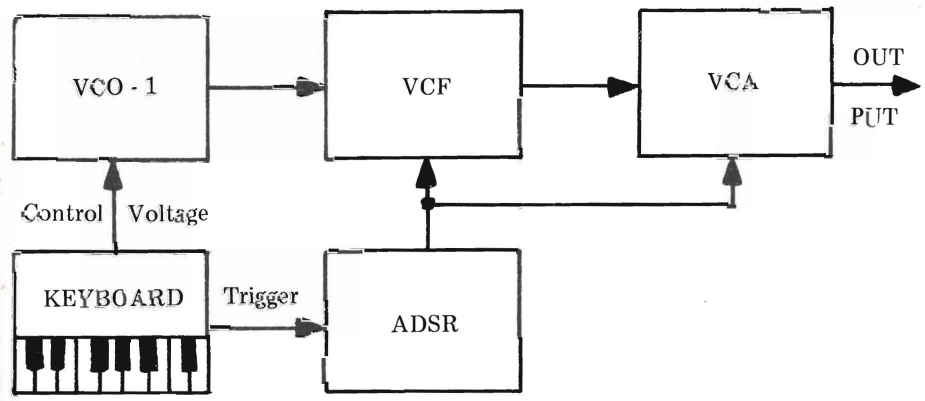


Figure 3.5.6. Block diagram of Figure 3.5.5.

Play a series of notes on the Keyboard. You will find that even with the long release time on the ADSR, the sound will die away completely. By opening the VCA Gain slider again, however, you will find that the filter is really partially open — without the VCA, such a filter setting would not produce an effect that would be nearly as pleasing, or as useful musically.

This partial block diagram (Figure 3.5.7) of the patch shown in Figure 3.5.5 shows the audio signal from VCO-1 passing through the VCF and VCA (the signal also passes through the HPF but is not affected because the HPF control is all the way down) and the control signals for the VCF and VCA both coming from the ADSR Envelope Generator.

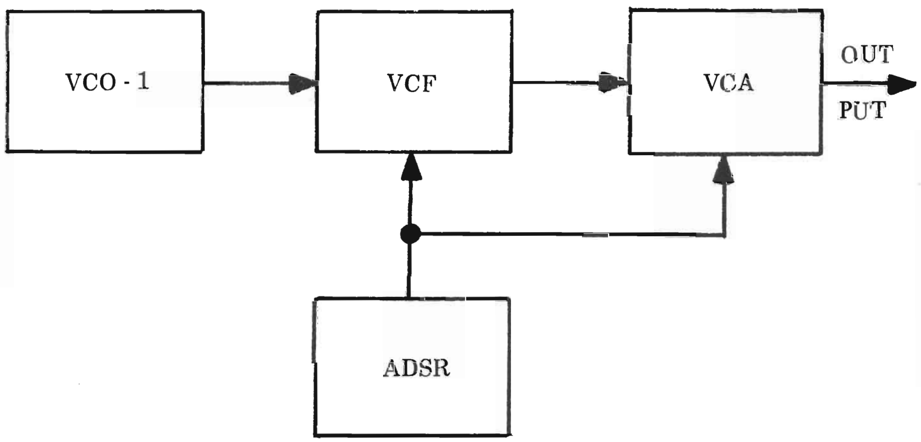


Figure 3.5.7. Partial block diagram of Figure 3.5.5.

Experiment 4: AR Control of VCA While ADSR-Controlling VCF

At times, you will want to have the option of controlling the VCF and the VCA with differently shaped envelopes. You'll note that both the VCF and VCA have attenuators and two-position switches (the one farthest right on Panel 5) permitting you to control either, or both, the VCA and VCF with the AR or ADSR generators.

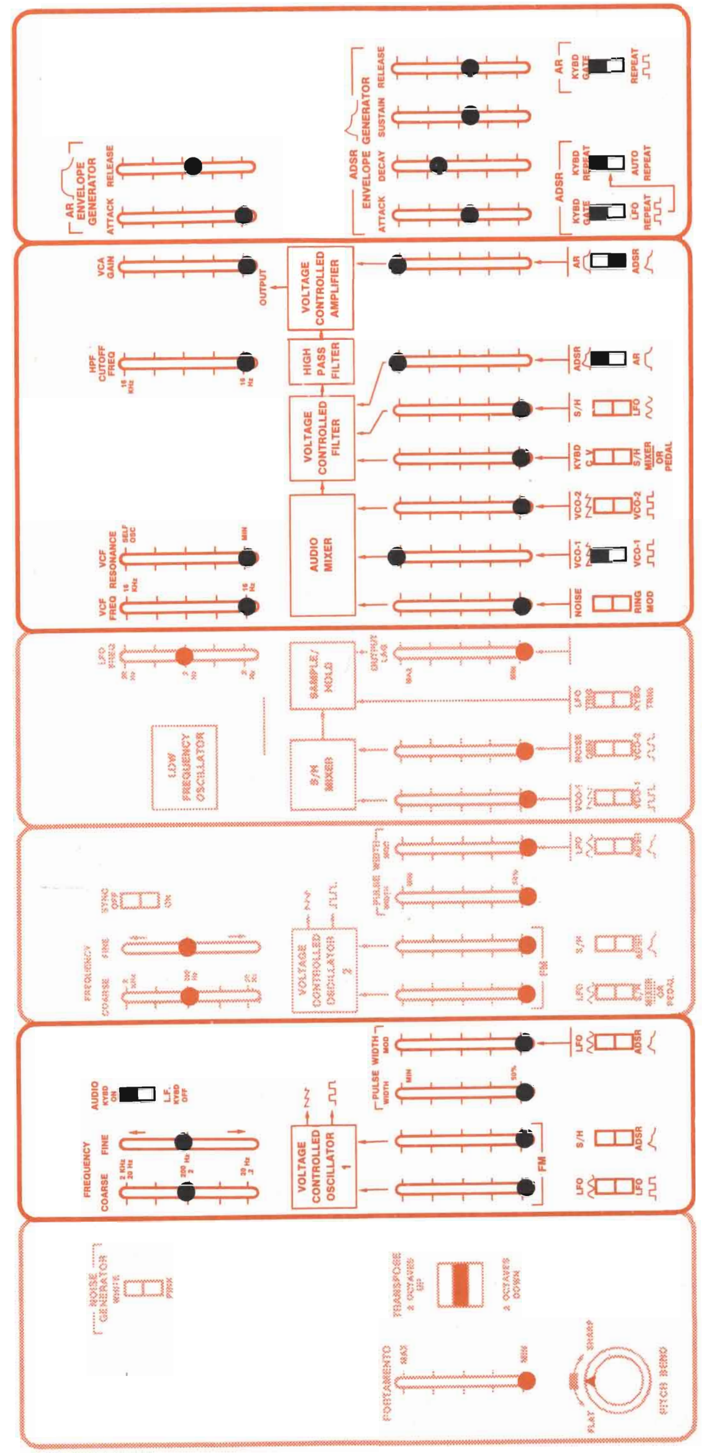


Figure 3.5.5. ADSR control of VCF and VCA.

Note in Figure 3.5.8 that the VCF is being controlled by the ADSR and the VCA by the AR. After playing a few notes using the keyboard, close the red AR Release slider all the way. While the ADSR envelope controlling the filter is still programmed for a long release, the AR controlling the VCA is overriding the release time by simply closing the VCA — just as you would turn off the volume control of a stereo. Though a record album might still by playing, you would no longer hear its sound coming from your loudspeakers.

Similarly, by slowing the attack of the AR, you can slow the attack of the sound, even though the filter is programmed by the ADSR for fast attack. Prove it to yourself by raising the AR attack slider.

We can represent the signal flow in this fairly complex patch by the following block diagram. Study this diagram and see how every interconnection shown on the block diagram is also shown on the Odyssey front panel (Figure 3.5.9).

You may have deduced by now that the VCA is working like a volume control that, when open, lets you hear what is coming out of the filter (VCF). Consequently, if you wanted to hear everything that was going on in the filter section (for instance, how the ADSR is opening and closing the filter), you must set the controls of the AR so that the VCA is "open" during the complete time of interest. In other words, if you had a very short attack time programmed on the ADSR, but a long attack time programmed on the AR, the ADSR would be finished with its attack (remember, the ADSR is opening the VCF) before the AR opens the VCA enough for you to hear what's going on in the filter. So if you want to hear that fast attack from the ADSR and VCF, you will have to open up the VCA quickly enough — and that means setting the attack time of the AR to be very fast also.

Similarly, if you have programmed the ADSR for a long release time so that the VCF will close slowly, you must also set the AR for a long release time so that the VCA will stay open long enough for you to hear the filter's long release.

Experiment on your own to establish in your mind the flexibility that this relationship between the VCF controlled by the ADSR and the VCA controlled by the AR can permit. You should also try reversing the position of the patch switches — control the VCF with the AR and the VCA with the ADSR. Note again that in either case you may also open the VCF Freq to any degree you desire, without continuous "bleed" of the unfiltered frequencies. Properly employed, based upon the understanding of the experiments above, the VCA becomes an extremely useful function in the electronic synthesis of numerous sounds and effects.

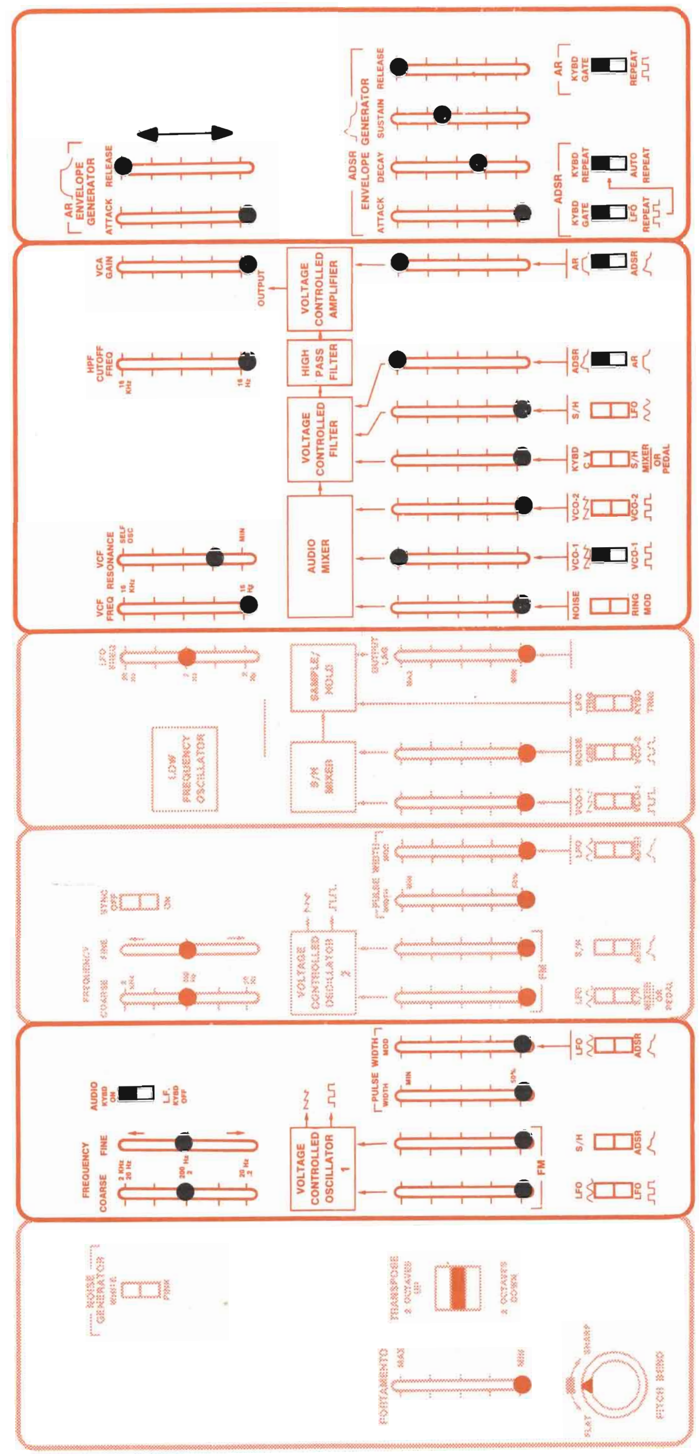


Figure 3.5.8. Basic patch for AR release-time experiments.

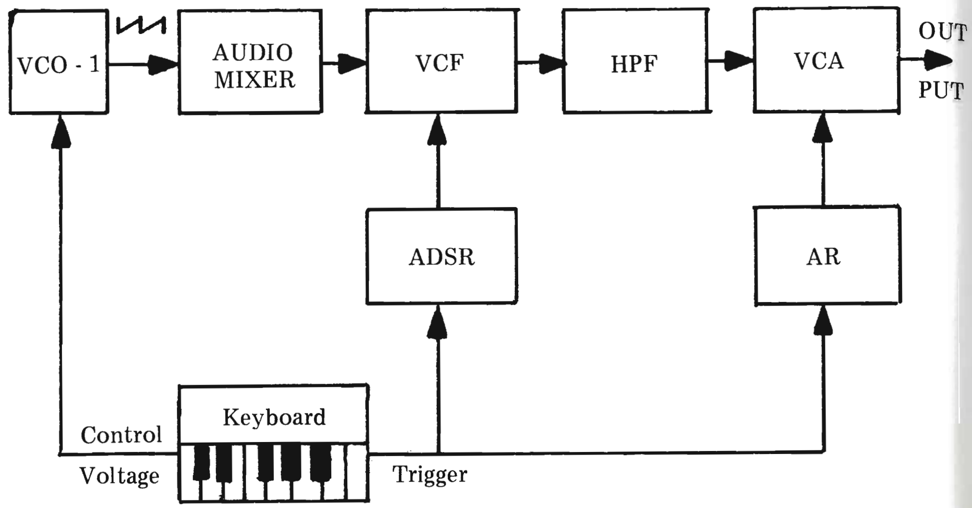


Figure 3.5.9. Block diagram of Figure 3.5.8.

Section 4: The Controllers

Having now examined both the signal sources and the signal modifiers of the Odyssey, we come to the third general category of the synthesizer's functions: controllers. You will be working again with a number of voltage-controlled functions used earlier in Part II: the AR and ADSR generators, the LFO, the Sample and Hold, VCO-1 in its low-frequency range, and the keyboard. At this point, however, we shall explore more fully the precise means by which these controllers operate to produce some of the effects you've created.

There are, of course, two methods of control: manual control and voltage control. When you use your hand to tune an oscillator, or to open the VCF or VCA, you are controlling these functions manually. The controller is your hand. However, each of these tasks can also be performed by a control voltage; hence, we have the voltage-controlled oscillators, filter, and amplifier. While your hand is a convenient control device, if you think in terms of the great number of controlling functions that must be performed rapidly in order to create even a simple sound, you'll see that voltage control is essential.

Consider, for example, what would be required to play a brief melody without the aid of voltage control. Certainly, it would be possible to play a relatively simple one by hand-manipulating the coarse-tuning slider of the oscillator being employed. Try it. You'll find, of course, that it requires some practice to "hit" the pitches (frequencies) required with any degree of accuracy. Moreover, you have no dynamic control (control over the loudness and softness), no expressive articulation of the individual notes (control of attack and duration), no control over the timbre (control of the harmonic content or tone color), and no way to go from note to note without "sliding" across all frequencies between any two pitches.

It becomes obvious, then, that we need the keyboard controller to provide accurate, instantaneous voltages for pitch determination; we require transient, or aperiodic, voltages that we can preshape and summon upon demand to control the amplitude, attack, and duration of various sounds; and finally, we often require voltages that extend our capacities even further by helping us control the controllers.

The next series of experiments is designed to acquaint you with the controllers that you have at your command on the Odyssey. In each instance, you will be voltage-controlling at least one particular function, and in most cases, more than one. As you perform each experiment, try to picture mentally the effect the control voltage is having upon the function being controlled. Think of how you would have to manipulate the controls manually to create the same effect.

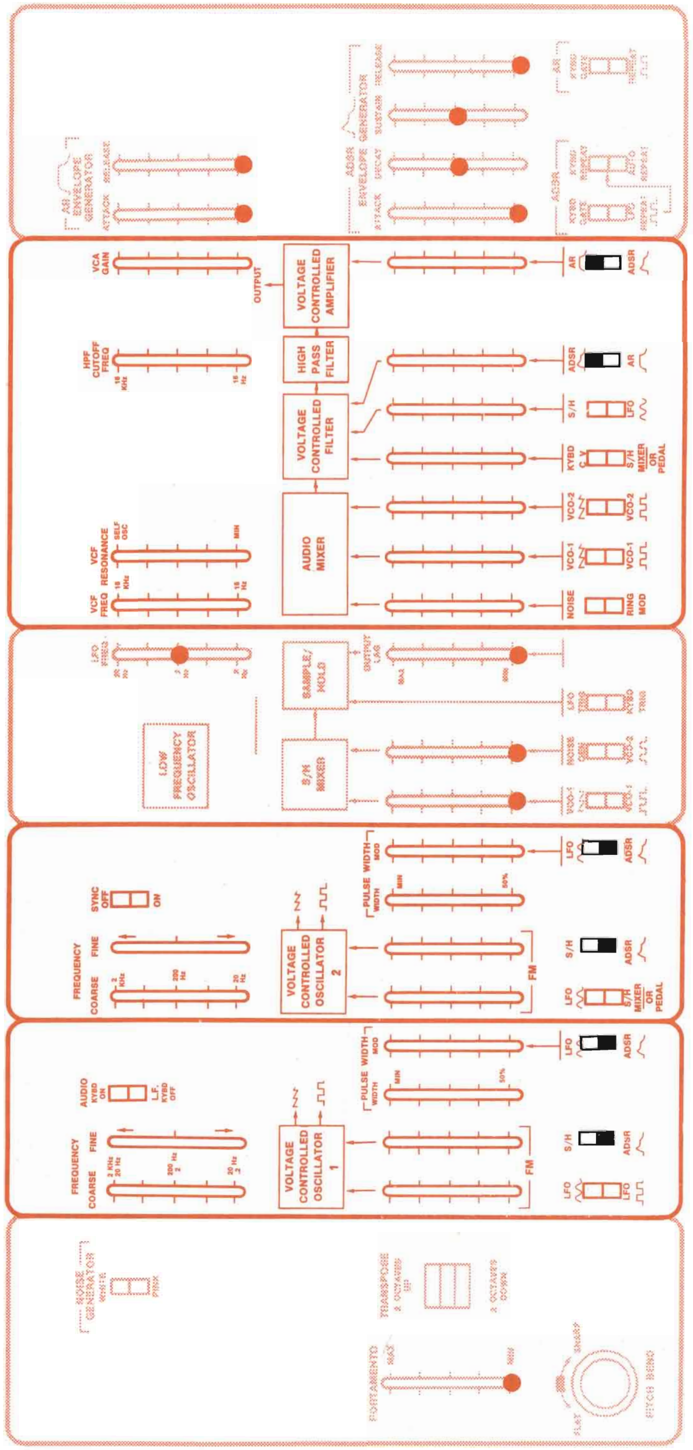


Figure 4.1.1. Basic connections for AR and/or ADSR control.

When you can do this, you'll be well on your way to not only an understanding but a genuine appreciation of the virtually limitless potential for creative synthesis that voltage control can and does provide.

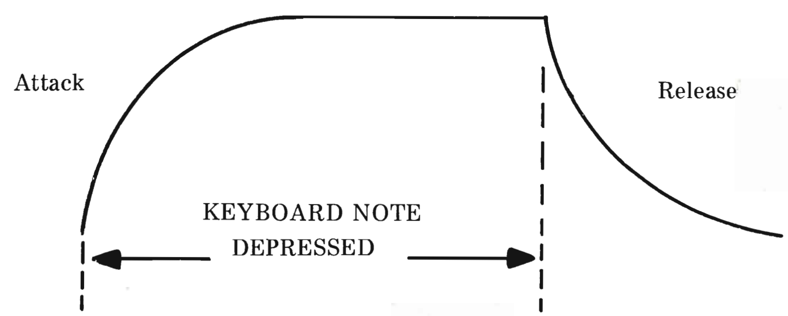
Lesson 1: The Envelope Generators

The AR and ADSR generators, in the case of each controller we'll examine, produce control voltages which you can connect to other functions of the Odyssey by using the two-position patch switches and corresponding attenuators. In the case of the AR and/or ADSR generators, the connections shown in Figure 4.1.1 can be made.

Note that there are eight possible connections — six for the ADSR and two for the AR. The ADSR can be used to control the frequency of both oscillators, the pulse width of both oscillators, the voltage-controlled filter, and the voltage-controlled amplifier. The AR can also be used to control both the VCF and the VCA. Furthermore, each envelope generator can control one or all of the functions to which it is connected simultaneously. You could conceivably use the ADSR to control frequency, pulse width, the VCF, and the VCA all at the same time.

Let's begin by exploring again the most common use of the envelope generators — controlling the VCF and VCA. We will use many of the same patches used in our discussions of the VCF and VCA, but we'll look at them from the point of view of the Envelope Generator.

Figure 4.1.2. AR envelope.



Experiment 1: ADSR Control of the VCF and VCA

Simply stated, the ADSR permits you to preshape an aperiodic waveform which will serve as a control voltage to open and close the VCF, VCA, or both. This permits you to control the way a sound begins, how long it lasts and how it sounds (the timbre) while it lasts, and finally, how long it takes to fade away. You shape this control voltage through the use of the four red control sliders in Panel 6.

1. *Attack.* How a sound begins. Most sounds have their own characteristic attack. A piano produces a sound having a relatively fast, immediate attack when a key is struck. A flute has a less sudden attack. A violin also has a gradual attack when a bow is drawn across the strings.

2. *Decay.* After the attack, a natural fading away of the sound occurs. A piano note begins to decay immediately; a guitar note decays more slowly at first. An organ note does not decay at all until the key being played is released.

3. *Sustain.* The harmonic content of the audio wave varies from stage to stage, from attack to release. The sustain slider permits you to control: (1) whether the initial decay will be halted before the final release, and (2) if so, what the harmonic content of the waveform will be during this period. For instance, a trumpet sound normally has a slight "overshoot" or emphasized attack. Expressed another way, the sound begins with a burst which drops back to some other level for the duration of the note. The Sustain control will affect the level to which the sound drops back after the initial attack on the sound. If the Sustain control is set to minimum, the sound will die out completely after the attack. If the Sustain control is set to maximum, there will be no "overshoot" and the sound will be at its loudest as long as you hold down a note.

4. *Release.* The final fade-time of the sound is controlled by the release slider. As mentioned above, a guitar string would have a rather long release time since it requires some time to cease vibrating, unless, of course, the string is muffled by the player's hand. An organ tone, however, is gone virtually the instant you release the key. While you can sustain it indefinitely by simply holding down the key, its release time is extremely quick.

The following diagram (Figure 4.1.3) expresses the operation of the four ADSR controls graphically. Since the main job of the envelope generators is to control the attack and decay characteristics of sounds (as we shall see, the envelope generators also have some less important applications), it is convenient to describe the operation of the ADSR controls by relating them to the sounds.

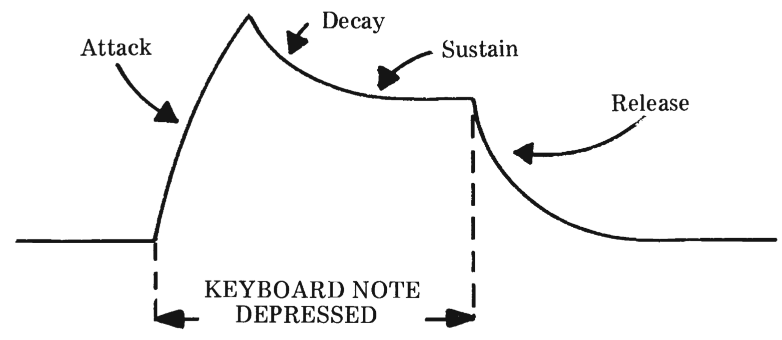


Figure 4.1.3. ADSR envelope.

Experiment 2: Settings of the ADSR Controls

Try the settings of the ADSR, using the patch shown in Figure 4.1.4. See if you agree that in terms of attack, decay, sustain, and release, these envelope settings are representative of the characteristic sound of each instrument. Experiment with the basic settings to see if you can refine them to produce an even more realistic envelope, based upon what sounds right to you.

Experiment 3: AR Control of the VCF and VCA

The AR generator functions much the same as the ADSR; the primary difference is that it has fewer controls and creates a less elaborate control voltage, one which permits you to control only the attack and the release of a sound. For this reason, the AR generator is somewhat less flexible than the ADSR, and is generally used for different purposes. With the AR alone you can create a perfectly acceptable bowed string envelope, or you can create the same organ effect that was created using the ADSR in Experiment 1. Set the Odyssey up according to the patch in Figure 4.1.4, except to change the position of the ADSR/AR switch directly under the HPF box to the AR position (down). Now the VCF will be controlled by the AR rather than the ADSR. Try the following envelopes.

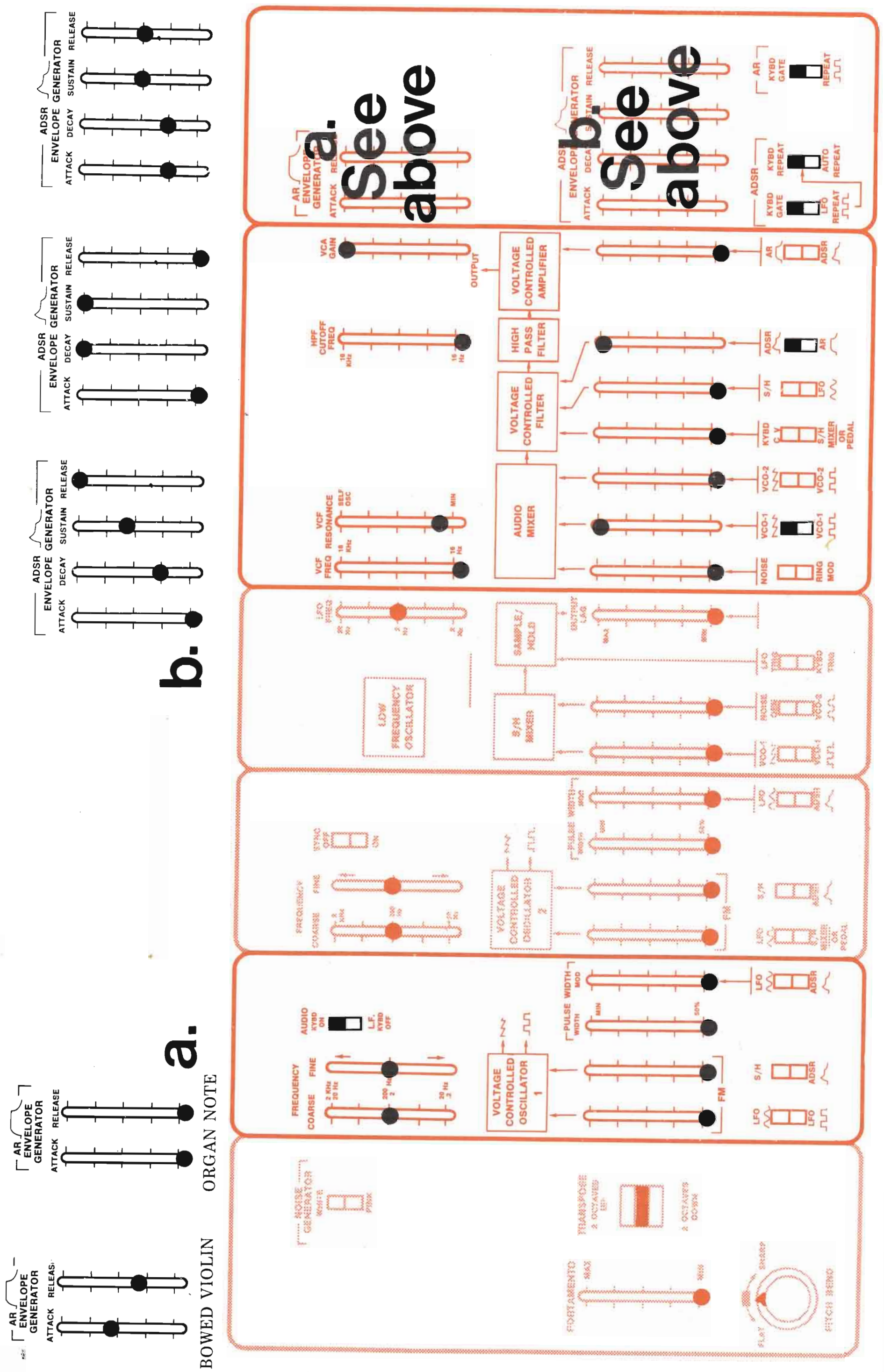


Figure 4.1.4. Envelope generator experiments.

Experiment 4: ADSR Control of Pulse Width

As you discovered in earlier experiments, as you change the width of the pulse waveforms generated by both of the voltage-controlled oscillators, you change the sound of the waveform. You have already used the Pulse-Width slider, controlled manually, to demonstrate this fact. That sort of control, however, once the width is set, still results in a static waveform having a steady unchanging tone. To prove this, set up the patch in Figure 4.1.5, next page. You may move the pulse-width (blue) "Width" slider of the VCO-1 to any position, but as soon as you stop moving it, the sound stops changing.

Why does this matter? It matters because if you looked at the waveforms of many conventional musical instruments on an oscilloscope, you would see that waveforms are not constant but are always changing as the sound sustains and then dies away. Such waveforms are *dynamic* waveforms; their timbre is constantly changing, even though these changes may be slight. By ADSR-modulating (changing) the pulse width of the oscillator, you can create a dynamic waveform, changing in harmonic content according to the shape of the ADSR envelope. To demonstrate, simply raise the pink attenuator marked "Mod" under the Pulse Width bracket on VCO-1. Listen as the sound of the waveform changes when you press a key. The change is being created by the ADSR control voltage which is, for convenience, being triggered by the Keyboard.

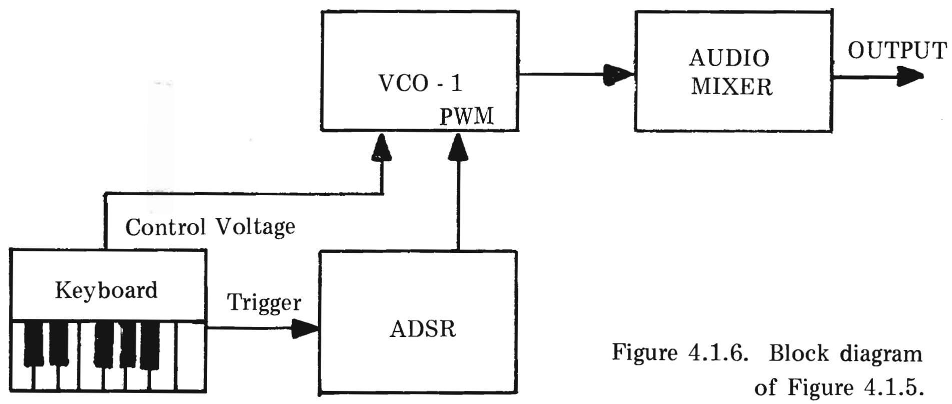


Figure 4.1.6. Block diagram of Figure 4.1.5.

You'll find dynamic waveforms extremely useful in recreating realistic instrumental effects such as a guitar, bass, or piano. Try the Electric Bass patch shown in Figure 4.1.7. Vary the pink Mod attenuator on VCO-1 to hear the effect that is being created by the ADSR modulation of the pulse width. You may find that you would prefer a slightly different bass effect than that shown in the patch given here.

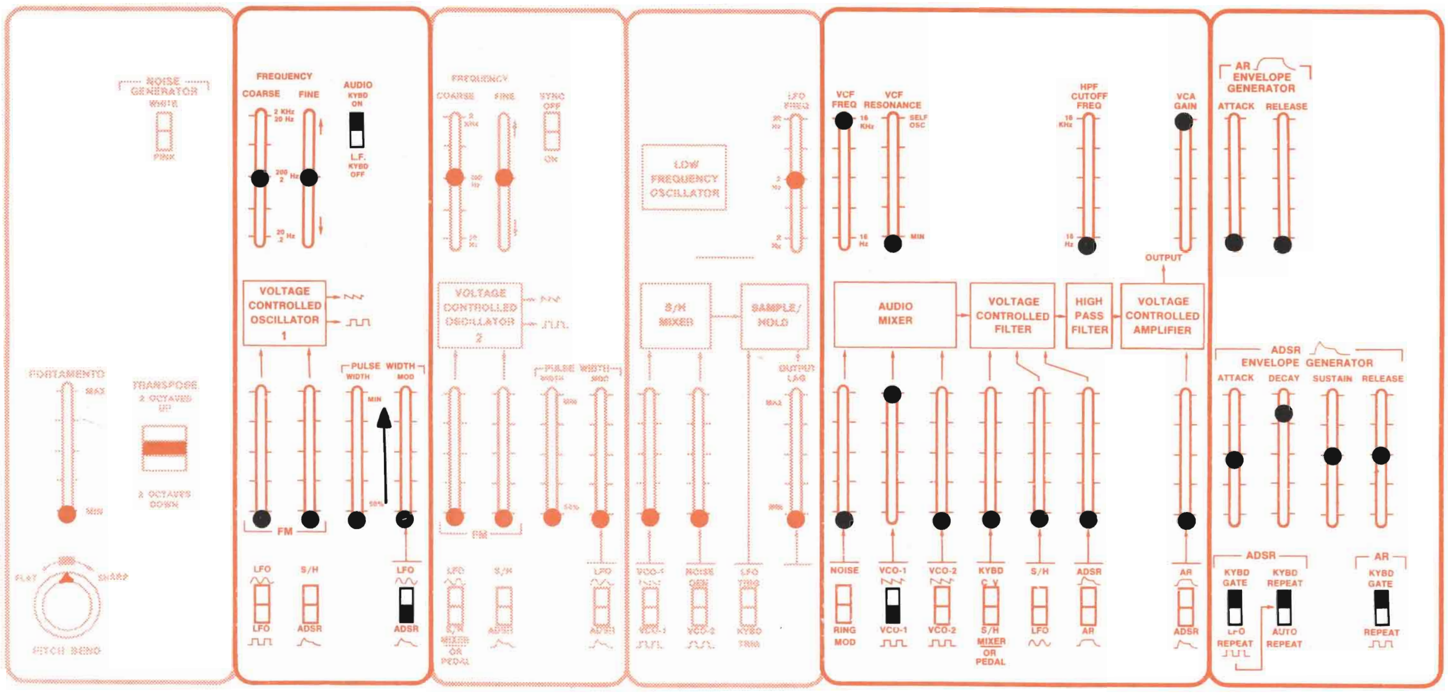


Figure 4.1.5. Basic patch for ADSR control of pulse width.

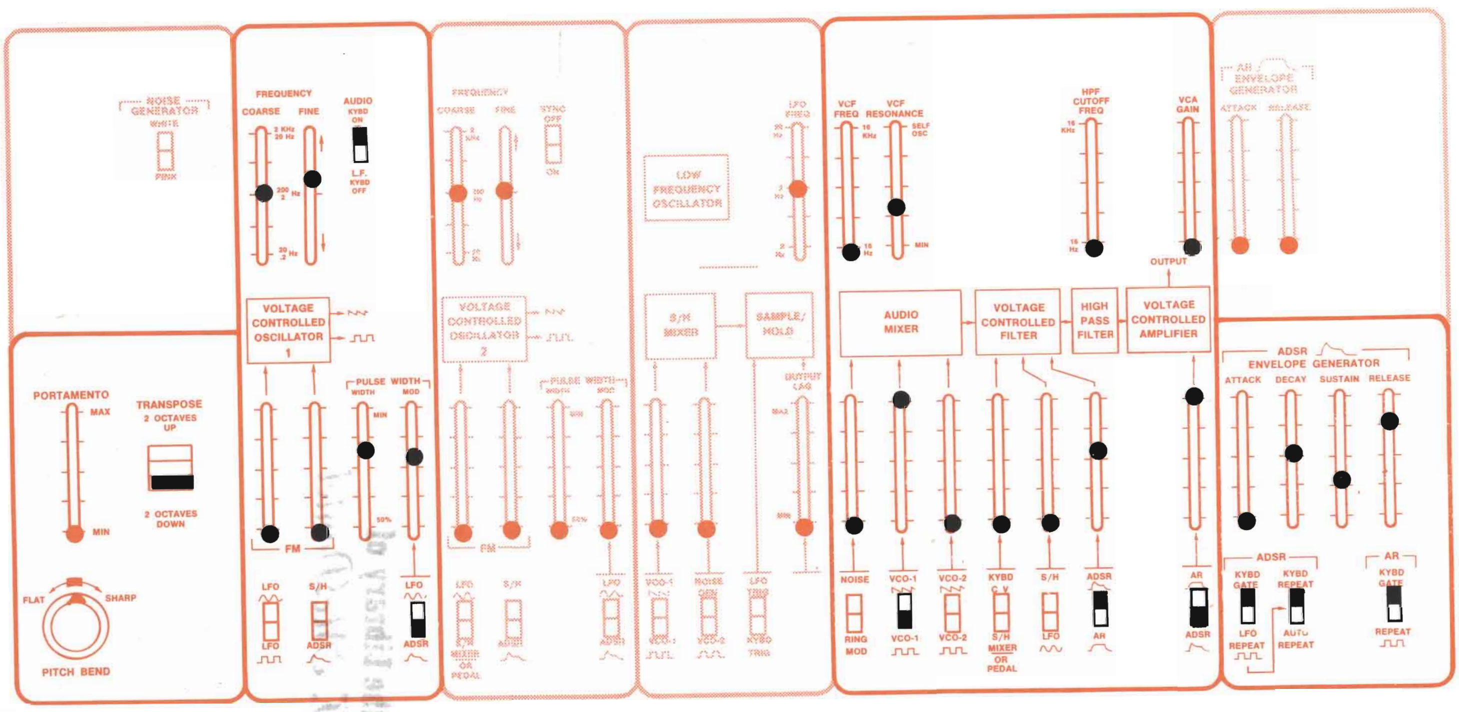


Figure 4.1.7. Patch for electric bass.

Experiment 5: ADSR-Controlled Frequency Modulation

The final experiment in this lesson utilizes the ADSR voltage to control the pitch of VCO-1. You'll find that this control voltage, when combined with the prewired control voltage from the keyboard, creates an unusual effect — much like an instrumentalist or vocalist "sliding" up to every pitch being played or sung.

The key to this patch is the setting of the ADSR slider controls. These will determine what effect the ADSR will have upon the pitch of VCO-1, and, with a bit of experimentation, you'll find that many ADSR settings do not produce particularly useful pitch effects. To guide your experimentation, you will find:

1. The attack slider determines the "rise time" that will elapse as the oscillator goes from the basic frequency (pitch) that you have set manually, to the pitch that the ADSR will ultimately "hold" until you release the key on the keyboard.
2. The sustain slider determines the pitch the ADSR will hold for you.
3. The release slider determines how fast the pitch will fall from the ADSR-controlled second pitch to the original pitch set manually.

While the control settings will vary slightly from one Odyssey to another, the patch shown in Figure 4.1.8 is programmed to do the following:

1. The ADSR modulation of the VCO-1 frequency will have a relatively fast "rise time," as determined by the attack slider.
2. The yellow attenuator on VCO-1 should be fine-tuned to limit the pitch rise to exactly one octave. Adjust the controls on your Odyssey accordingly.
3. The release time will be relatively long, during which you'll hear the pitch of VCO-1 fall back to the original octave determined by the manual setting of the frequency controls.

Lesson 2: The Low-Frequency Oscillator

The Low-Frequency Oscillator (LFO) belongs to the group of components known as controllers. The output of the LFO is never used as an audio signal, but rather is used exclusively to control other functions of the Odyssey. In addition to the pink LFO Freq slider shown in Figure 4.2.1, locate all other LFO-related controls as illustrated.

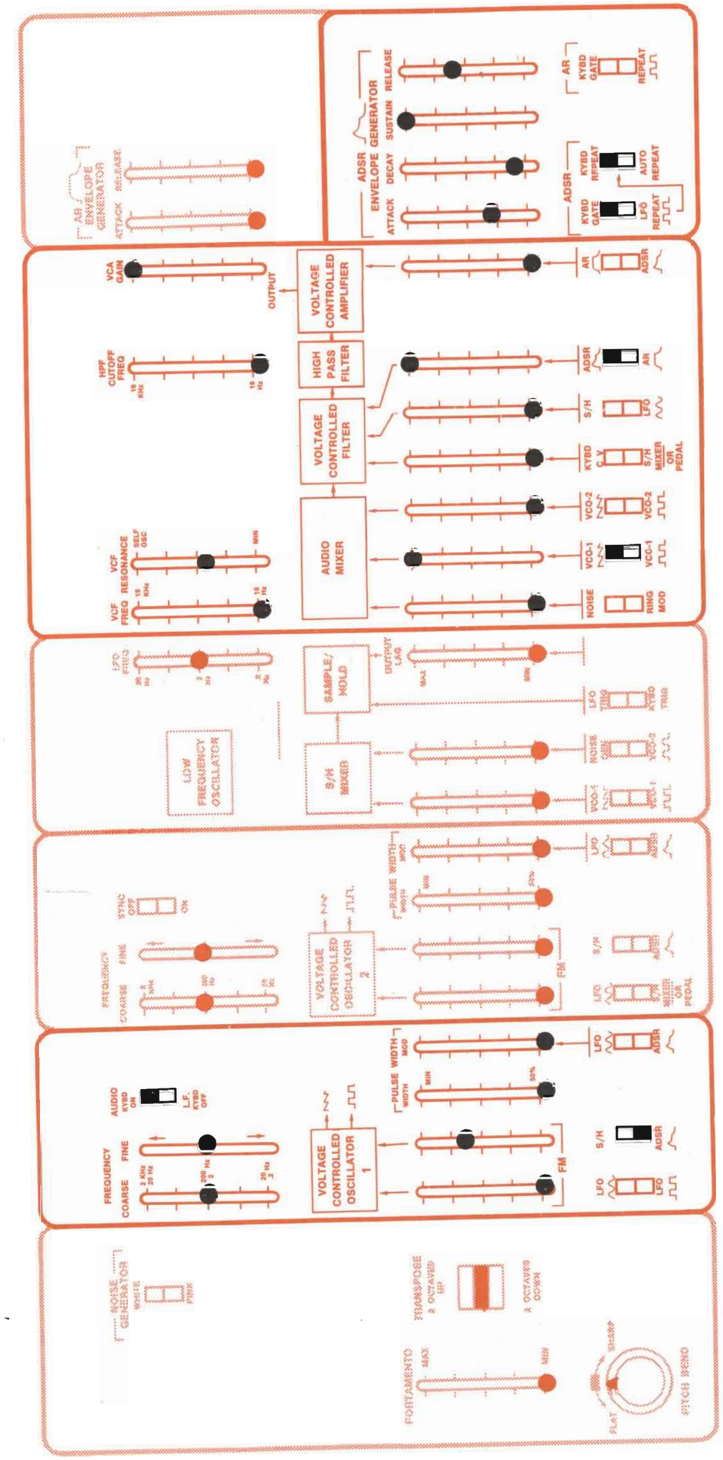


Figure 4.1.8. Patch for ADSR control of frequency modulation.

Experiment 1: Vibrato.

Make sure that the LFO switch under the pink FM slider of VCO-1 is in the sine wave (∿) position; then, gradually raise the pink attenuator above that switch — you'll hear a vibrato. As the slider is moved to higher positions, the slight pitch change brought about by modulating the frequency of VCO-1 with the LFO becomes greater. Lowering the slider has the opposite effect.

The vibrato effect you heard was created by the control voltage supplied by the LFO. In this case, the control voltage was in the form of a sine wave. At this point, it would be a good idea to study Figure 4.2.2. Notice that the sine wave is balanced on either side of the zero-volts line, while the square wave produced by the low-frequency oscillator is "positive-going" only.

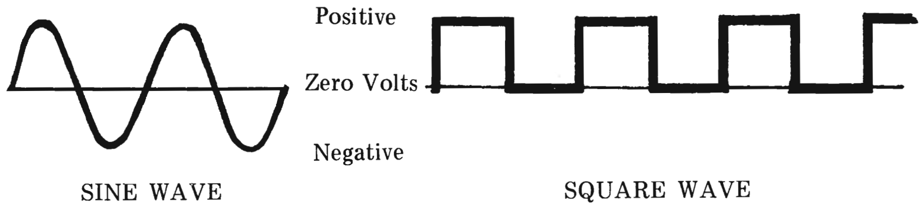


Figure 4.2.2. Comparison between sine wave and square wave.

It is important to understand this if you are to use the LFO most effectively: when controlling an oscillator, a positive-going voltage can only drive the pitch up (raise the frequency), while the sine wave which goes positive and negative will alternately both raise and lower the frequency relative to the oscillator's initial setting. Thus, to create a vibrato, which is a cyclical one-pitch deviation (alternately above, and then below, the original frequency) you must have a control voltage which goes both positive and negative, as shown above in Figure 4.2.2. The sine wave answers this need by providing a smoothly changing voltage which, when applied to the control input of either oscillator, creates a perfect vibrato.

As you slowly raise the pink attenuator, you can increase the vibrato "depth" to the point where the LFO is driving the oscillator both up and down more than an octave in each direction. Before leaving Experiment 1, try different attenuator positions in order to "tune" the deviation from the oscillator's basic pitch. While a normal vibrato deviation is less than one semitone and usually repeats at the rate of 4 Hz to 8 Hz (LFO Freq setting) many interesting new effects are possible with wider deviations and faster and slower rates. Try varying the pink LFO Freq setting to get faster and slower vibrato rates.

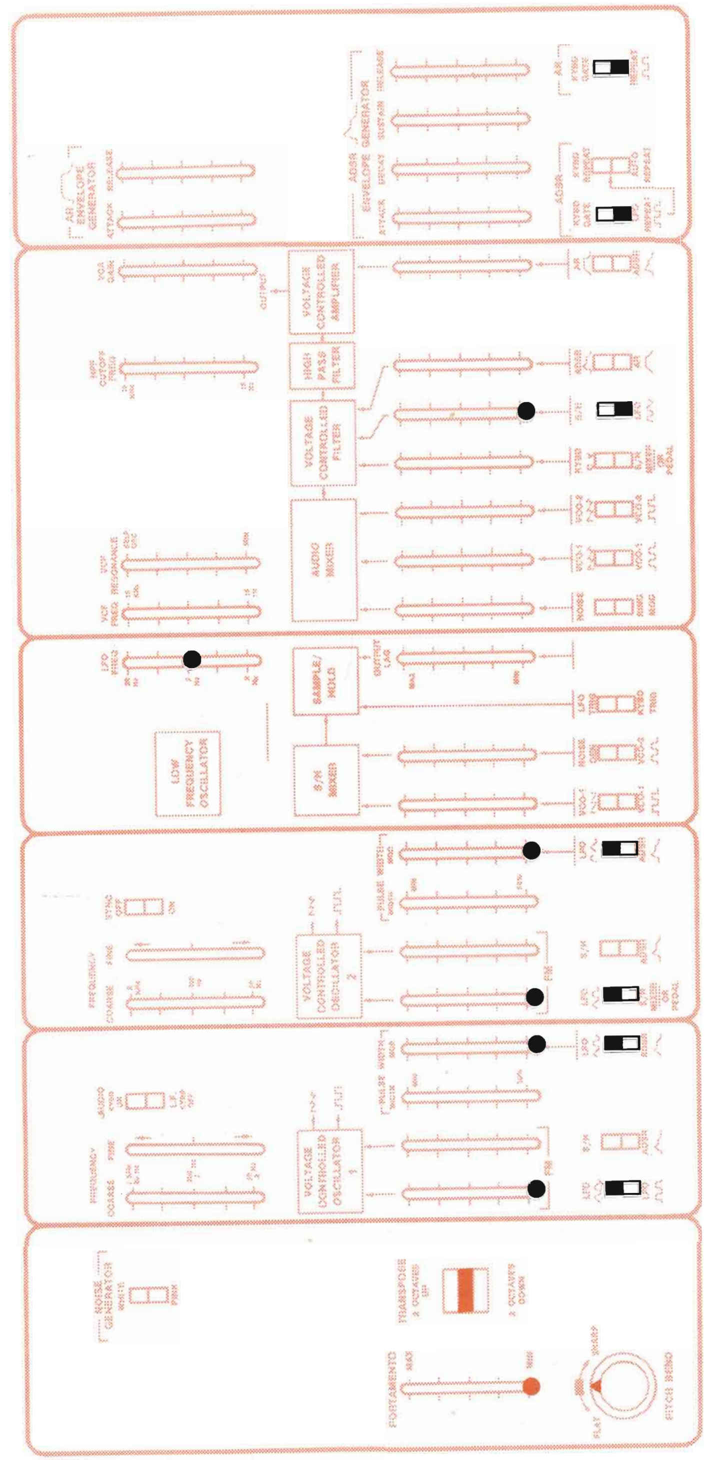


Figure 4.2.1. LFO connections to other functions.

Experiment 2: Trill

Set the controls exactly as you did at the beginning of the previous experiment (Figure 4.2.1), with one exception: move the two-position patch switch on VCO-1 from LFO sine down tc LFO square (⎍⎍). Raise the frequency-modulation slider as you did before. This time you will hear a trill. Simply stated, a trill is the alternation of *two* pitches. If you wish to increase or decrease the rate of the trill, use the pink LFO Freq slider; to increase or decrease the interval (distance) between the two notes of the trill, adjust the pink control-voltage input attenuator. Musical intervals from a half step or less on up to well beyond an octave are possible.

The manner in which the LFO is controlling the oscillator is roughly the same as in Experiment 1. The only difference is that this time, since the square wave is positive-going only, the pitch goes up to the level determined by the LFO and then simply falls back to the pitch originally set. Listen to prove this to yourself. One of the two pitches in the trill will always be the same pitch you originally set using the coarse- and/or fine-tuning sliders of the oscillator.

You should also have noticed that the audible effect of the LFO on the oscillator, in both this experiment and the previous one, conforms to the visual representation of the shape of each control waveform used. When the sine wave was employed, the pitch of the oscillator gradually rose and fell in a smoothly changing series of frequencies; when the square wave was employed, the pitch rose immediately from the original pitch, held its level, and then fell suddenly back to the first level — just as the shape of the square wave rises, flattens, and falls.

Experiment 3: Voltage-Controlled Pulse Width Using the LFO

Set the Odyssey controls as shown in Figure 4.2.3. When the Mod slider is raised, it now becomes possible to produce a voltage-controlled pulse-width change, the rate of which is controlled by the LFO Freq slider. While you may find ADSR control of the pulse width (as performed in Experiment 3 of the previous lesson) more useful, experiment with the use of a slow sine wave to create a continuously changing timbre of the oscillator as you're playing. Some interesting effects are possible.

As with all voltage-controlled functions, you are again simply applying a given voltage — determined by the position of the control input attenuator — to a voltage-sensitive function.

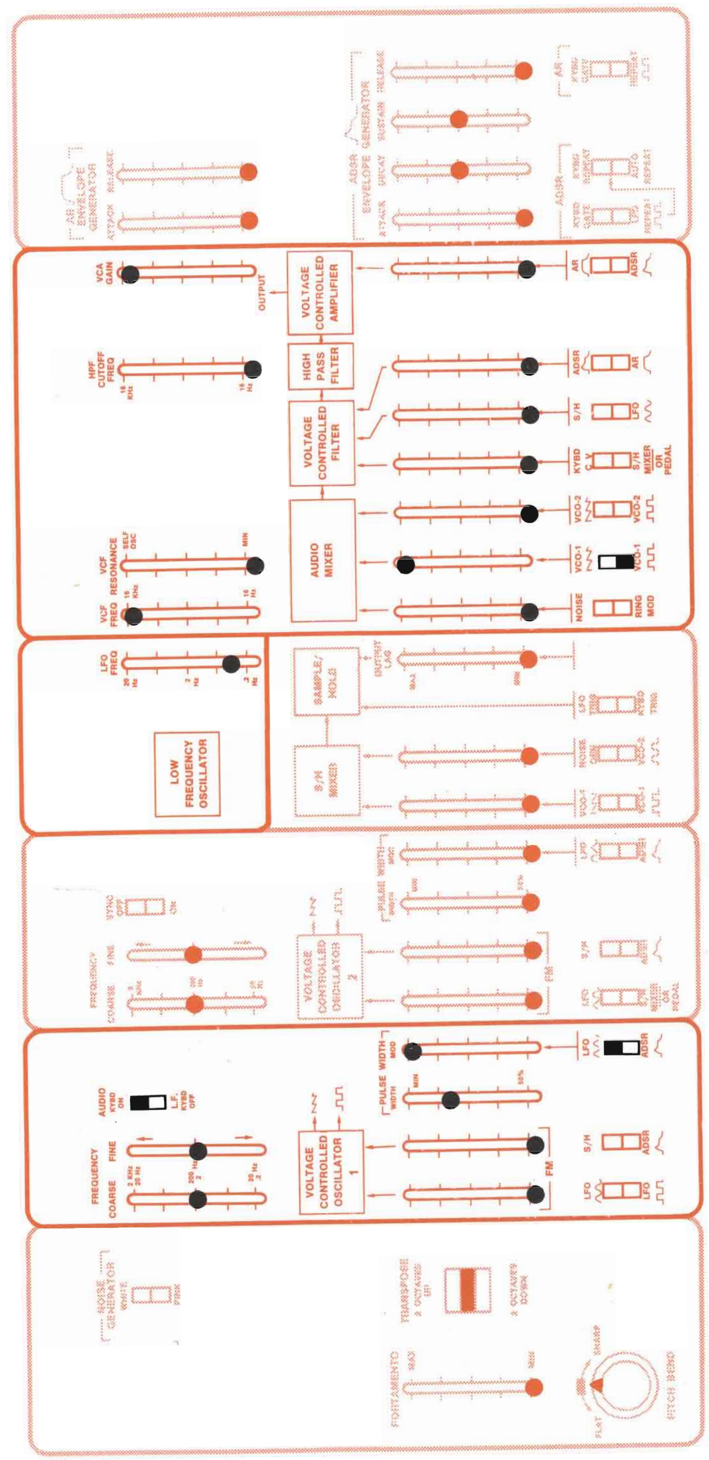


Figure 4.2.3. Basic patch for LFO control of pulse width.

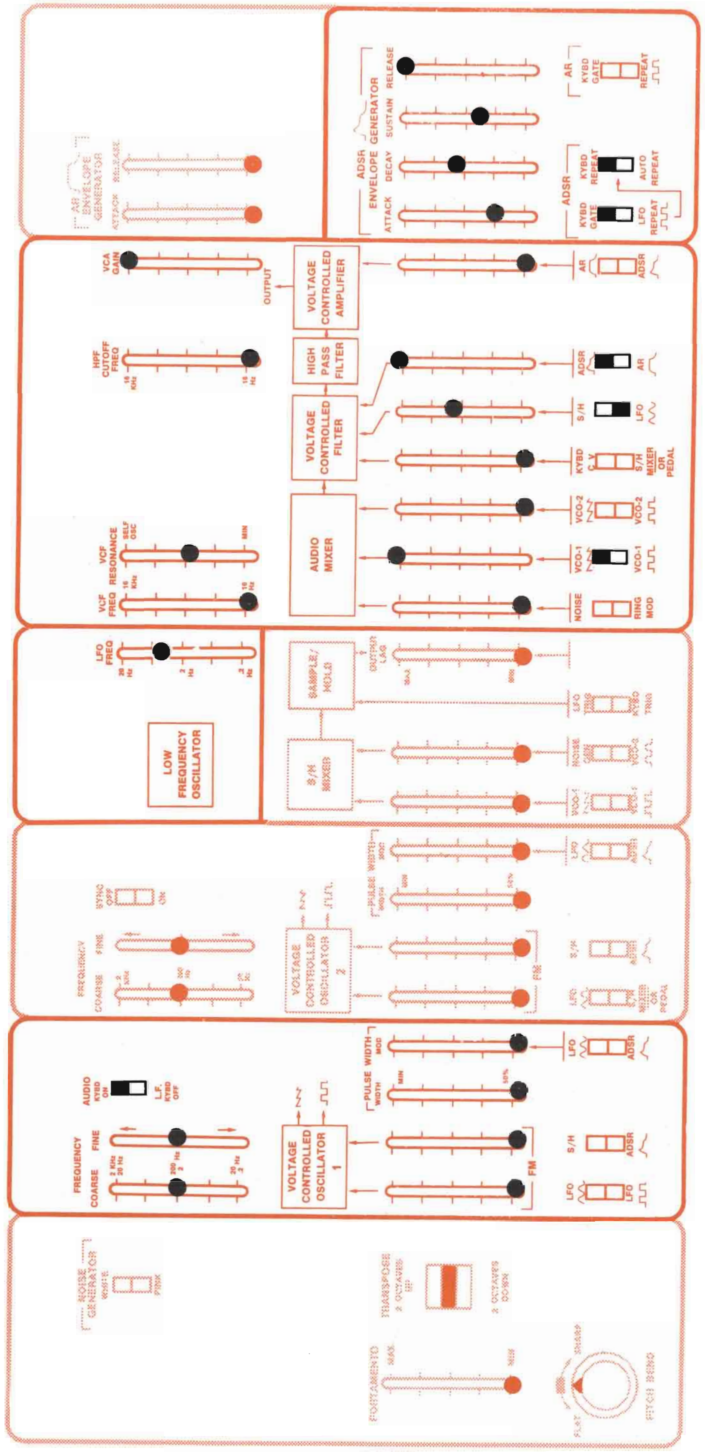


Figure 4.2.4. Modulating the VCF with the LFO to create tremolo effect.

As you spend more and more time with the synthesizer, try to conceptualize in this manner. Work at visualizing what is happening, and why. You will find that after only a short time you will develop a "library of mental pictures" that will greatly aid your control and understanding of the synthesizer.

Experiment 4: Controlling the VCF with the LFO

You'll recall that in Section 3, Lesson 1, Experiment 5, you created a tremolo effect by using the pink LFO slider to control the VCF. This rapid opening and closing of the filter creates the tremolo effect. Try the patch shown in Figure 4.2.4.

Experiment with different settings of the black VCF Resonance control and the yellow S/H-LFO ∿∿ attenuator. As you have probably guessed, you are again simply using the LFO to generate a control voltage, which, when applied to the control input of the VCF, opens and closes the filter at the rate determined by the LFO Freq slider.

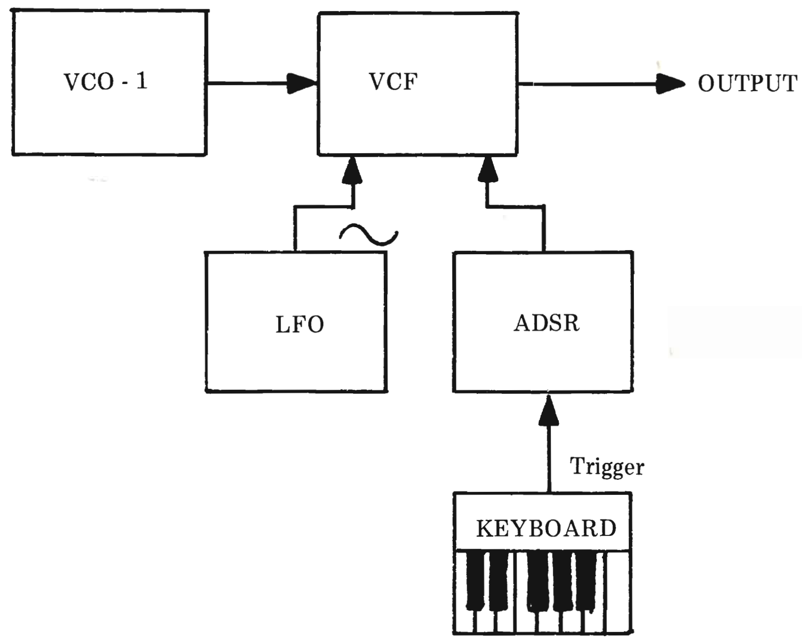


Figure 4.2.5. Block diagram of Figure 4.2.4.

Lesson 3: Sample and Hold

The Sample and Hold circuit of the Odyssey produces stepped output voltages by systematically sampling portions of the waveforms(s) which are routed to its signal input. The voltages which result from the action of the Sample and Hold circuit can then be used to control VCO-1, VCO-2, and/or the VCF. As you've certainly found from the brief experiences with the sample-and-hold function you've already had in Part II, some extremely interesting audio effects can be created by utilizing the Sample and Hold output voltages to control the oscillator(s). In this lesson, you will discover how these control voltages are created.

As its name implies, the Sample and Hold circuit does two things: (1) it samples, at the command of the LFO or the keyboard, the waveform that is being fed into the S/H Mixer (Figure 4.3.1); and (2) after the command to sample is past, the Sample and Hold circuit will hold the voltage level of the last sample taken until another sample command is presented to the circuit. The series of control voltages thereby produced can then be routed to the VCOs or to the VCF for control purposes. Most often, the voltages produced by the Sample and Hold are used to control the pitch of the oscillator(s) and/or the filter in oscillation.

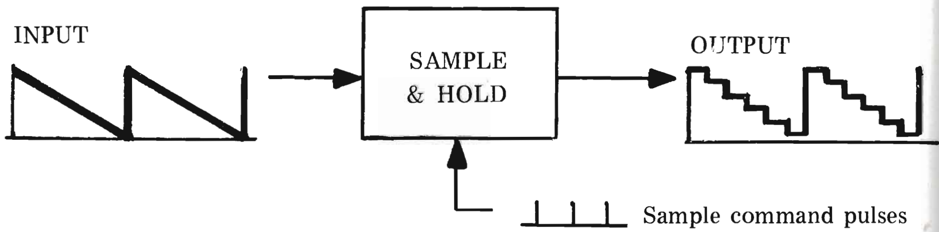


Figure 4.3.1. Diagram of the S/H function.

Now that you have an overview of this circuit's function, let's examine a visual representation of precisely how its control voltages are produced (Figure 4.3.2). Study the left-hand diagram for a moment. You'll notice that the output voltage (shown at the bottom) assumes the same voltage as the input signal at the precise instant the input signal was sampled. Moreover, you'll see that while the voltage of the input signal continues to rise, the voltage output of the Sample and Hold maintains the level of the previous sample until

the next sample is taken. You'll see that the same holds true for the illustration on the right; the only difference is that a random, aperiodic waveform (noise) is being sampled. As you might expect, sampling the noise wave produces output voltages which are random in nature, while sampling the periodic waveforms of VCO-1 and VCO-2 will produce repetitive patterns that can be controlled by the frequency controls of the oscillator providing the input signal.

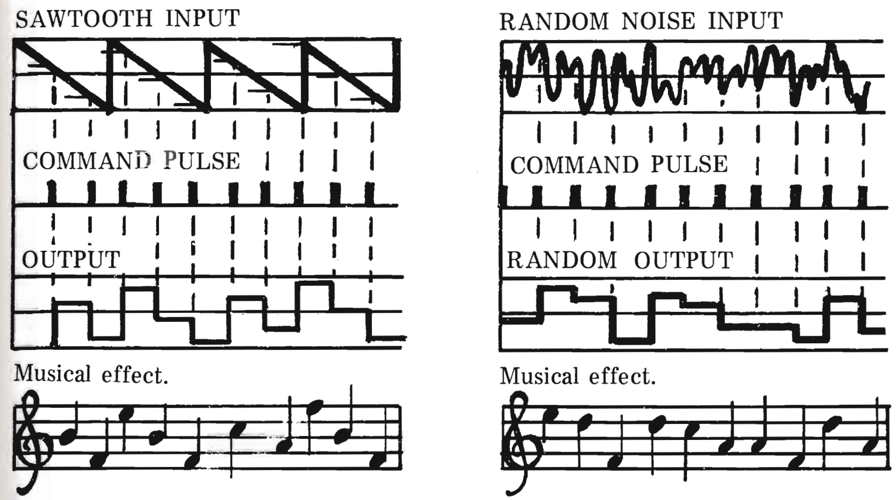


Figure 4.3.2. Diagrams of S/H operation.

Experiment 1: Sampling a Low-Frequency Sawtooth to Produce a Series of Descending Pitches

For this experiment, set up the controls as shown in Figure 4.3.3. Notice that the waveform being fed into the Sample and Hold mixer is a sawtooth wave from VCO-1. This is the waveform that will be sampled to produce the control voltages. Note also that the two-position switch on VCO-1 (upper right of panel, to the right of the blue frequency sliders) is down, in the low-frequency position.

The "fall" of this waveform will be slow enough to permit the LFO Freq setting you are using enough time to sample several times on the negative-going ramp of the sawtooth. This series of falling voltages is then routed to VCO-2 and applied to VCO-2 as control voltages, producing the descending series of pitches you hear.

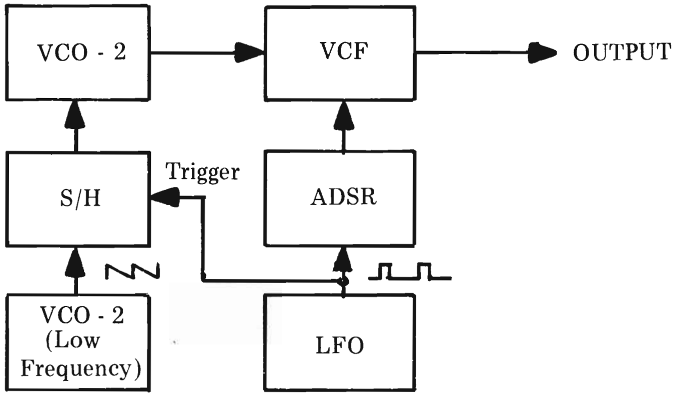


Figure 4.3.4. Block diagram of Figure 4.3.3.

Before going on to Experiment 2, be sure to try various positions of the coarse-tuning slider of VCO-1. You'll find that each position produces a different pattern of pitches. As you change the frequency of the waveform being sampled, you are changing the points at which the Sample and Hold circuit takes its samples (Figure 4.3.5).

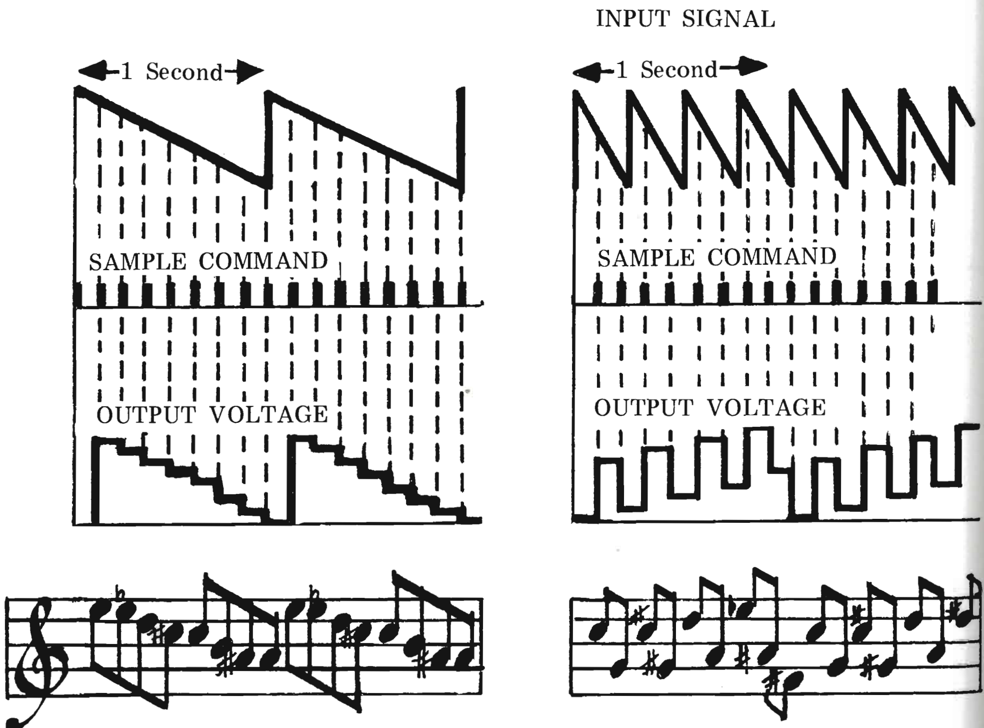


Figure 4.3.5. Creating different patterns with the S/H.

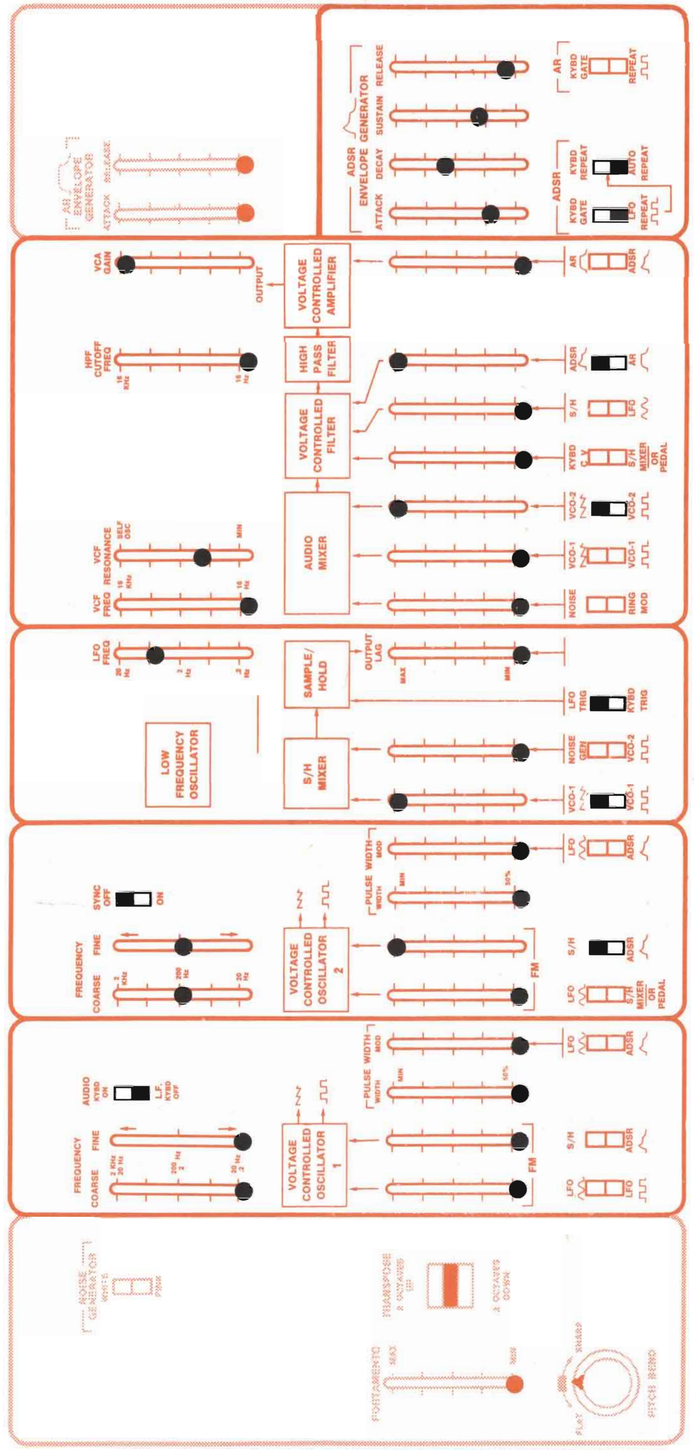


Figure 4.3.3. Basic patch for producing a series of descending pitches.

Experiment 2: Sampling White Noise to Produce a Series of Random Pitches

Having now determined in Experiment 1 that repetitive patterns can be created by sampling a periodic waveform, the patch shown in Figure 4.3.6 will produce a series of random pitch patterns by sampling the white noise supplied by the noise generator. (If you would like a visual review of exactly what is happening in terms of sampling the noise wave, review Figure 4.3.2 presented earlier in this lesson.)

Note that while the series of pitches will still be random, you can limit the over-all range within which the pitches fall by moving the white noise input attenuator on the S/H mixer down — try it about half-way. Experiment with different positions of this attenuator to control the range of the sample. (You will also find that a similar limiting effect can be produced by simply attenuating the yellow control-voltage input to the oscillator.)

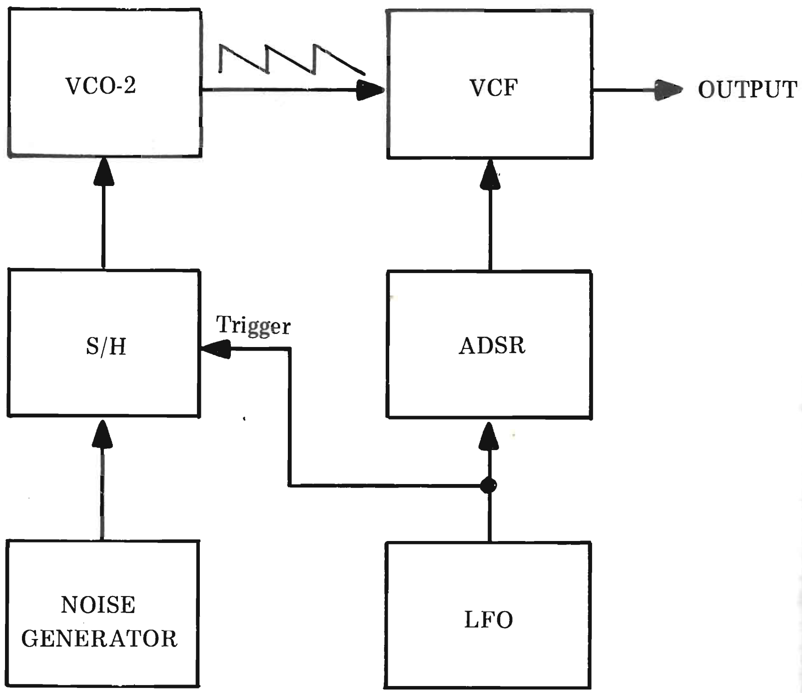


Figure 4.3.5. Block diagram of Figure 4.3.6.

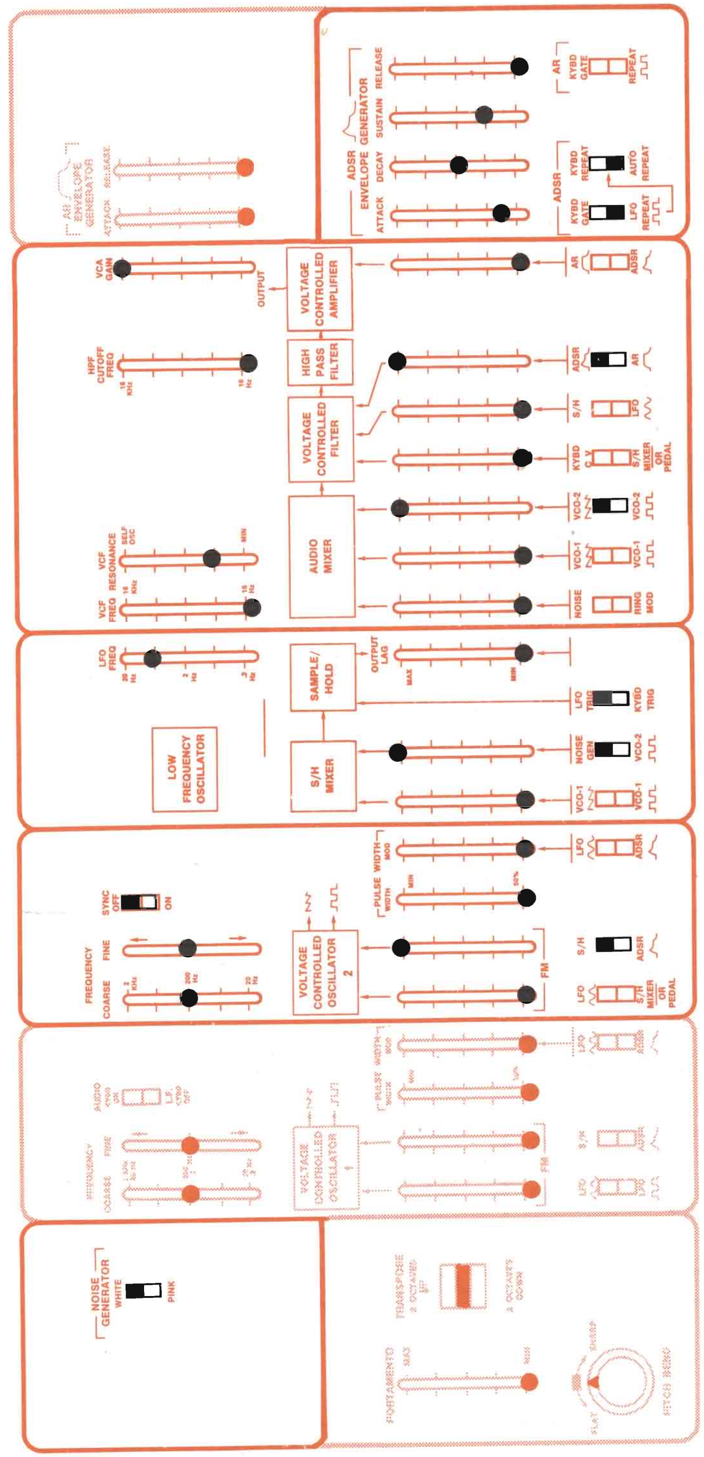


Figure 4.3.6. Basic patch for producing random pitches.

Experiment 3: Sampling a Mixed Wave to Control the Pitch of the VCF in Oscillation

When the filter is put into oscillation, the Sample and Hold circuit can be used to control the pitch of the audio sine wave produced by the filter, just as you can control the pitch of the audio waveforms produced by the voltage-controlled oscillators. Set up the patch shown in Figure 4.3.7. Notice that this time you are actually sampling a mixed waveform; the input from the VCO-1 is providing a low-frequency square wave, while the input from VCO-2 is providing a square wave that is actually in the audio range.

You should understand that you can sample audio frequency waves just as you would sample any voltage. To prove this, lower the white noise/VCO-2 input attenuator and leave the blue input attenuator from VCO-1 open (up). Move the two-position switch below the blue VCO-1 input attenuator up to the sawtooth position. Listen for a moment as the Sample and Hold circuit samples the low-frequency sawtooth wave. You'll probably hear patterns similar to those you created in Experiment 1. Then move the two-position Audio KYBD switch at the top of VCO-1 up, to the audio frequency range; listen to the resulting pattern of pitches. Again, you'll find that by varying the frequency of the wave being sampled (VCO-1), you will be able to vary the pattern of the pitches you hear. You can also vary this pattern by changing the rate of the sample, controlled by the LFO Freq slider. Experiment with each of these two methods of changing the audio pattern.

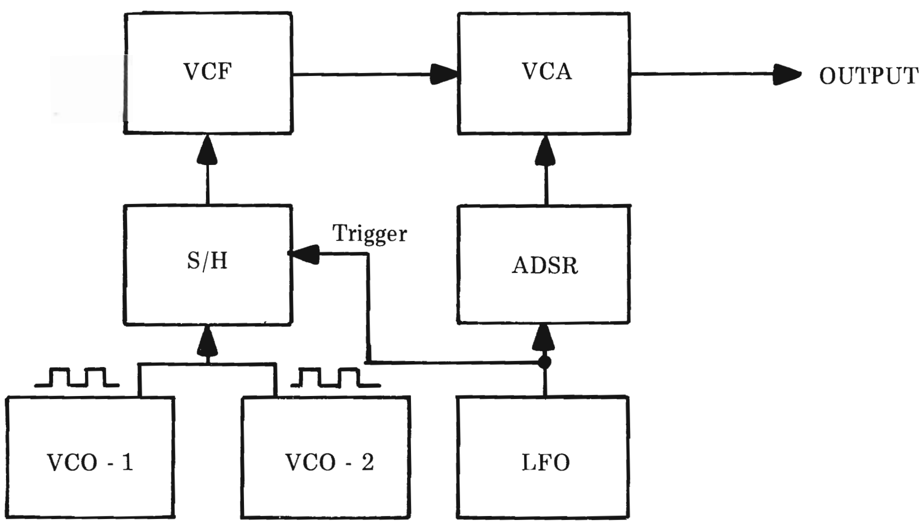


Figure 4.3.8. Block diagram of Figure 4.3.7.

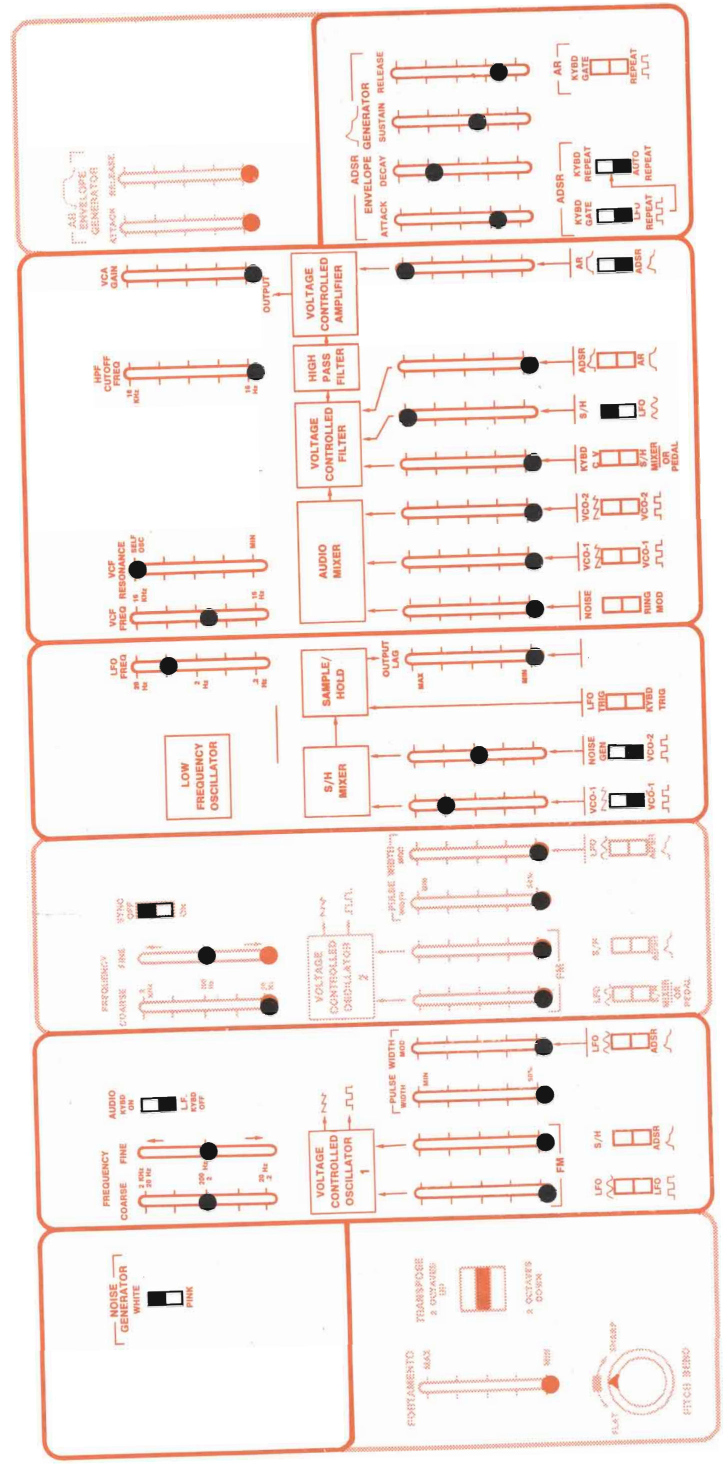
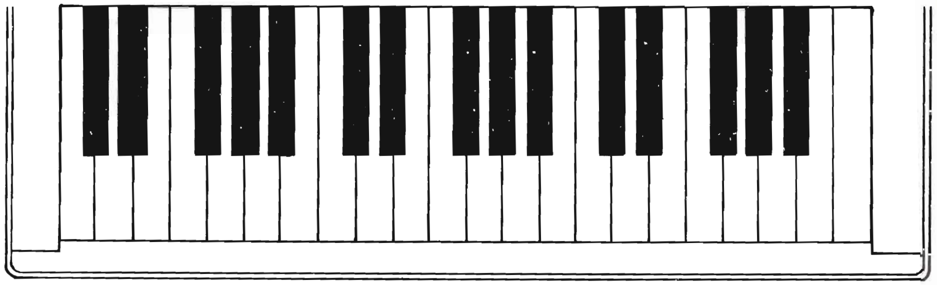


Figure 4.3.7. Basic patch for S/H control of VCF.



Odyssey keyboard.

Lesson 4: The Keyboard Controller and Related Controls

As you have undoubtedly realized by this time, when you are using the keyboard, you are simply using another method of voltage control. The keyboard of the Odyssey provides an interesting combination of voltages, each having its own particular function: (1) a *gate* signal, (2) a *trigger* signal, and (3) a *control voltage.* These voltages are internally routed to the functions they are most often used to control.

The function of the gate voltage is to indicate that at least one key is depressed. The gate voltage is an on/off type of signal used to activate the envelope generators. At the precise instant that any key is depressed, a second signal is generated — a trigger voltage. The function of the trigger voltage is to indicate the exact instant at which any key is depressed. This trigger output appears every time a key is depressed, regardless of how many keys are already being held down. The trigger signal is also used to start the envelope generators, and to trigger the Sample and Hold.

When any key is depressed, the keyboard control voltage assumes some value, depending upon which key is depressed and upon the settings of the Transpose and Pitch Bend controls. The function of the keyboard control voltage is to provide a control over the frequency (pitch) of the oscillators, and over the pitch of the filter when in oscillation.

These signals principally serve to provide the sort of response that we have come to expect from a keyboard through our experience with pianos, organs, and accordions. These voltages are prewired, and precalibrated to provide this convenience. When you play the lowest C on the keyboard, followed by the C one octave above, the proper control voltage is supplied to the oscillators to produce the pitch change that is expected. It is important to understand each of the three voltages — gate, trigger, and control — and how they perform their functions.

Let us now move on to the variable keyboard controls: Portamento, Pitch Bend, the Transpose switch, the Odyssey's two foot-pedal controllers, and the KYBD CV input attenuator to the VCF.

Experiment 1: Portamento

Before beginning this experiment, be sure that the Portamento foot switch is *not* plugged into the Odyssey.

The Portamento control, when the slider is raised, introduces a variable lag, or slide, into the control-voltage output that determines the frequency of the oscillators. Its effect is to prevent the control-voltage output from responding instantly as different keys are depressed. This means that the VCO being controlled by the keyboard will not move sharply from one pitch to another, but instead will "slide" from pitch to pitch. Figure 4.4.1 shows graphically how this happens.

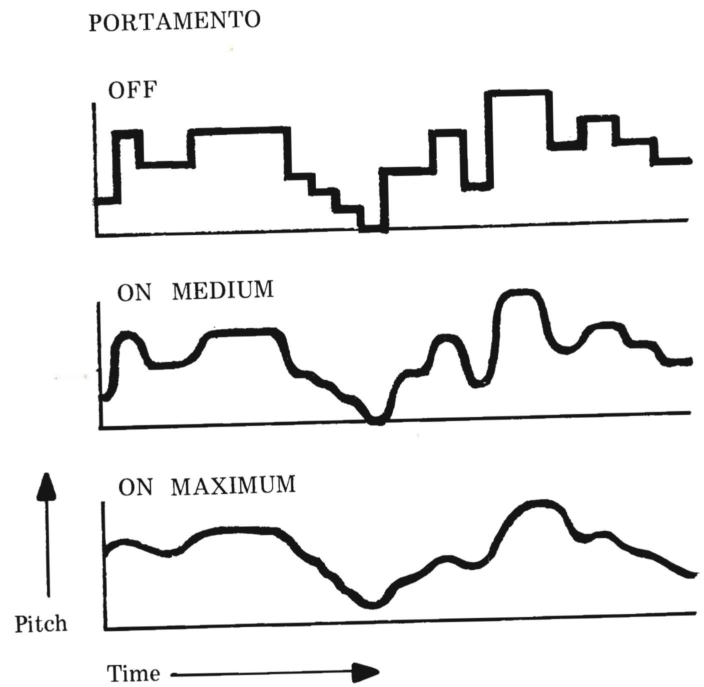


Figure 4.4.1. Diagram of effect of Portamento control.

Set up the patch shown in Figure 4.4.3 and play several notes on the keyboard. Play a familiar tune. Then raise the Portamento slider and play the same series of notes again. Try different degrees of portamento. You'll find that while the effect is certainly interesting, the movement from pitch to pitch is so slow as it reaches its maximum that it is of less use musically that the lesser degrees of portamento.

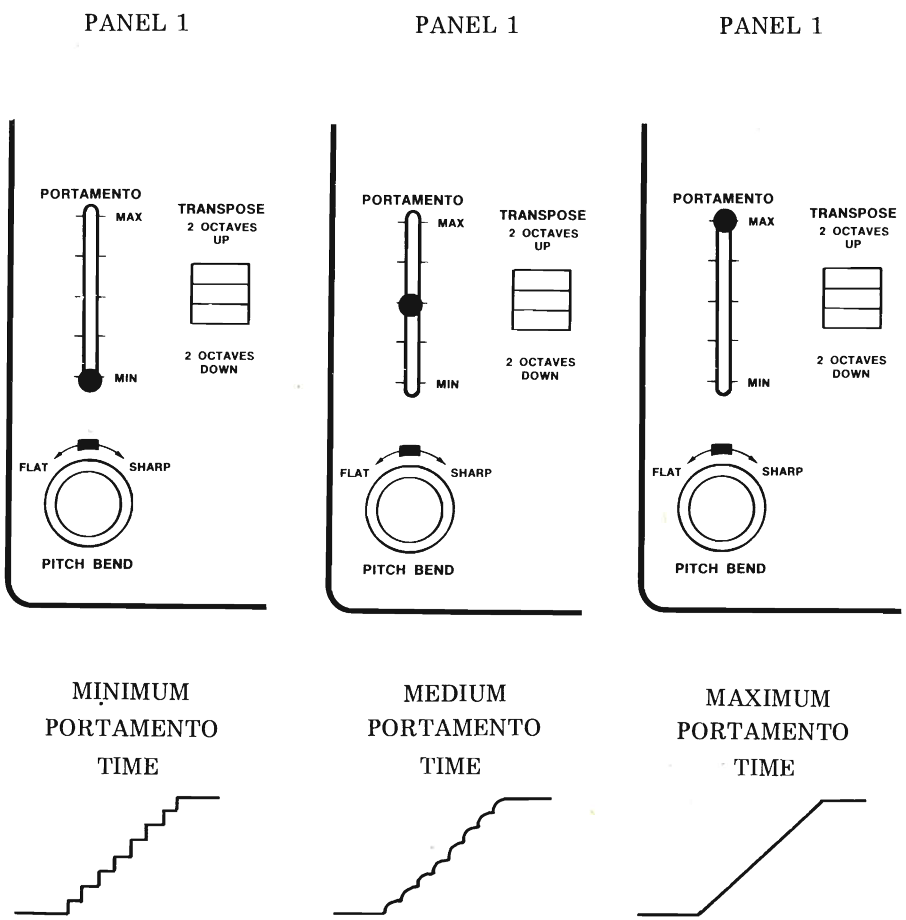


Figure 4.4.2. Portamento control positions.

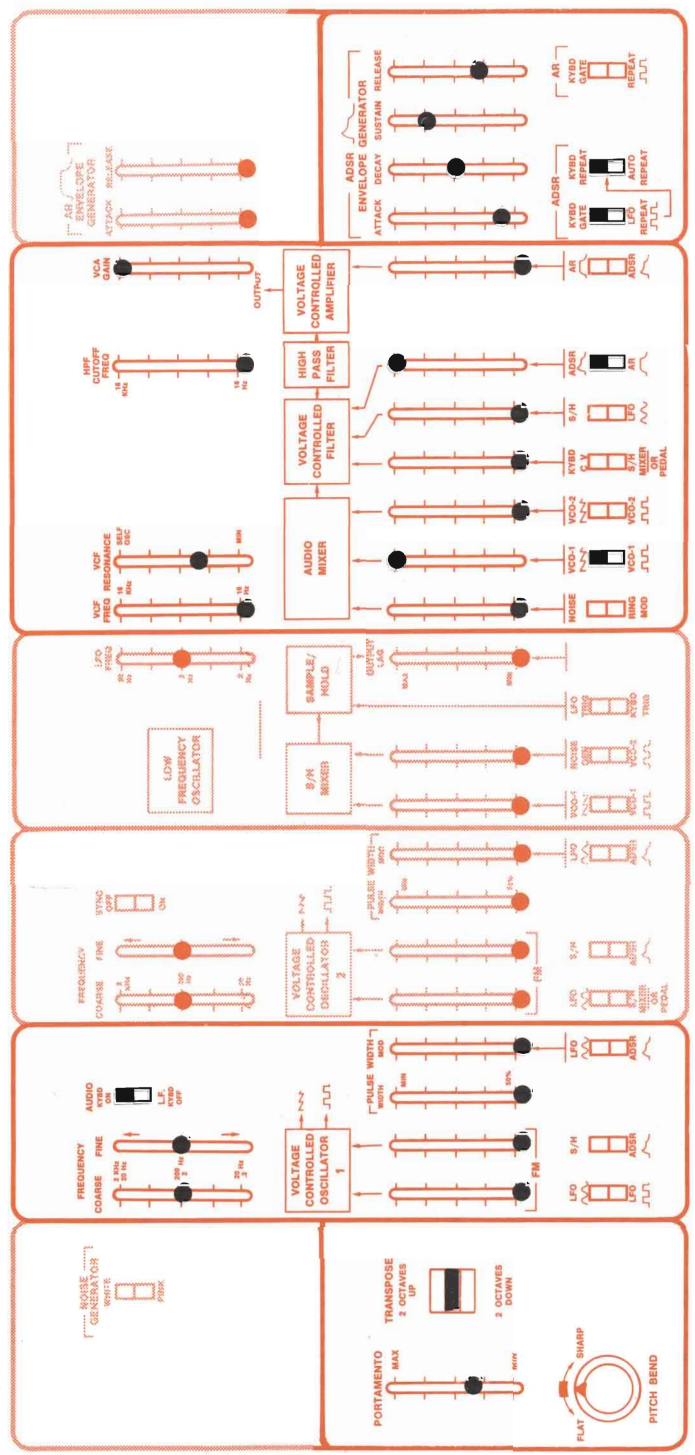


Figure 4.4.3. Basic patch for portamento experiments.

Experiment 2: The Transpose Switch

The Transpose switch permits you to extend the range of the Odyssey keyboard two octaves in either direction, providing a total range of seven octaves. It is simply another voltage controller. Using the patch shown in Figure 4.4.4, try the Transpose switch in each of its three positions. You'll find that it does make a marked difference in the sound of the same patch, depending upon which octave you are in.

If the frequency you are using is a relatively low one to begin with, you may find that by transposing down two octaves the pitch becomes a series of subaudio "clicks;" if the frequency with which you began was a relatively high one, by transposing up two octaves you may go above the range of human hearing. For this reason, most of the patches in this guide are based upon the basic frequency settings of the oscillators as shown in Figure 4.4.4. This will provide the most flexibility in your own patches as well.

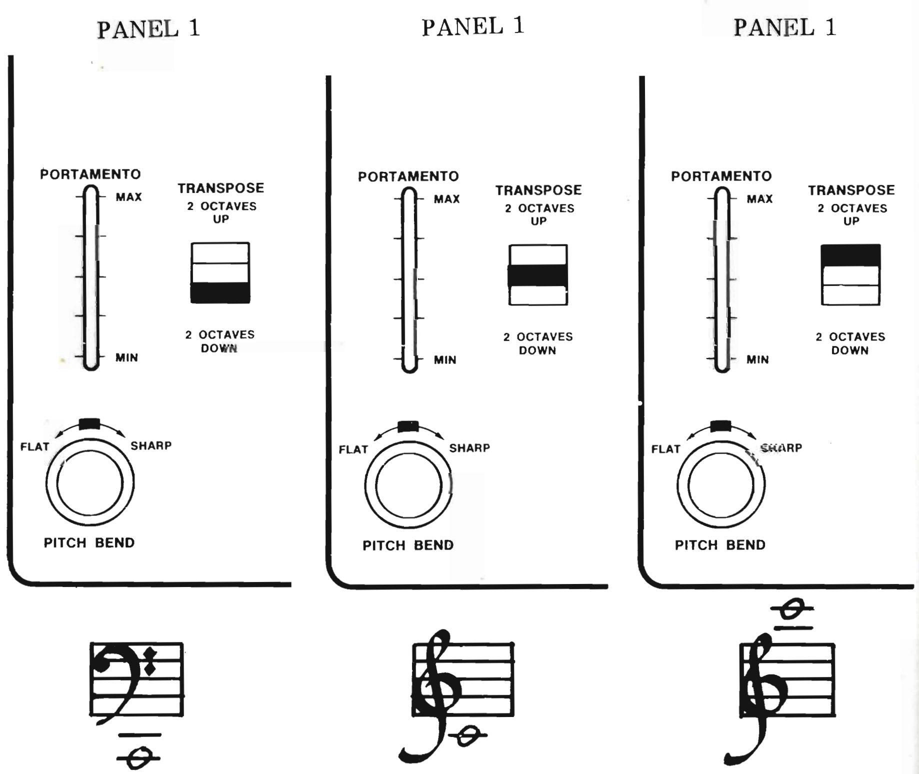


Figure 4.4.5. Transpose switch musical effect.

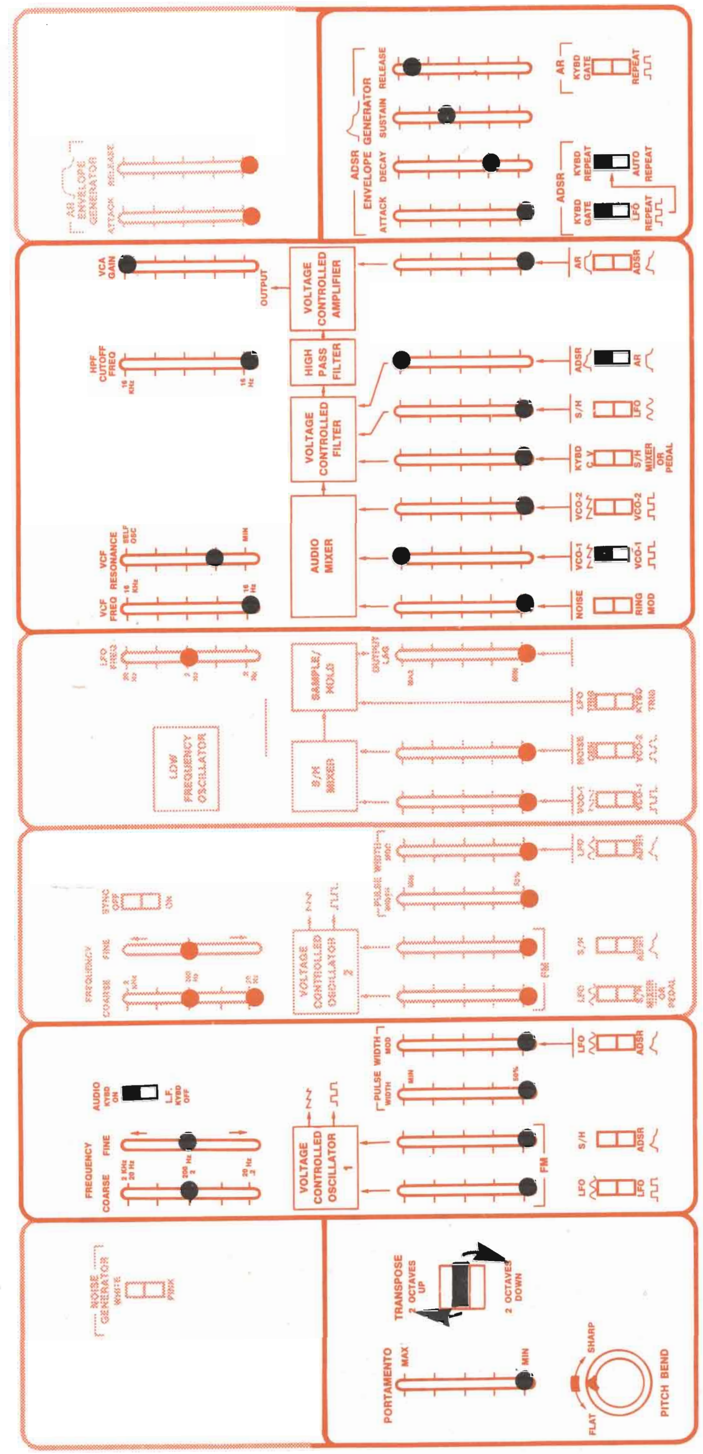


Figure 4.4.4. Basic patch for transposition experiments.

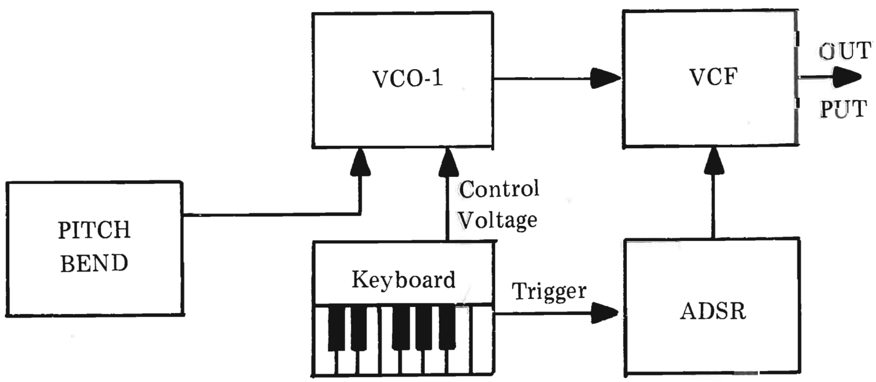
Experiment 3: Pitch Bend

Below the Portamento slider, you will find a knob marked "Pitch Bend." You'll find this control useful in at least two ways: first, it will permit you to tune easily to other instruments after you have tuned the two oscillators with each other. You'll find this a convenience that will save time, eliminating the need to bring one oscillator into tune with whatever instrument(s) you are going to play with in ensemble, and then tuning the second oscillator with the first.

The second primary use for this control is as the name would imply: you can use it to "bend" pitches, in order to recreate more realistically the effect guitars, other stringed instruments, and even some wind instruments can create. Naturally, you can use it to create imaginative, unique effects of your own as well — effects that need not be imitations of any existing instrumental sound.

When recreating the effect of the pitch bend of more traditional instruments, however, it would be wise to limit the pitch deviation to about one half-step. This is the most common effect employed, even by rock guitarists. Set up the patch shown in Figure 4.4.6, with the Pitch Bend knob set at twelve o'clock, as shown. You will find that it takes a little practice to make the pitch deviation smoothly and return to the original pitch. Some performers prefer to set the Pitch Bend knob all the way counterclockwise so that, after bending the pitch upward, by simply rotating the knob fully counterclockwise again the pitch returns to the original oscillator frequency. The one disadvantage to this system, of course, is that it will not permit you to tune the Odyssey to other instruments, unless you tune only in an upward direction within the limits of the Pitch Bend knob's control voltage.

Figure 4.4.7. Block diagram of Figure 4.4.6.



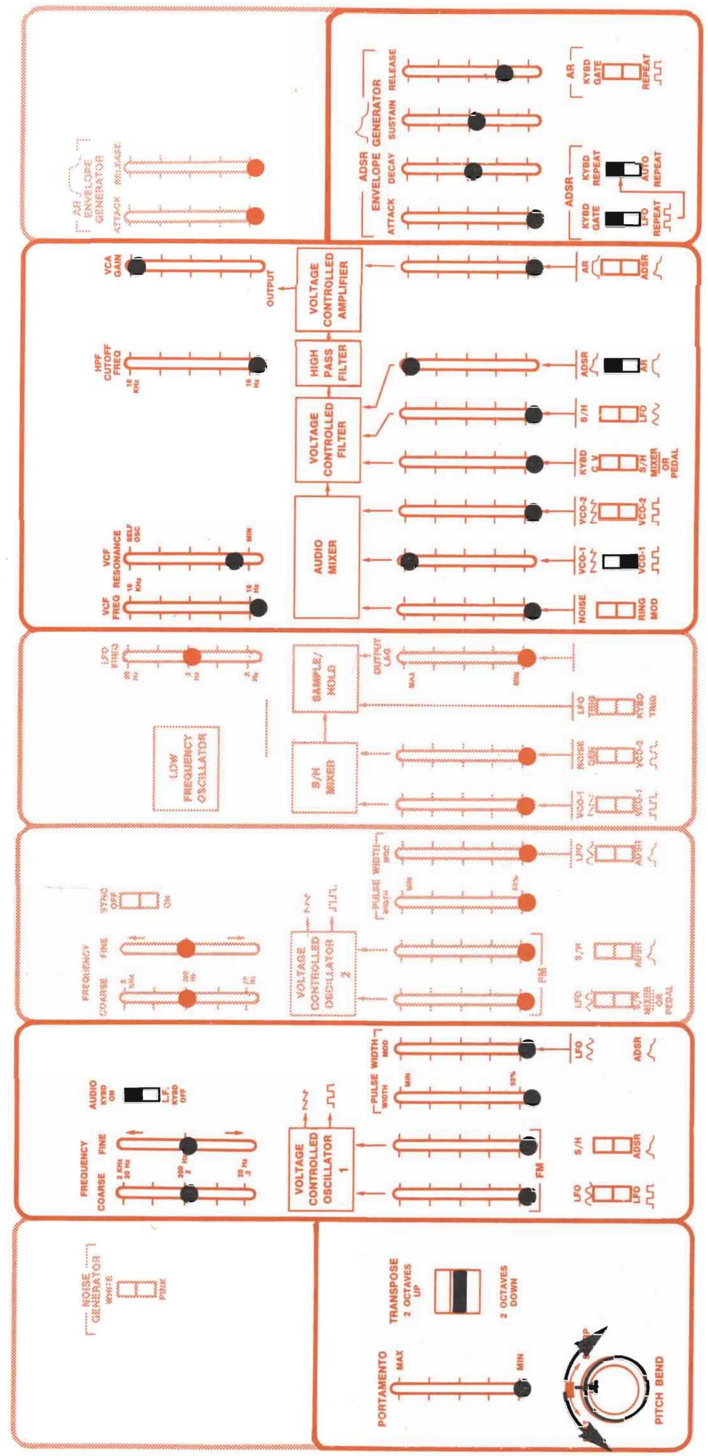


Figure 4.4.6. Basic patch for pitch-bend experiments.

Experiment 4: Using the Keyboard Control Voltage to Create Keyboard Intervals of Less Than One Half-Step

You will find that larger, studio-model synthesizers often have a provision by which you can alter the keyboard control voltage so the total interval of, say, a four-octave keyboard might be less than one octave in terms of actual pitch. The musical result would be, of course, an "octave" with as many microtones as there were keys on the keyboard (in the case of a four-octave keyboard, forty-nine). You can approximate this effect on the Odyssey by setting up the patch shown in Figure 4.4.8. (This effect was previously observed in our discussion of the VCF.)

Having set the controls as shown, raise the black KYBD CV input attenuator all the way. When you play a few notes on the keyboard, you will discover that the filter (in oscillation, to produce the pitch you are hearing) is responding normally in terms of the musical response you expect to hear from a keyboard. Now lower the KYBD CV slider a little and again play a few keys on the keyboard. Try a tune you know well. You will find that the keyboard is now responding with intervals of less distance than you would expect. Lower the slider again, and repeat this exercise. You will soon find that you can create all sorts of microtonal patterns, depending upon the degree to which you attenuate the control voltage of the keyboard as it is applied to the filter.

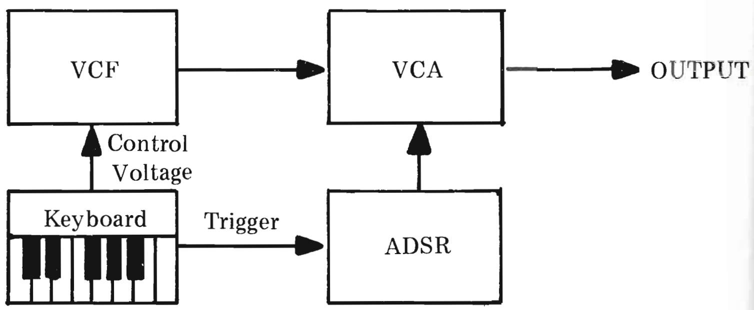


Figure 4.4.9. Block diagram of Figure 4.4.8.

As an interesting conclusion to this experiment, see if you can tune the keyboard to respond with quarter-tone intervals where there would normally be half-step intervals. This will mean that by playing the lowest note on the keyboard and following it with the same note two octaves higher, you should hear an interval of only one octave. You may want to return the black slider to the highest position in order to use the keyboard to fix the octave interval in your mind before tuning with the slider to create the quarter-tone intervals.

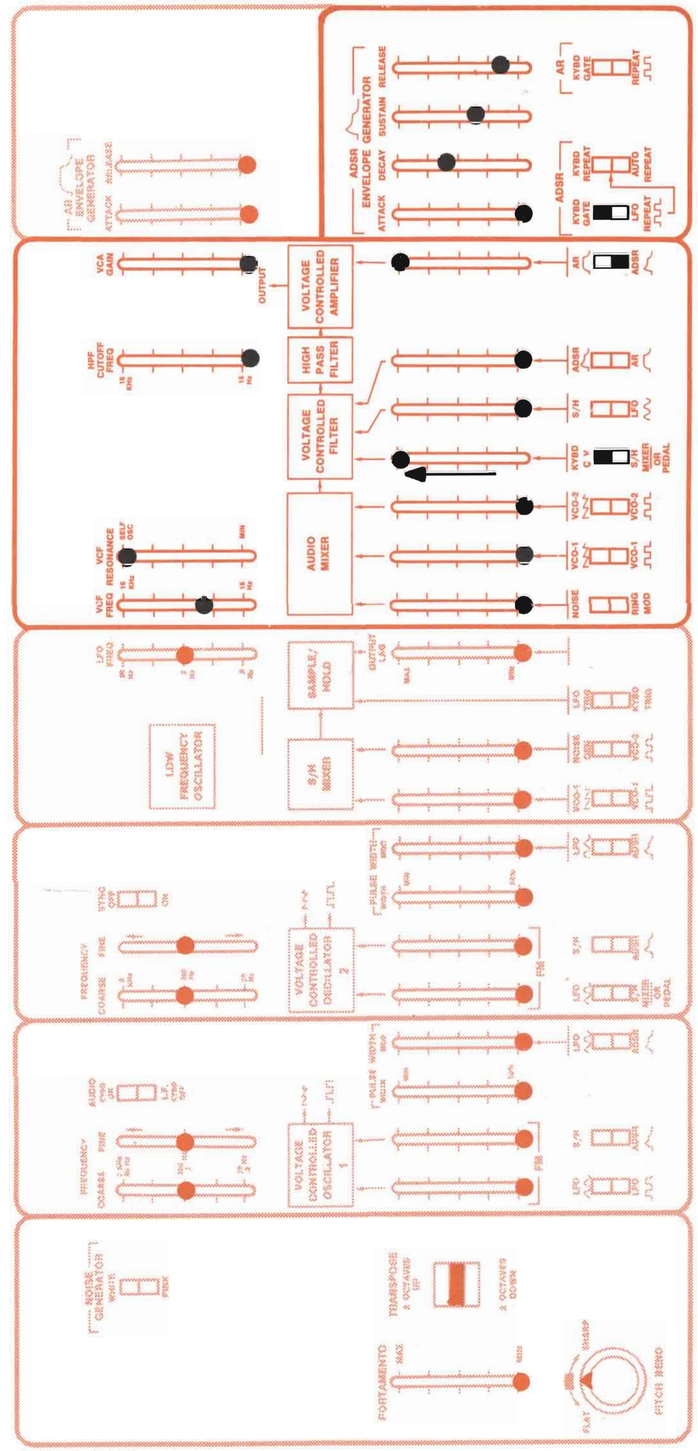
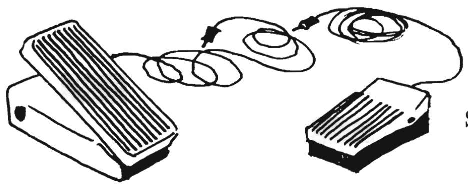


Figure 4.4.8. Basic patch for keyboard control of VCF.

Experiment 5: The Foot-Pedal Controllers

Two foot pedals are supplied with the Odyssey: the first is a small, square pedal that acts like an on/off switch for the Portamento control slider. When this foot switch is plugged into the appropriate jack on the rear of the Odyssey, the portamento function (described in Lesson 4) is disconnected except when you step on the pedal. This feature allows you to preset a certain amount of portamento on the front panel control and then to introduce this portamento by stepping on the switch.

PEDAL  SWITCH

The larger pedal, which operates somewhat like the accelerator on an automobile, can also be plugged into the back of the Odyssey, permitting you to control with your foot certain functions shown on the control panel. When this pedal is plugged in, it may be used to control the pitch of VCO-2 and/or the opening and closing of the filter (VCF). Quite a number of interesting effects can be created by using the pedal to control VCO-2, including a totally new sound in music performance — phase-synchronized oscillators. You'll also find that by using the pedal to control the VCF, you can very easily create an excellent wah-wah pedal effect. Set up the patch shown in Figure 4.4.10. This patch will provide the basis for experimentation with both the foot switch and the variable pedal.

First, play a series of notes on the keyboard, using the foot switch to introduce portamento only when you want it. If you have a fairly long portamento time set by the control slider, you will find portamento most effective only when you come to a note that can be sustained long enough to permit the pitch to slide up or down to the key you are playing. In musical terms, this would mean a half-note would be required in the melody, possibly even a whole note.

Second, experiment with the effect of the variable pedal controller, using it first to control VCO-2, and then the VCF. All that is necessary to introduce this control voltage to either of these functions is to raise the attenuator above the two-position switch on each function that is labeled "S/H Mixer or Pedal." When the pedal has been plugged into the back of the Odyssey, the S/H Mixer is automatically disconnected from both control inputs and is replaced by the pedal control voltage.

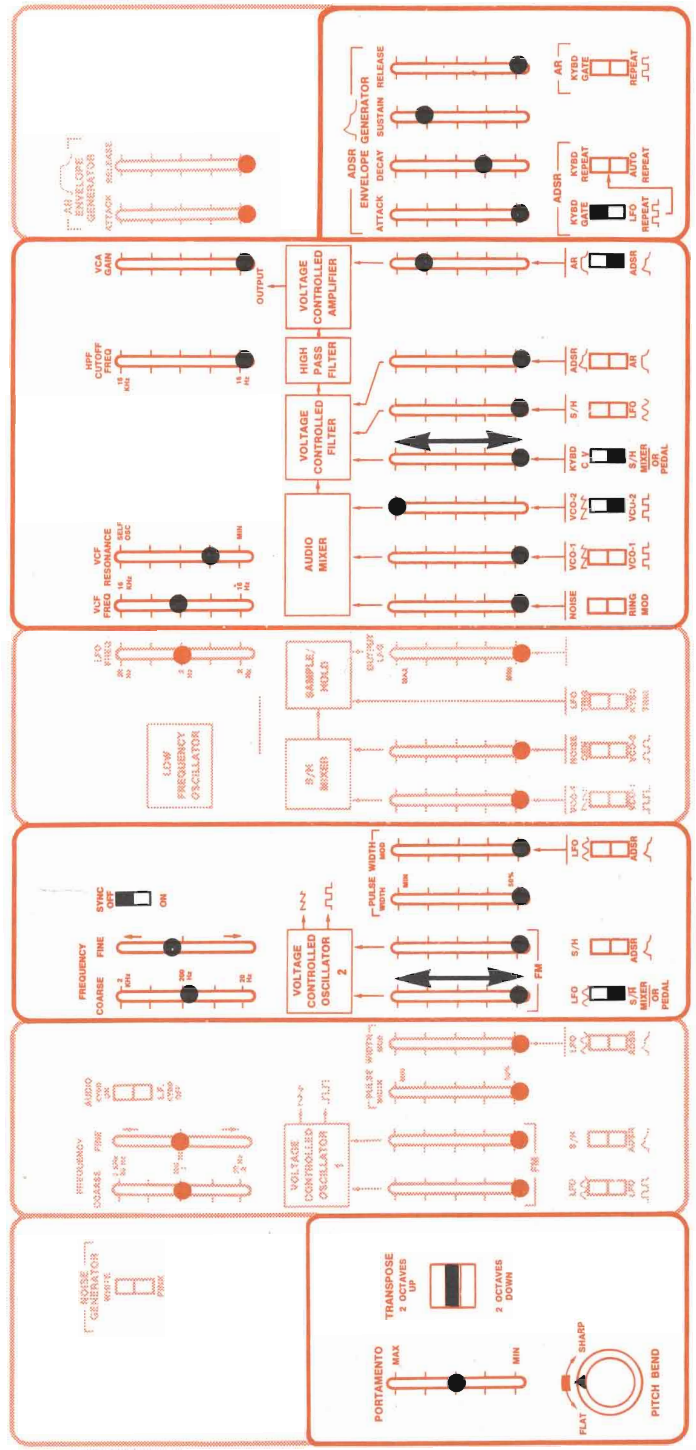


Figure 4.4.10. Basic patch for foot-pedal experiments. (Be sure foot pedal is plugged in.)

Experiment 6: Polyphonic (Two-Voice) Keyboard Capability

Until only very recently, keyboard controllers on all synthesizers could play only one note at a time. These units were designed to synthesize instrumental voices, and logically, since a clarinet or an oboe cannot play more than one note at once, the synthesizers were designed to function in a similar manner. While this is not taking into account the additional technical difficulties in creating a multi-voice keyboard, the fact remains that most synthesizers still have monophonic keyboards.

Keyboard performers, however, are accustomed to polyphonic (chord) response from their instruments. The Odyssey, therefore, has been designed to permit the player to play two independent pitches at the same time. This is accomplished by having the VCO-1 play the lowest key of any two notes being played, while VCO-2 plays the higher key. If you are a keyboard player, you will find this feature extremely useful in both composition and live-performance situations. For a basic two-voice patch, use the control settings shown in Figure 4.4.11. Feed both oscillator signals into the audio mixer and tune the frequencies to exactly the same pitch (unison). This can be accomplished by holding down the lowest key on the keyboard with one hand and tuning with the other. You can also manually open the filter and the VCA for the purposes of tuning. After you have tuned, play a brief melody or phrase you know well, using the two-voice capability. While the patch shown is designed to reproduce the traditional response in terms of pitch, as related to the intervals you are playing on the keyboard, you should not ignore the possibility of different oscillator tunings to create different effects. Try these, for example:

1. Tune VCO-1 and VCO-2 to the same pitch, but one octave apart. This tuning provides a bright-sounding, useful effect.

2. Tune VCO-1 and VCO-2 a major third apart (a major third, in terms of keyboard distance, is five steps — counting black and white keys, and counting the basic pitch as "one" and the key that produces the pitch of the major third as "five.") You'll find that this tuning produces surprising musical results the first time that you play a two-voice line that you know well enough to anticipate what you would hear if it were played using traditional tuning.

3. Tune the two oscillators to other intervals, including a fourth apart, a major second apart, and a minor second apart, as well as any interval your own curiosity might suggest.

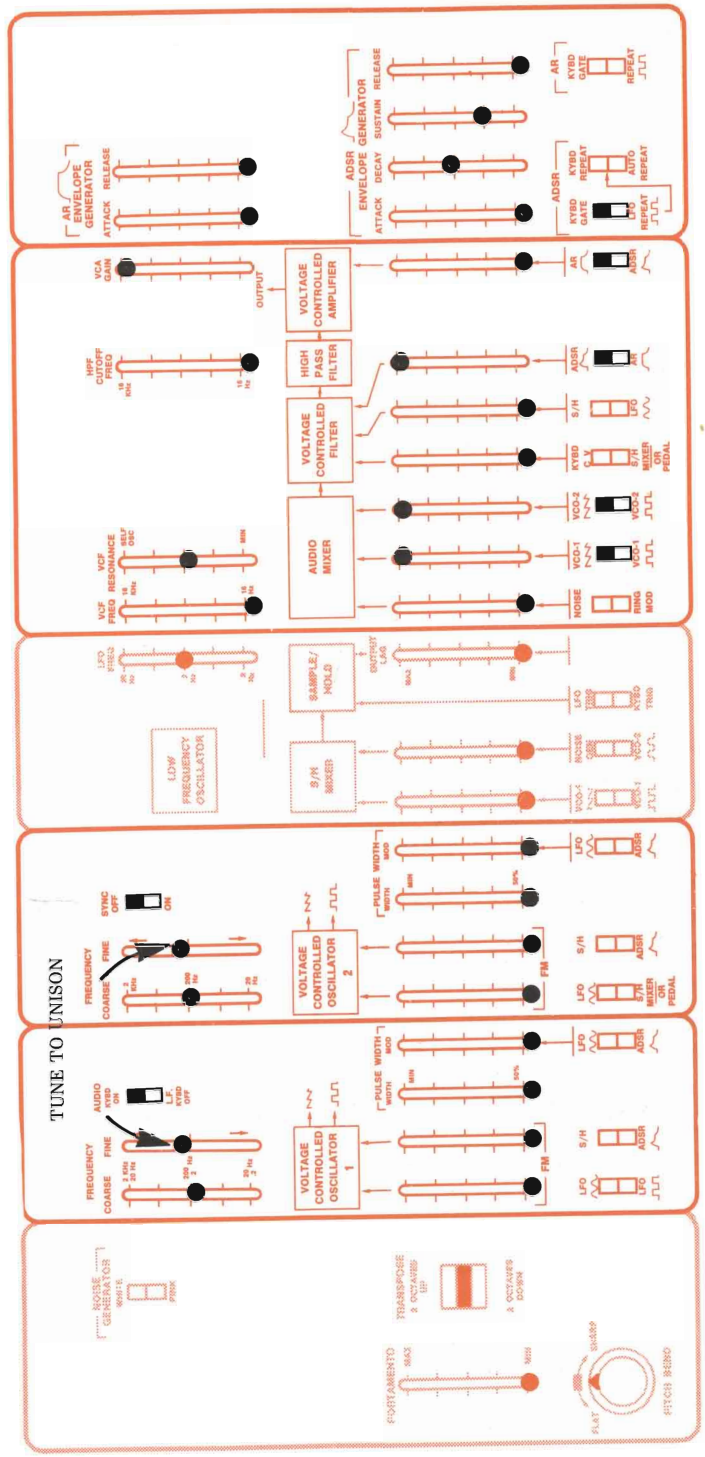


Figure 4.4.11. Basic patch for two-voice experiments.

PART III

At this point you have now completed those parts of this text designed to (1) acquaint you with the fundamental terms and principles of electronic sound synthesis, and (2) to apply them specifically to the operation of the ARP Odyssey. By this time you should be thoroughly familiar with the location and function of all the controls of the Odyssey; you should also have firmly in mind the information presented in Part I. Ideally, that information has been reinforced by the experiments performed in Part II, so that you can now move on to musical applications without being hindered by having to refer constantly to these sections to perform even the simplest experiment. Naturally, a certain amount of reference to that material is inevitable — and is encouraged at points in Part III which involve recalling specific patches or exercises performed earlier. On the whole, however, it is assumed from this point on that you have the basic knowledge required to undertake more sophisticated musical applications.

Part III presents separate sections on five general topics: timbre, melody, harmony, transposition, and setting up an electronic music studio. Within each section, however, you will find related information, ranging from a discussion of ear training to a full-length subsection on tape music techniques. In the event that you are not taking the material sequentially, refer to the Index for page locations of specific subjects.

Section 1: Timbre

The qualities of a particular sound that enable you to distinguish that sound from any other sound can be generally defined as *timbre.* A number of experiments in Part II of this text were designed to use various functions of the Odyssey to change the timbre of the sounds being produced. Moreover, the Odyssey — unlike a trumpet or a violin — is capable of radical modifications of timbre. While a trumpet (despite all you might do short of recording it and then modifying the recorded signal) will still sound like a trumpet, the Odyssey permits almost total timbral flexibility. It therefore is ideally suited to the systematic explanation of the subject of timbre.

Timbre is created and/or affected by a number of interacting factors. Think for a moment about why a flute sounds different (has a different timbre) from a guitar. Your study of Part I of this text will provide a number of basic answers. You'll recall that it was established that different sounds have differently shaped waveforms. It would follow then, that the flute and the guitar obviously produce differently shaped waveforms; taking this one step further, based upon information also contained in Part I, these instruments produce acoustical waveforms containing harmonics differing in number and amplitudes. For example, a flute produces an almost pure sinelike wave; the guitar produces a waveshape more like the dynamic pulse of the Odyssey — containing almost all harmonics, but with an ever-changing harmonic content from the time that the string begins to vibrate until it ceases vibrating.

This changing nature of the harmonic content of an instrument's characteristic waveform brings up an important point: the harmonic content, and therefore the timbre, of waveforms produced by traditional instruments is not constant but, as stated earlier, varies considerably from the moment the sound begins until it dies away. When any such instrumental tone is sounded, the build-up of sound from zero to maximum initial volume is a complex process termed the "attack transient." During this time, pitch, volume, and the spectrum of over-tones go through complex changes to which the human ear is extremely sensitive and perceptive. For instance, a wind instrument usually starts its attack by producing only the fundamental tone, and higher harmonics are added as the sound builds up (Figure 1.1). This is because the column of air in a wind instrument starts to vibrate in a simple harmonic vibratory mode, but breaks up into a more complex set of vibratory modes as time progresses.

We have established, therefore, that attack and decay, as well as the basic waveshapes or harmonics of any sound contribute measur-

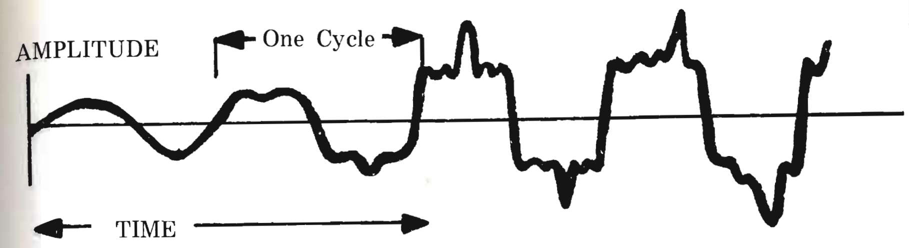


Figure 1.1. Attack transient pattern.

ably to the timbre. While these may be the most important, timbre is also affected — or more properly, our *perception* of timbre is affected — by such things as volume, the pitch (frequency), the presence of a vibrato or tremolo, and even the duration of a sound. Other factors may occur to you; if so, test them against the basic definition that timbre is any quality which makes a sound different or distinctive. Be sure, however, to also take into account overlapping ways of saying the same thing.

For example, it might be argued that the timbre of a large flat-top acoustic guitar is different from that of a solid-body electric, even if the acoustic guitar is also amplified. Therefore, it would seem to follow that the difference in the construction and materials of the two guitar bodies is a factor affecting timbre. To a point, this is correct; however, carry it one step further by reducing the timbre difference to the lowest common denominator of sound synthesis — the waveform. The reason that the bodies of the guitars have an effect upon the timbre is because they each act as a filter/resonator, absorbing certain frequencies while emphasizing others. Thus, the net result is two different waveshapes, just as it would be if you began with the same initial sound source on the Odyssey but processed it two different ways with the VCF and the HPF.

It is important to remember, then, that while considering factors affecting timbre, your key to real understanding is to relate tangible physical factors (such as the guitar bodies) back to the basic component parts of any sound. By doing so, you will not only enjoy a greater understanding of the world of timbres around you, but you will also have a significant head start toward reconstructing or synthesizing those sounds you've heard.

The experiments in this section are designed to further your knowledge of timbre by permitting you to change and carefully control a number of those factors that contribute to the distinctiveness of several sounds. If you have access to a tape recorder and can get together a number of students who play different instruments, you'll find the first experiment provides a dramatic illustration of how timbre is affected by the attack and decay characteristics of a sound. The same experiment, of course, can be performed utilizing the ADSR envelope generator of the Odyssey. Your understanding will benefit, however, if you actually hear what happens to the sound (timbre) of familiar, conventional instruments when robbed of their distinctive attack and decay.

Experiment 1

On a day that has been agreed upon in advance, have several students bring their instruments to class. If possible, try to have at least six to ten different instruments represented. Perform the following sequence with each different instrument.

1. Turn on the recorder and set it to record. A tape speed of 7½ ips (inches per second) is best.
2. When the tape is rolling, signal a student to begin a single tone and hold it until the signal is given to stop. Each tone should be sustained at least five seconds. Wind instruments will have no difficulty in doing this, nor should bowed strings. Other instruments, such as a guitar, a piano, a triangle, or a xylophone, should be struck once and then simply allowed to ring. Have each student play the same note (C, for example) but in the mid-range of his own instrument.
3. When all instruments have been recorded, edit the tape so that the *total* attack and decay of each instrument has been cut out of the tape and the remaining parts — the two or three seconds of continuous tone — spliced back into the reel with brief silences between each example. (See the section on Tape Editing for techniques.)
4. Now play the tape and listen for the differences in the timbre of each instrument. In some instances the change is so radical that the instrument will be difficult to identify; in almost every case, it will be agreed that the attack and decay characteristics contribute measurably to the distinctive timbre of each instrument.

Two enjoyable variations of Experiment 1 can also be performed. The following variations are given for those classes of students wishing to pursue further this aspect of our study of timbre.

Experiment 2

Perform the four steps outlined in Experiment 1. This time, however, have the students play a note as close in pitch as possible to that being played by all other instruments. Try middle C or, if you have

the proper pitch pipe or tuning fork, A-440. After editing, you'll probably find the instrumental timbres even more difficult to identify — particularly if you change the order or sequence of instruments from the order in which they were recorded.

Experiment 3

Save the pieces of tape edited out of the main reel — those pieces having the attack and decay recorded upon them. Splice all of the attack segments together with brief silences between. Are these segments easier to identify then the central portions retained on the primary reel? You may also want to try the decay segments only. Other combinations, such as splicing the attack of one instrument to the decay of another, will occur to you as you experiment. These, however, are beyond the immediate scope of our discussion here as to how the attack and decay characteristics contribute to the distinctive timbre of any instrument.

Relating it all to Synthesis

Throughout this text, most subjects are approached from two directions: that is, starting from a central concept, such as timbre, it is intended that (1) your understanding of traditional musical concepts and/or phenomena will be expanded, and (2) at the same time you will also apply this knowledge as it relates to the Odyssey — thus expanding your ability to recreate sounds you've heard and to synthesize totally new sounds with an ever-growing degree of ease and control.

Relating the attack and decay of instrumental sounds to the Odyssey, we find that in order to achieve the expressive qualities of a traditional wind instrument, it is necessary that an electronic musical instrument be able to "process" waves to build up in a similar way, from a simple harmonic mode to a complex wave. In order to accomplish this, electronic musical instrument designers turned to a device known as a "voltage-controlled low-pass filter." This type of filter has a variable *frequency response*, in that its operation can be varied to allow more or fewer high-frequency waves to pass through relative to the low frequencies. Its operation can be shown by considering how a complex wave with a large number of harmonic overtones is modified, as shown in Figure 1.2.

In the actual operation of the Odyssey, the control of the frequency response is automatically achieved by application of the control voltage generated by the envelope generator(s) to the voltage-controlled filter (VCF), whose frequency response is a function of the control voltage.

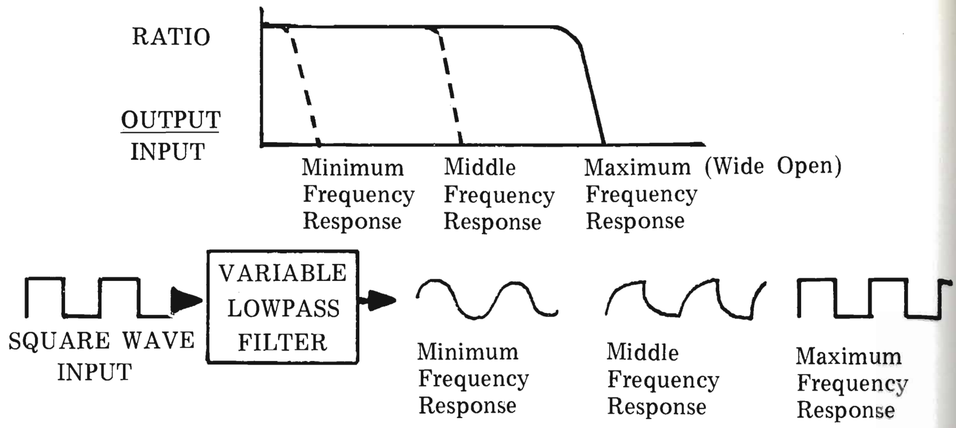


Figure 1.2. Diagrams of filter frequency response.

The waveform resulting from this processing by the VCF has the attack and decay characteristics preprogrammed by the ADSR or AR generators. Therefore, in Figure 1.3, you'll see that the harmonic content of the square wave begins (during the attack time) the gradual build-up spoken of earlier. Similarly, during the release time, the harmonics die away in accordance with the preprogrammed decay or release time. Visually, then, you now understand those portions of the instrumental sounds that were edited out of the tape in Experiment 1.

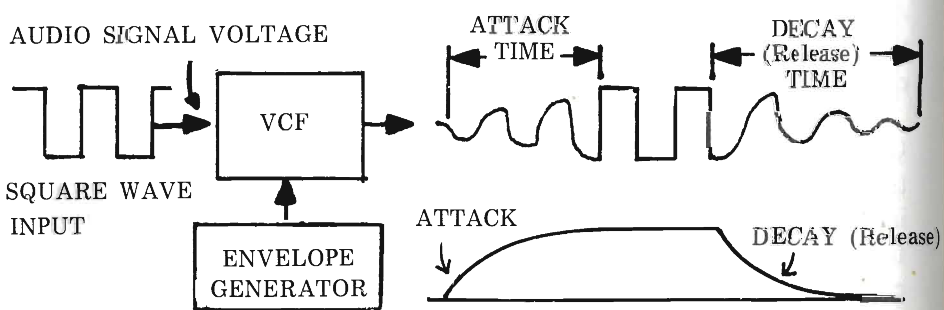


Figure 1.3. Creating an attack and release with the VCF and Envelope Generator.

Experiment 4

This experiment will provide an opportunity to "edit" with the ADSR generator of the Odyssey. In this instance, however, you'll have the ability to control totally each of the four parameters — attack, initial decay, sustain, and release — that contribute to the timbre of the sound you are creating.

You may begin this experiment with the clarinet patch shown in Figure 1.4. You should, however, try a number of the instrumental

patches that appear at the end of this section. In each instance, you'll find that as you modify the settings of the ADSR, the realism of the patch is very much affected.

If you want to simulate the "chopping" of all attack and decay characteristics, as was done by editing the tape in Experiment 1, set the ADSR as shown in Figure 1.5.

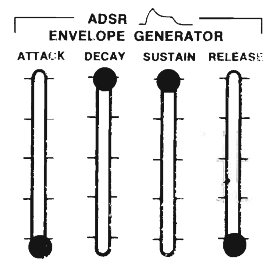


Figure 1.5. Control settings for attack and decay experiments.

Experiment 5

To demonstrate easily the effect that the straight harmonic content of a wave has upon timbre, set up the patch shown in Figure 1.6. Now, without the interacting factors of attack and decay, you can demonstrate that the basic harmonic content itself is a determining factor in the timbre of any sound. To do so, gradually raise the pulse-width slider on VCO-1 from 50% to 10%. You'll hear both the number and amplitude of the harmonics increase as you move toward 10%, producing a brighter, more nasal sound. For a visual reference as to what is happening, refer to the following diagram, (Figure 1.7) reproduced from Part I.

Note that when the pulse-width slider is in the 50% position, the square wave produced contains only the odd-numbered harmonics; as the wave shape changes to a narrow pulse, the even-numbered harmonics are added and the amplitude of the higher harmonics increases markedly.

Before going on to Experiment 6, you should also listen to the sawtooth wave from VCO-1 and the sine wave produced by the VCF in self-oscillation. If you need to review how the patch for each of these waveforms is created, refer to the appropriate sections of Part II.

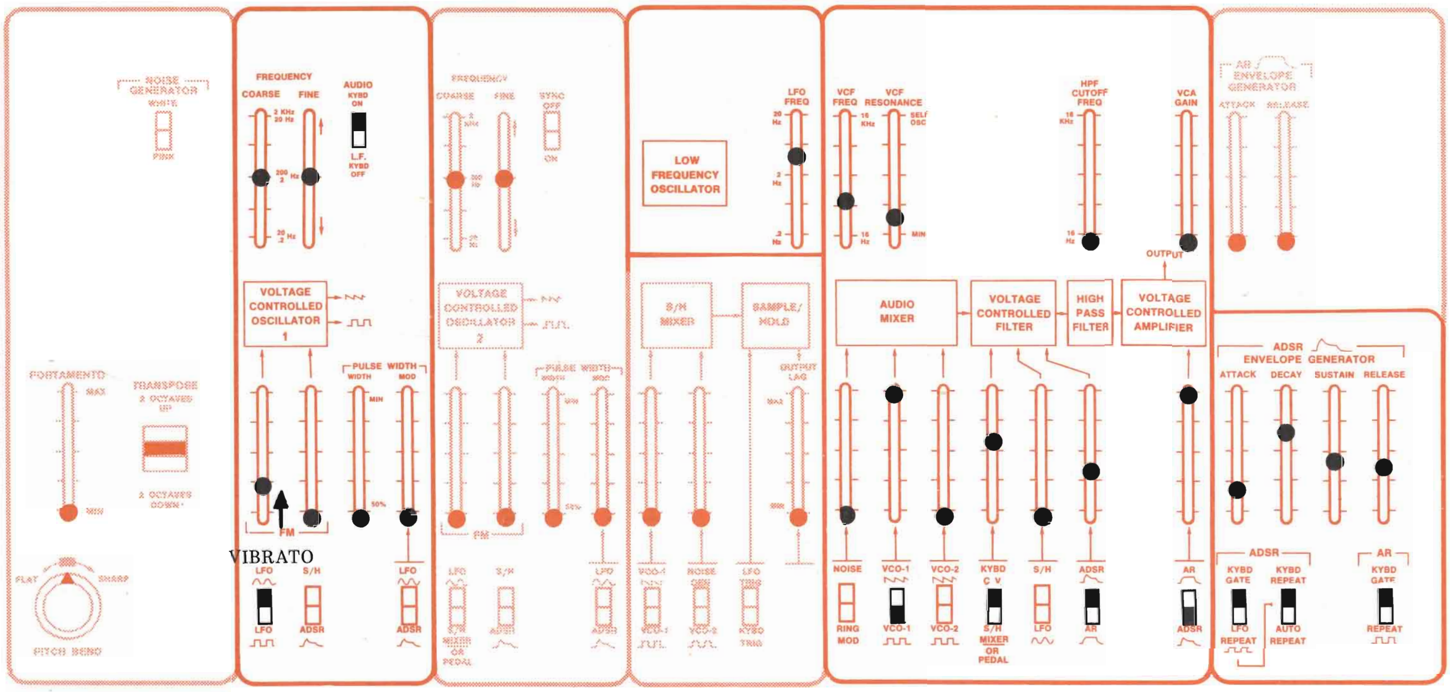


Figure 1.4. Basic clarinet patch.

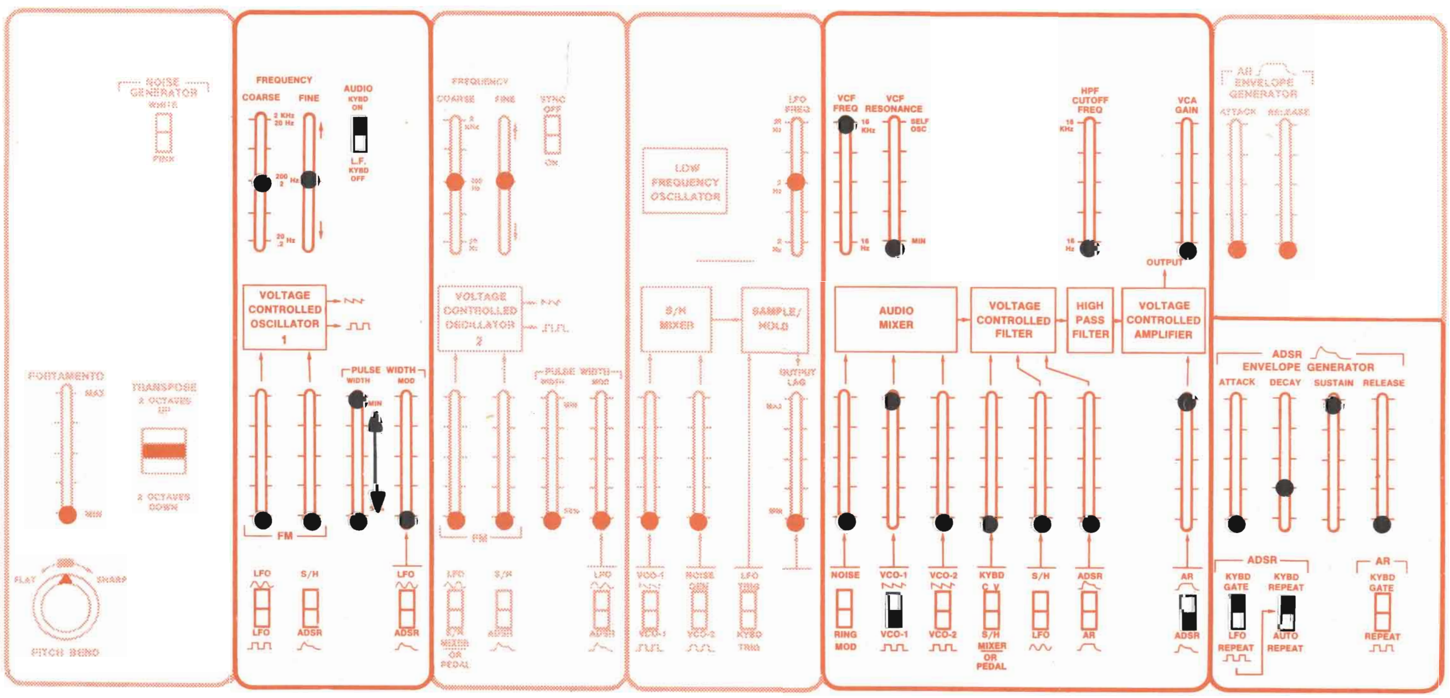


Figure 1.6. Basic patch for timbre experiments.

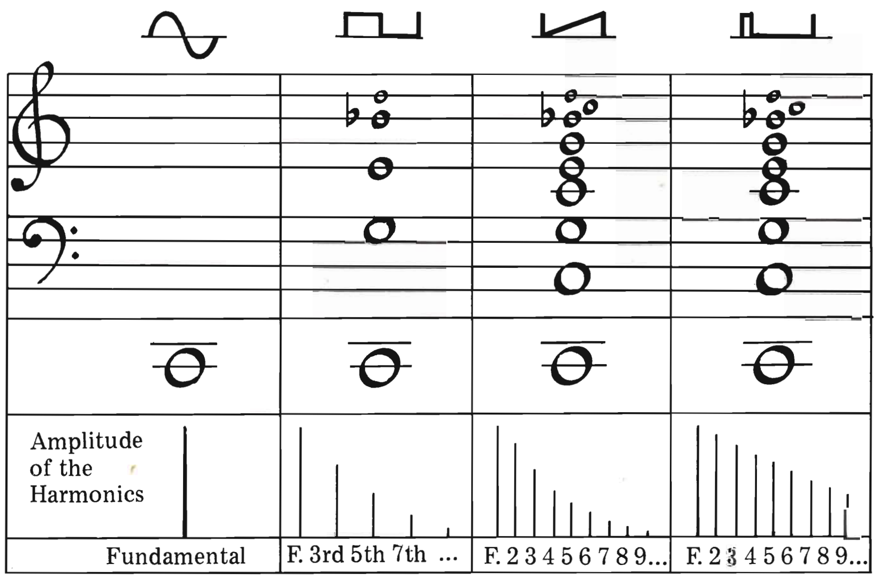


Figure 1.7. Harmonic series for different waveforms.

Experiment 6

Set up the electric bass patch shown in Figure 3.1.13. You are now ADSR-modulating the pulse width (harmonic content) of VCO-1. To demonstrate the difference in timbre that this is creating, simply lower the input attenuator that is permitting the ADSR to affect pulse width. You should also try various settings of the input attenuator, in order to achieve maximum realism to your ear.

Experiment 7

Earlier in this section, we stated that the pitch of a sound affects the way we perceive timbre. Take the same electric bass patch you used in Experiment 6 and transpose the effect up two octaves by using the Transpose switch. The result, of course, is totally unlike an electric bass; try the same procedure with the tuba patch. If you could actually see on an oscilloscope what is happening to the waveform, you would see that the basic waveshape has not changed; the frequency has simply increased. Technically, then, when there is no change in the basic shape of the wave, there can be no change in

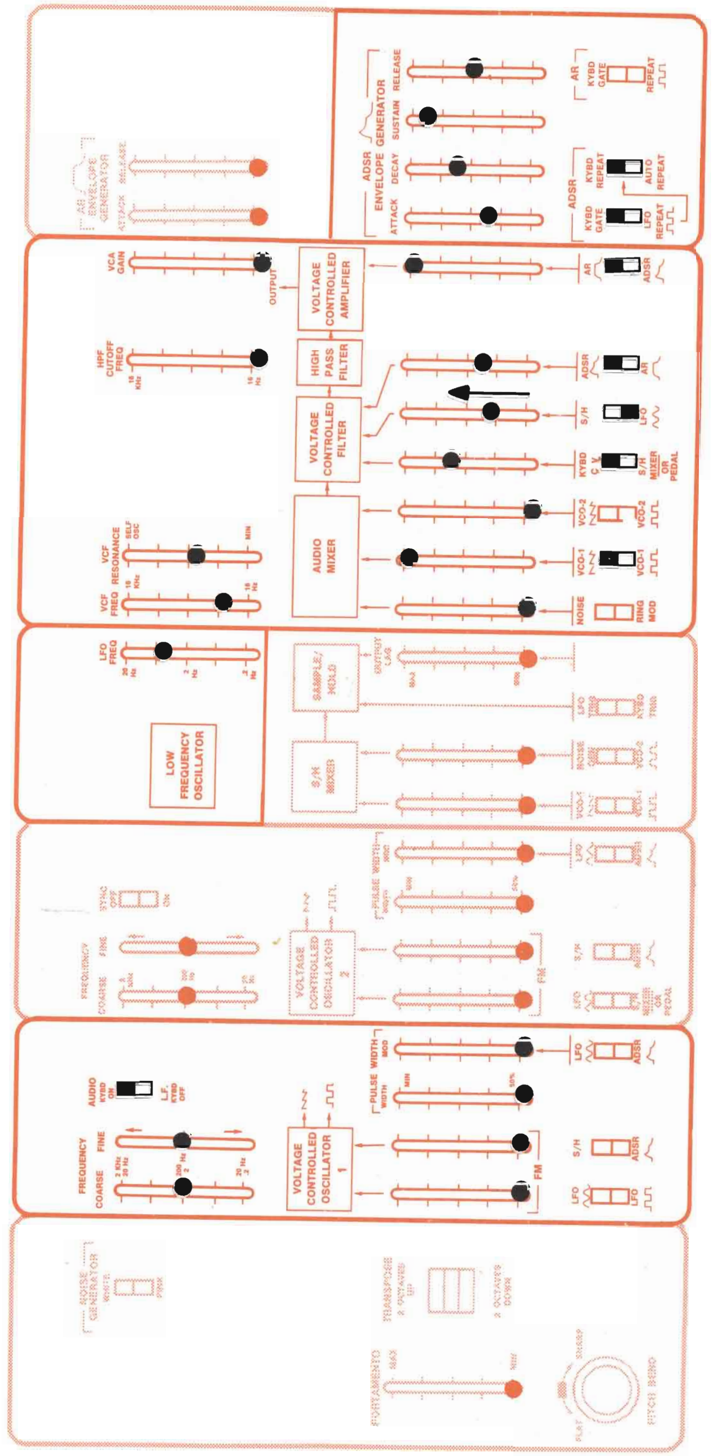


Figure 1.8. Basic flute patch for perception of timbre.

the sound or timbre. What does happen is a twofold effect: first, by raising the fundamental two octaves, the harmonics are correspondingly higher, though in the same relationship to the fundamental. Nevertheless, by radically raising the pitch, it is possible that the upper harmonics will be above audible range, thus changing your *perception* of the timbre. Conversely, if you lower any effect so that the fundamental is below the lowest frequency humans perceive as pitch, you will again drastically change the perceived timbre, without changing the waveshape. This is one reason why two instruments producing the same basic waveshape in different frequency ranges may sound quite different.

The foregoing hints at the second point: we associate certain pitch (frequency) ranges with certain instruments through our own musical experience. Therefore, an electric bass transposed up beyond its characteristic range is simply inconsistent with what we expect to hear and the instrument's traditional musical purpose of providing a rhythmic and tonal foundation. More on this subject as it relates to orchestration appears later in Part III under "Melody."

Experiment 8

The final experiment in this section demonstrates how a vibrato or tremolo affects the timbre of instrumental effects. Set up the flute patch shown in Figure 1.8.

First, raise the input attenuator on VCO-1, permitting the LF sine wave to create a vibrato. A vibrato, of course, is a cyclic pitch change — the control voltage of the LFO actually raising and lowering the frequency of VCO-1. As you raise the slider further, permitting the vibrato to grow wider, the timbre is an effect totally uncharacteristic of anything traditional, except perhaps a musical saw or certain types of police sirens. Is a timbre change involved? In the strictest sense of basic waveform shape, perhaps not; nevertheless, as in the case of the "soprano" electric bass, our *perception* of the timbre is certainly different.

Now lower the attenuator on VCO-1 and raise the input attenuator for the LFO on the VCF. Again, play a few notes using the flute patch. Include a few sustained tones in your playing. Does the tremolo effect create a change in timbre? Definitely. What is actually occurring is amplitude modulation of the upper harmonics, controlled by opening and closing the filter with the low-frequency sine wave. Just as in the case of the ADSR control of the filter we examined earlier, here an actual change in the waveshape does take place as harmonics are weakened or totally removed.

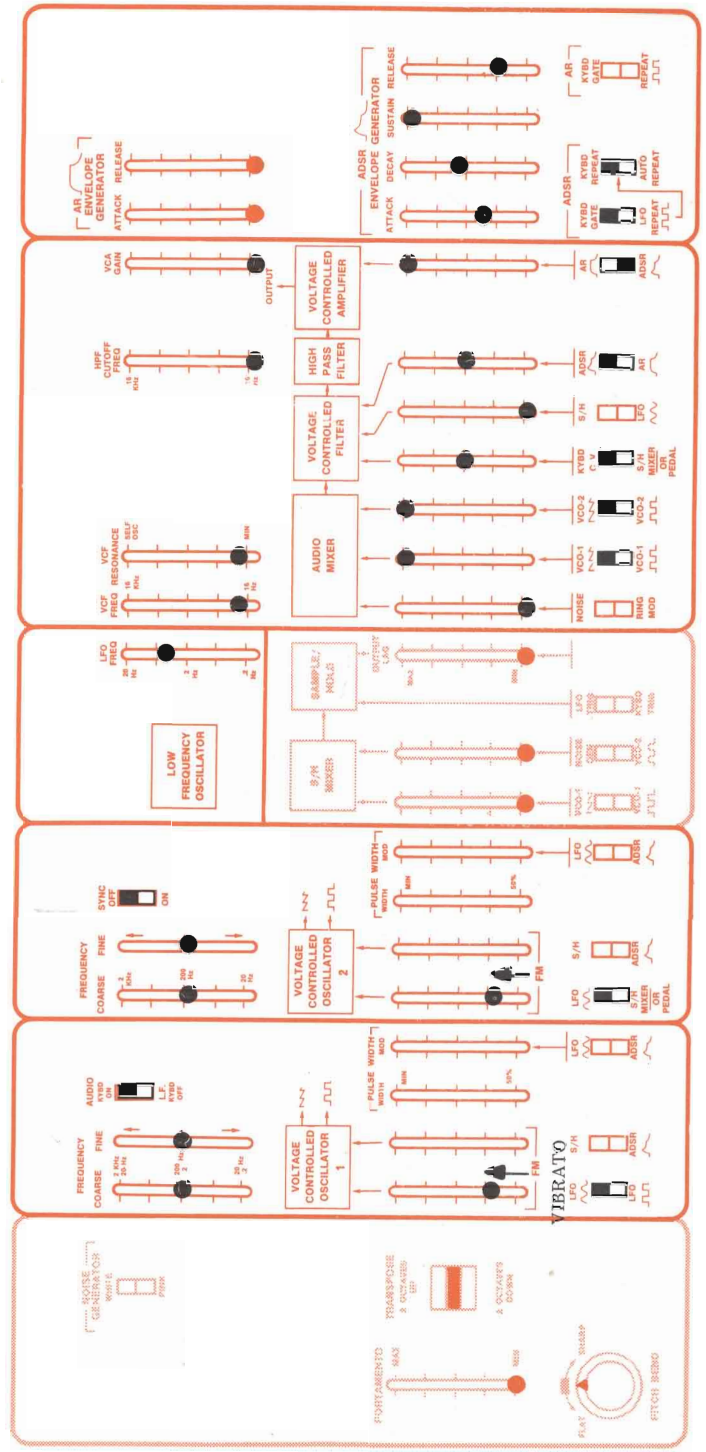


Figure 1.9. Trumpet.

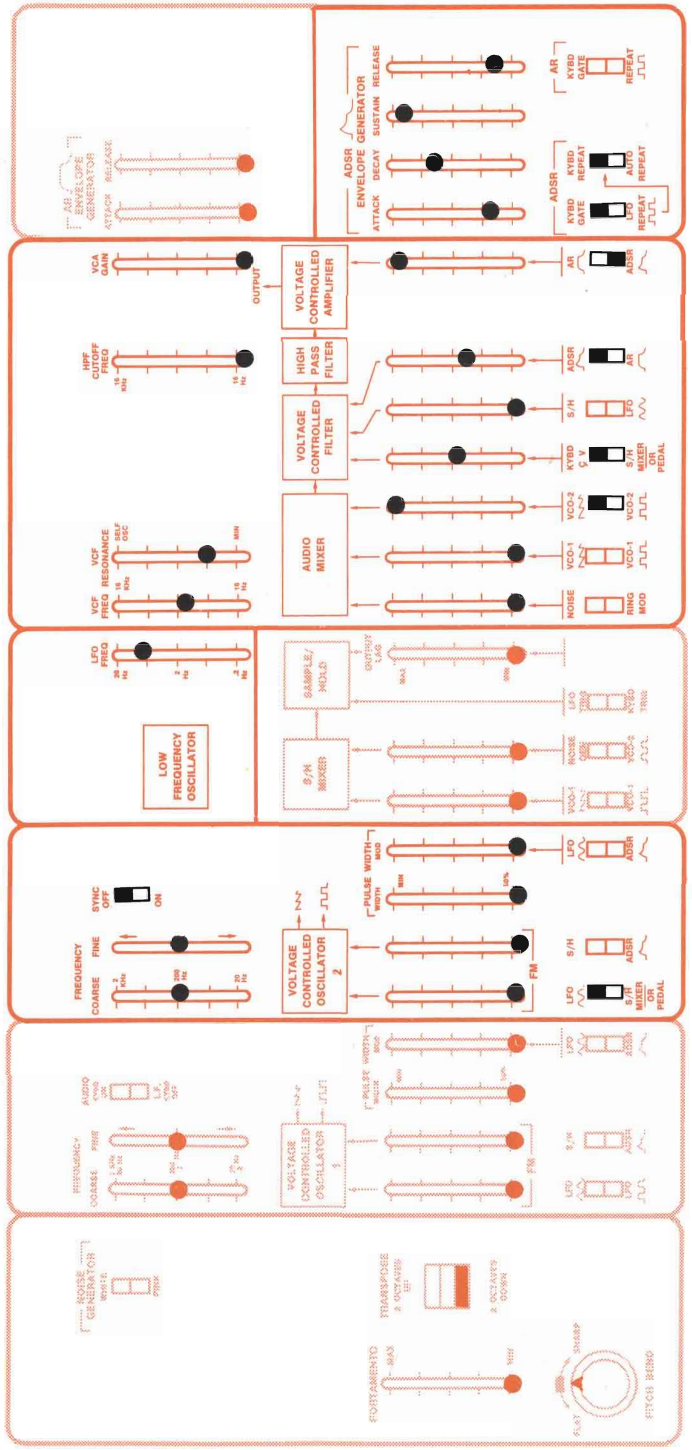


Figure 1.10. Tuba.

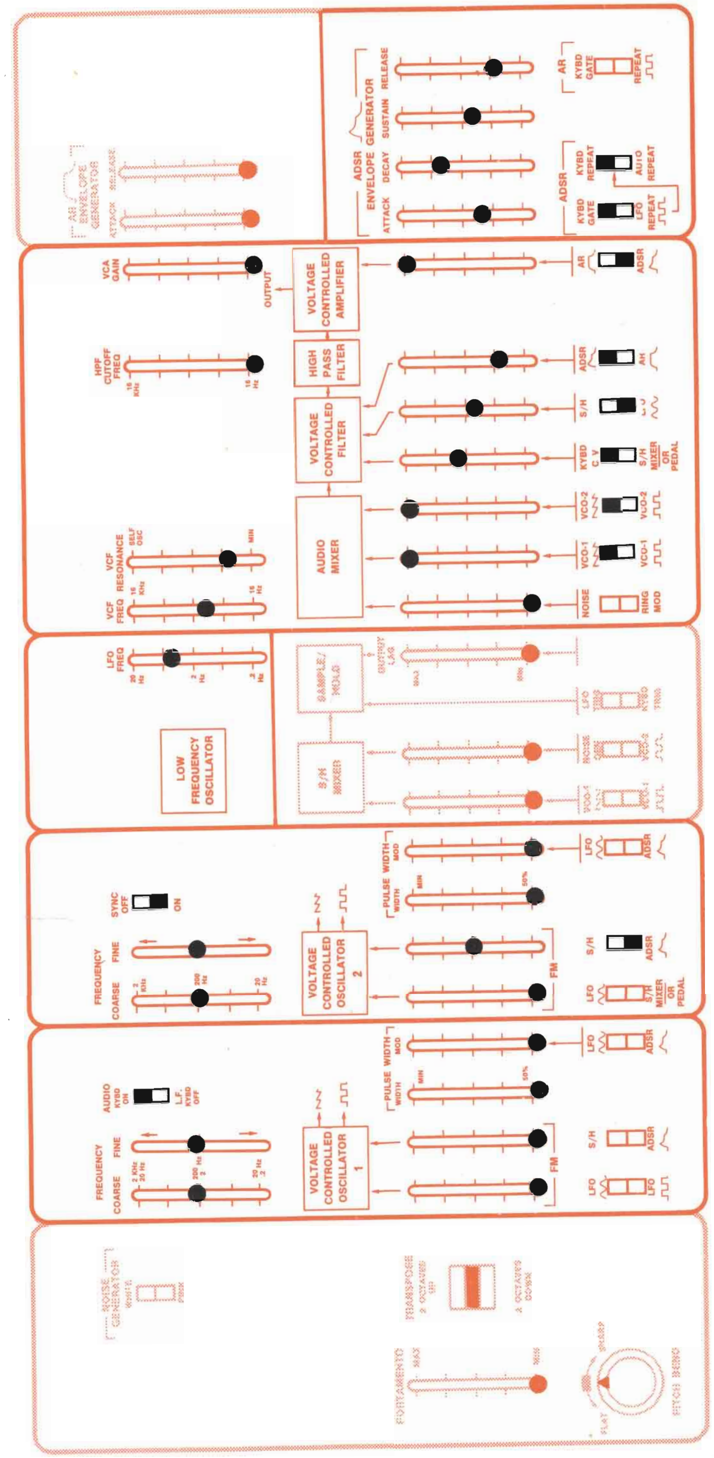


Figure 1.11. Flute.

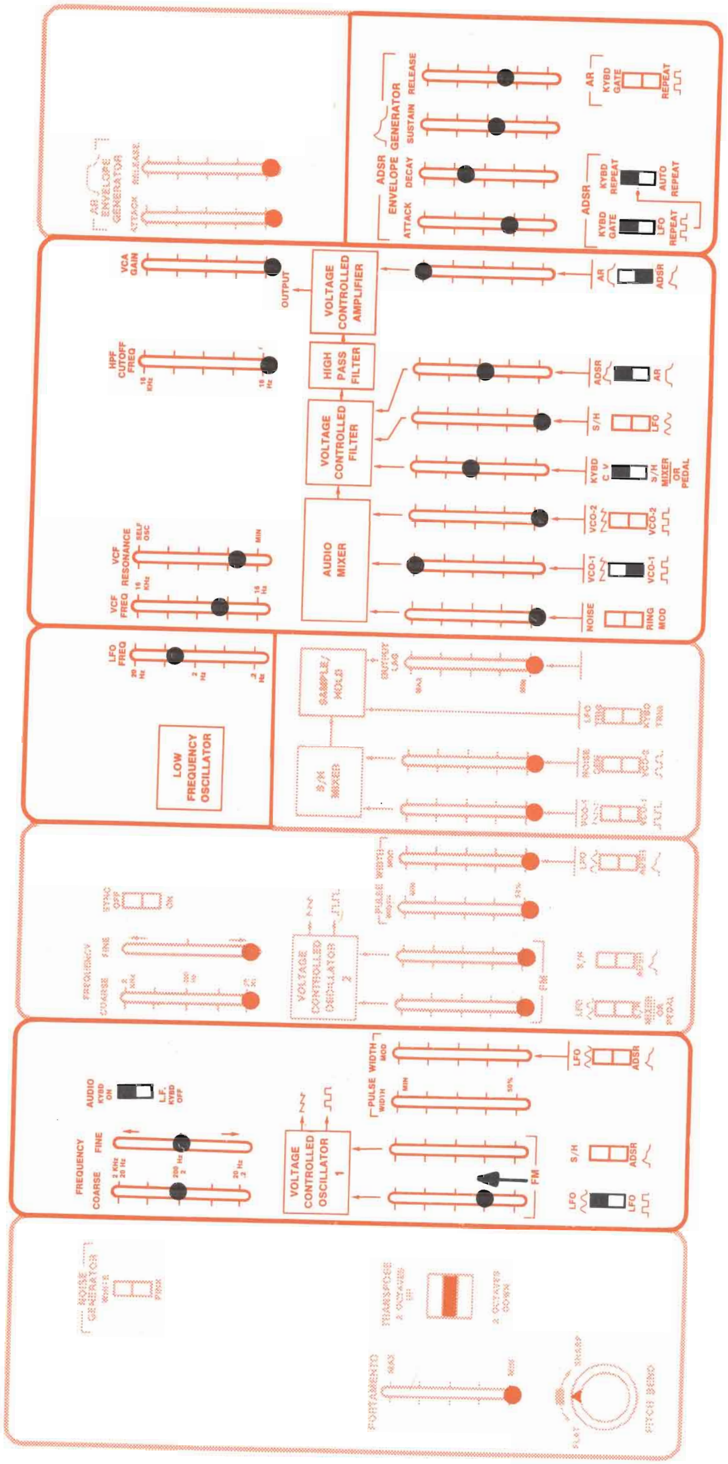


Figure 1.12. Clarinet.

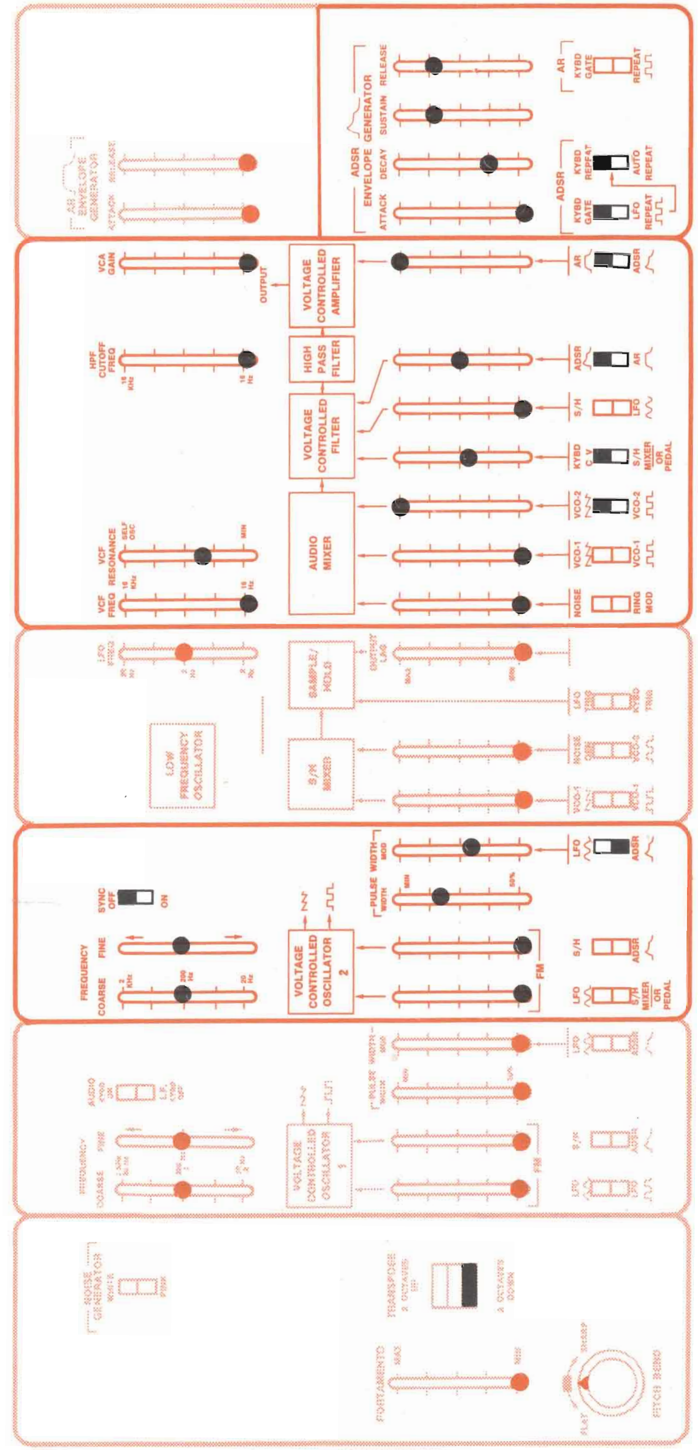


Figure 1.13. Electric bass.

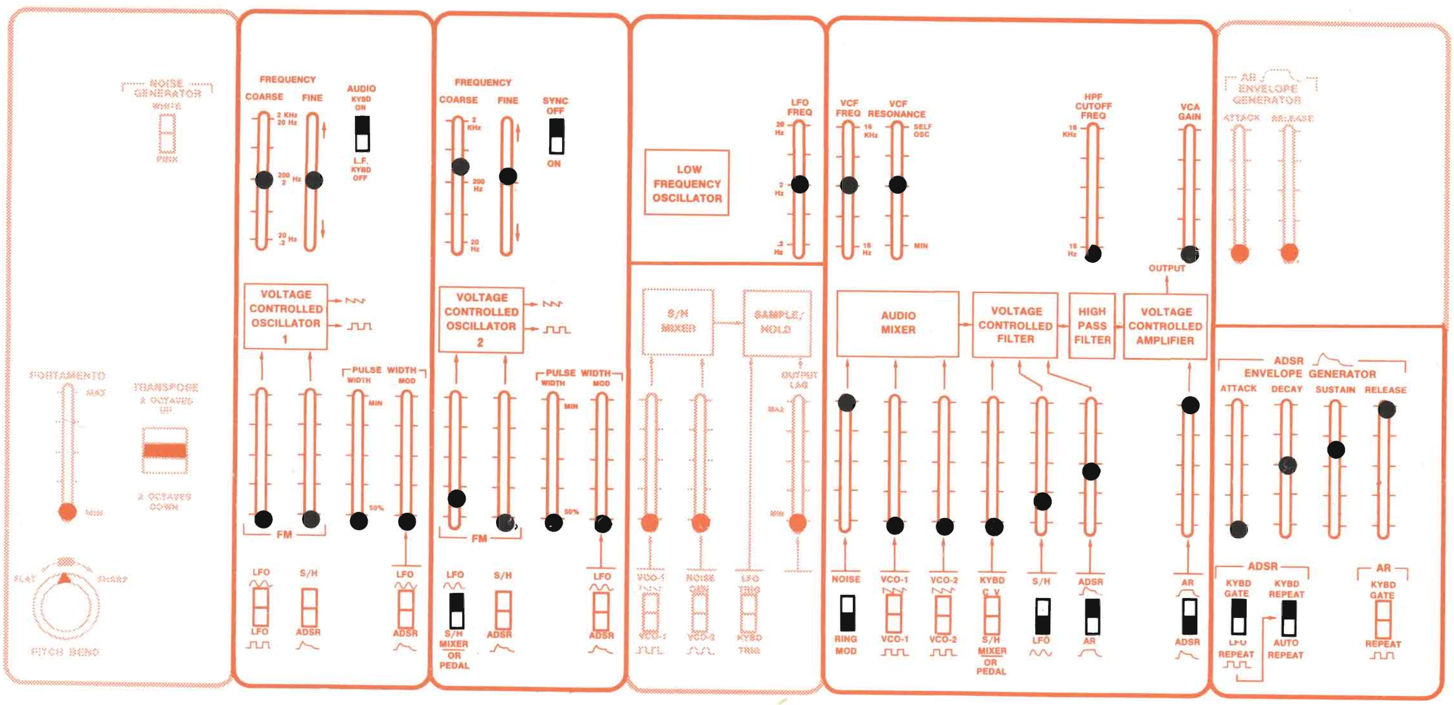


Figure 1.14. Gong/Chime.

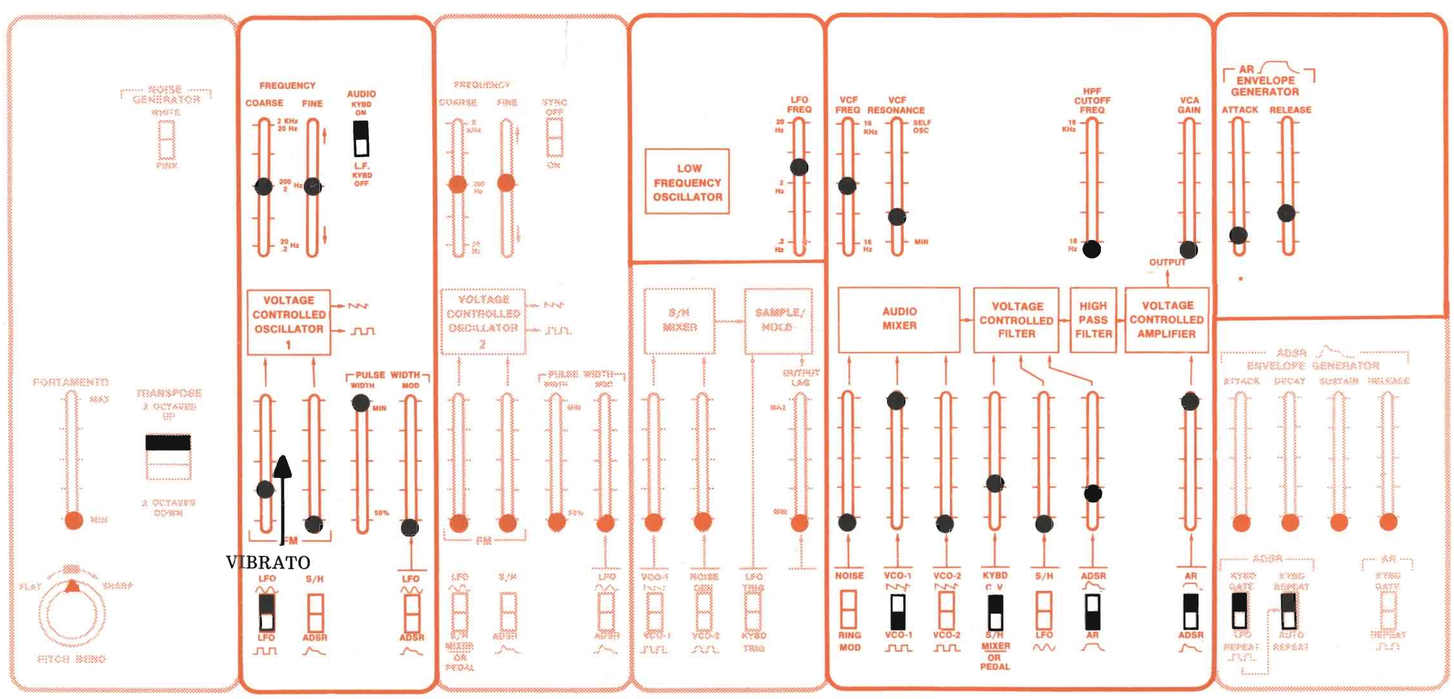


Figure 1.15. Violin.

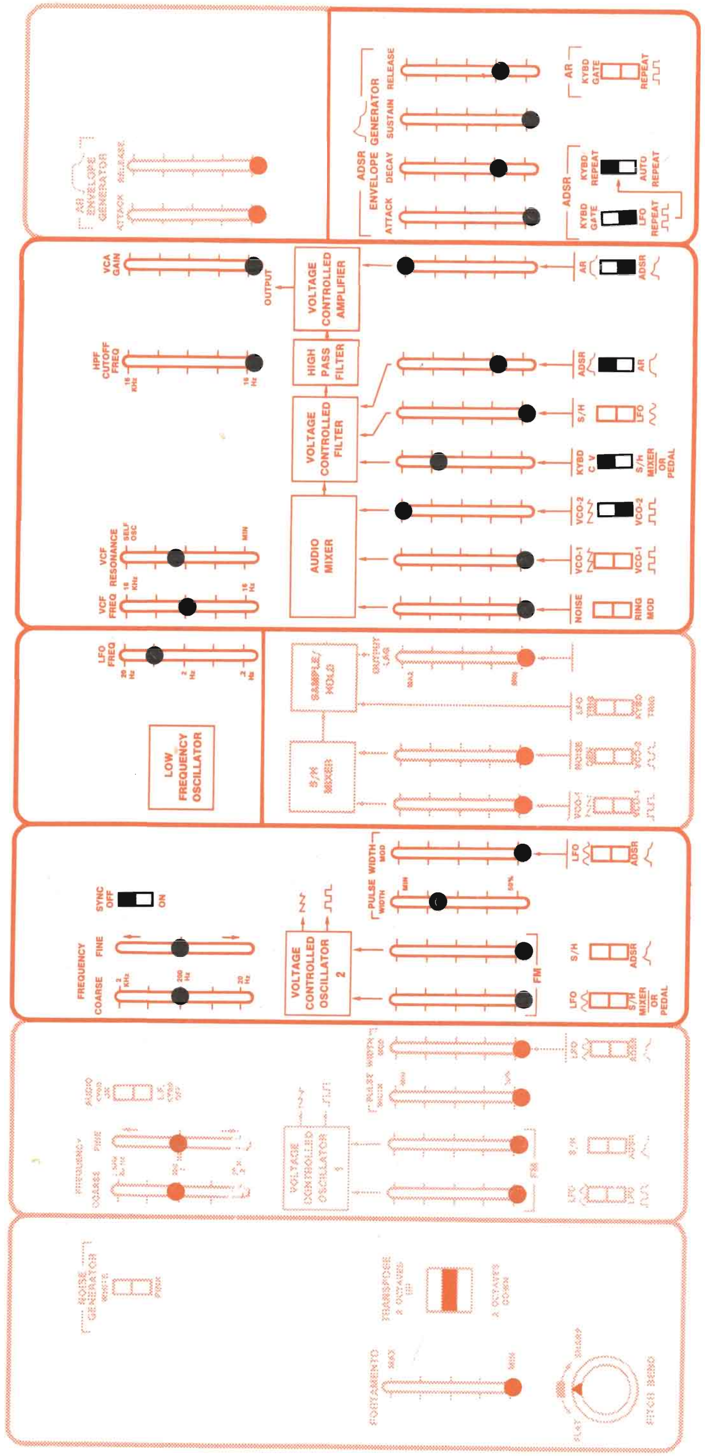


Figure 1.16. Banjo-repeat.

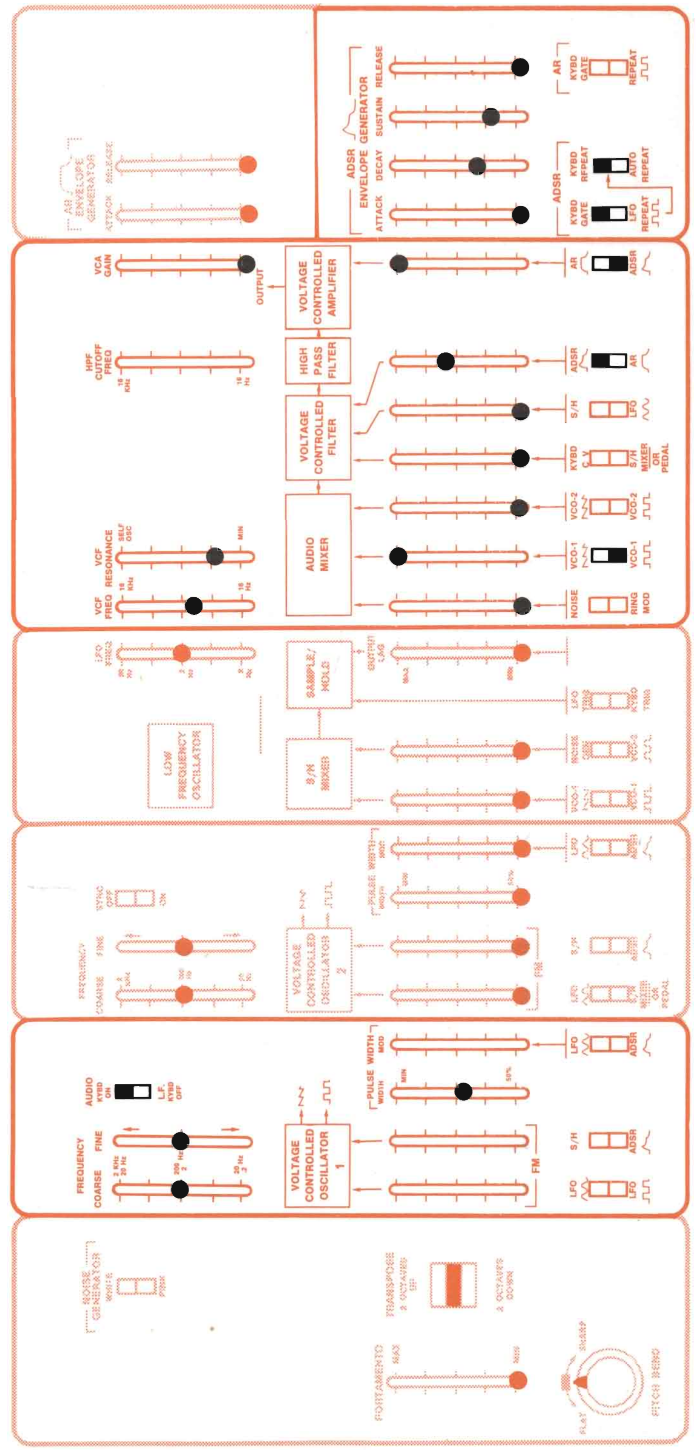


Figure 1.17. Guitar.

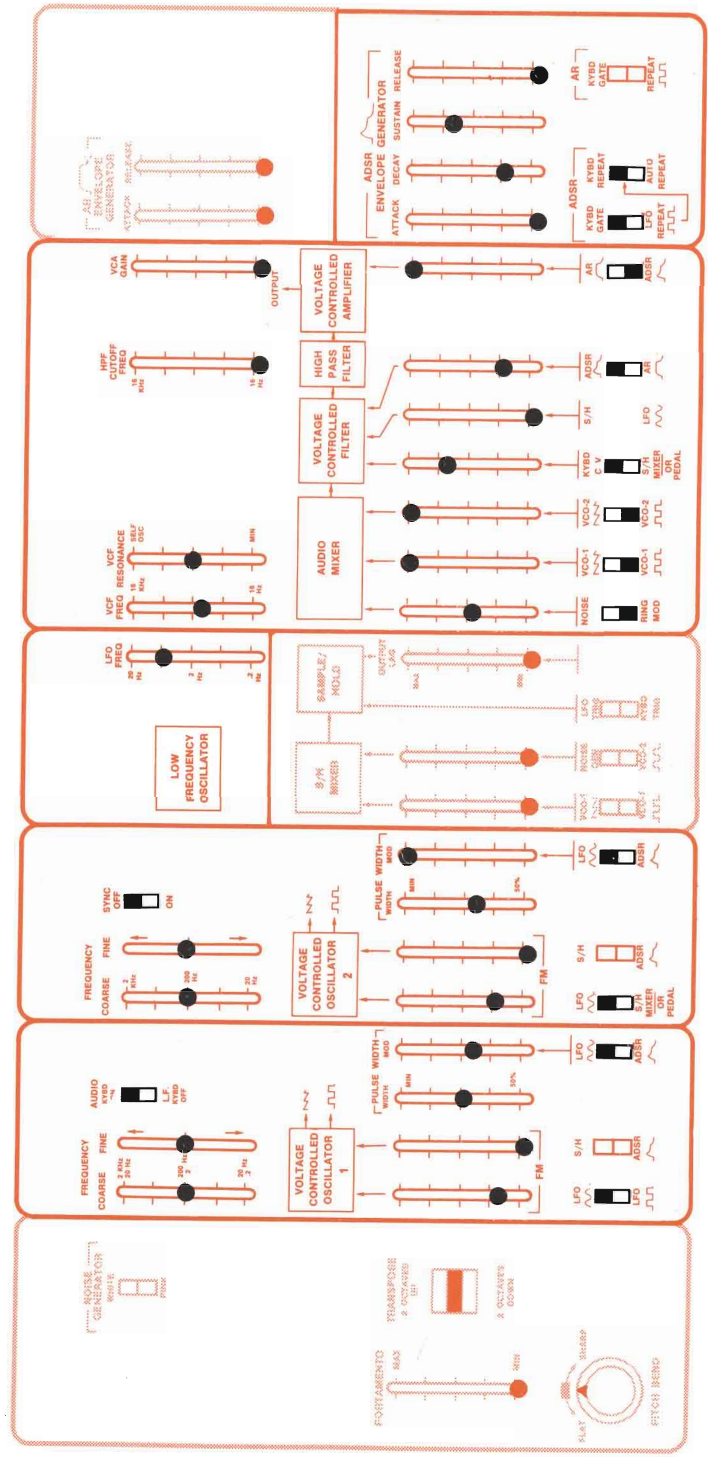


Figure 1.18. Jazz organ.

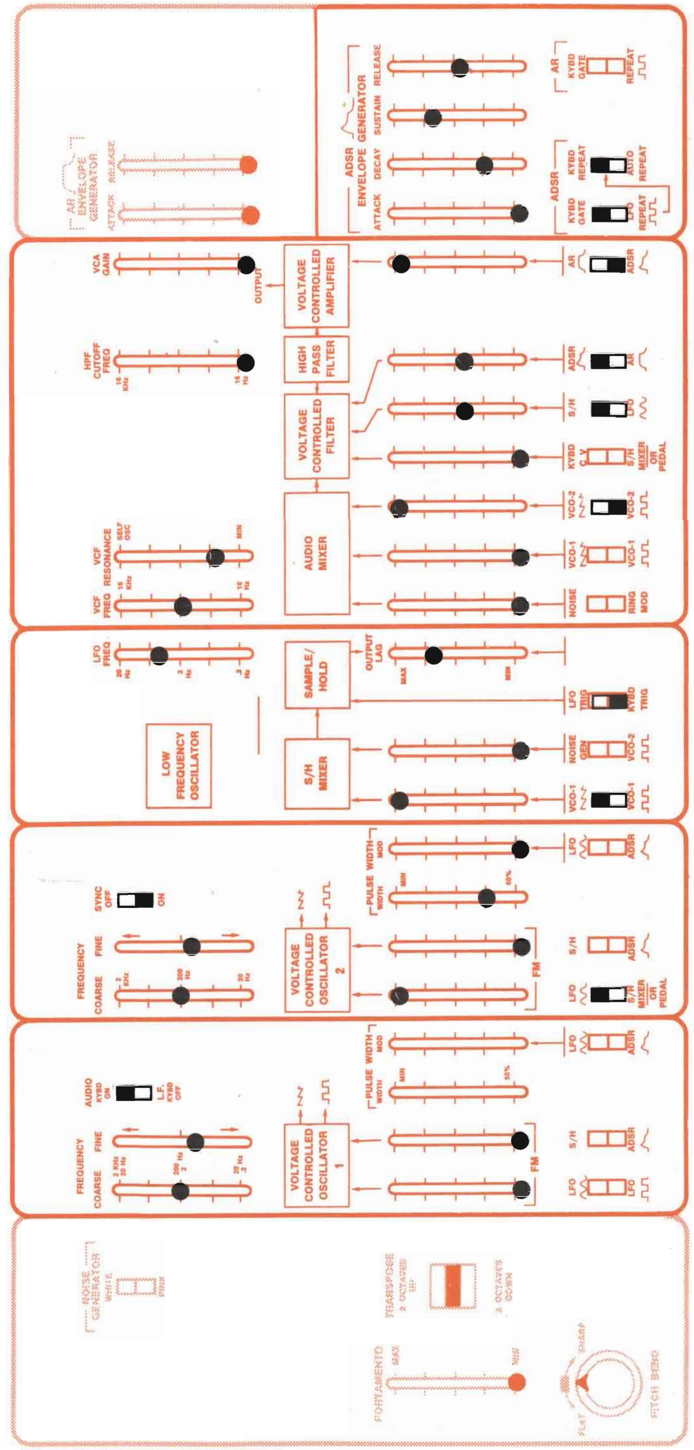


Figure 1.19. Guitar/Leslie.

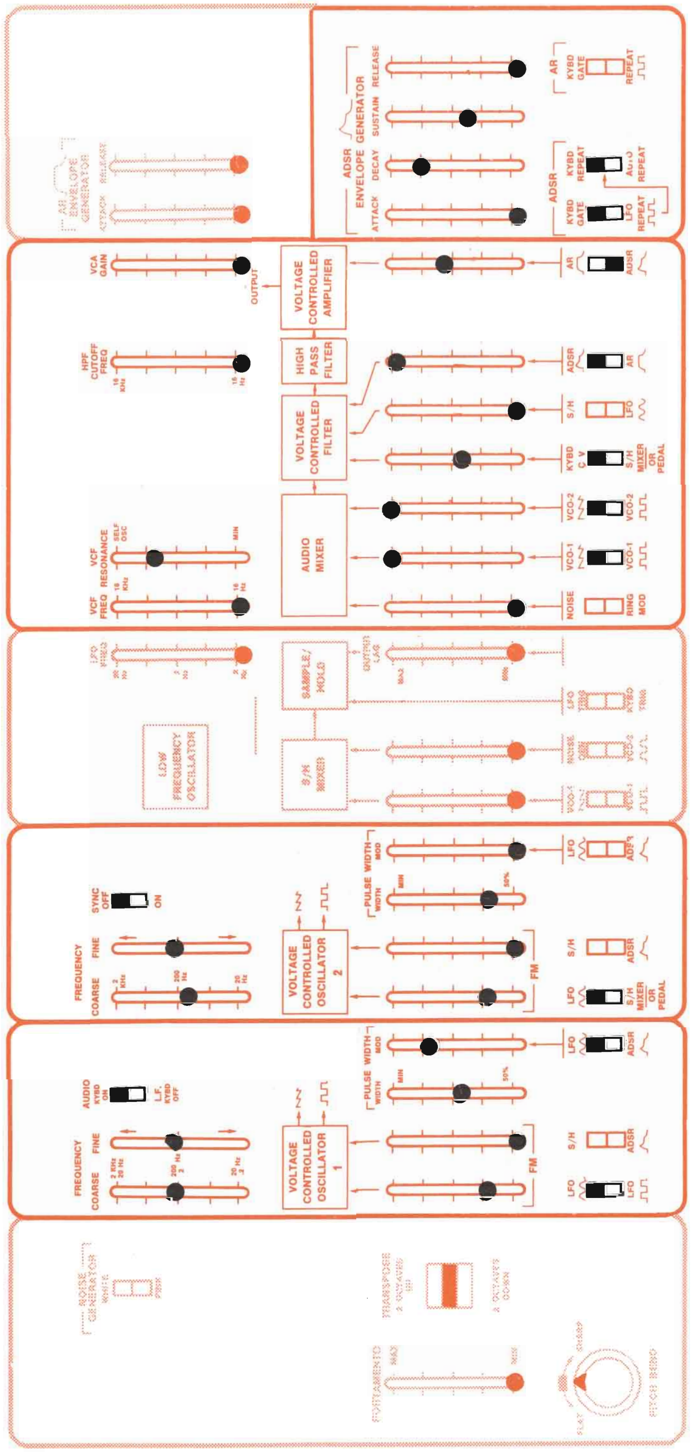


Figure 1.20. Wow sound.

Section 2: Melody

Have you ever asked another person to whistle or hum a tune which you were trying to recall? If so, you were asking the person to produce the pitches which make up the *melody* of a song. In a sense, melody is a kind of musical identification tag, as evidenced by our ability to recognize a large symphonic work by just one short melodic theme. Familiar songs are usually quickly remembered with the playing or singing of a few of the melody notes. You may have heard melody referred to as the "tune" of a song — it's just another common way of identifying the important aspect of melody.

The actual effect of a melody upon the listener, however, is dependent upon several interacting factors, all of which can be classified as submelodic component parts. To illustrate this concept, an analogy can be made to speaking; we — all — know — people — who — talk — in — a — dull — monotone — voice — just — like — this. You may know others who talk so fast when they are excited that theirwordsruntogetheralotlikethis. Inflection (the pitch of someone's voice), speed (tempo), rhythm (the way they say a phrase and the pause before continuing), and the basic tonal characteristic that makes your voice different from anyone else's (timbre) all play a part in the effect your words will have upon your listener. Similarly, in a melody, rhythm (the duration of the notes), tempo (speed), the relationship of the pitches (tonality), and the timbre (the qualities of sound that make it distinctive) interact to create a specific effect for the listener. A musician's awareness and control over these submelodic components will largely determine whether the melody is uninteresting, or even unpleasant, as opposed to being a melody that has all the potential of becoming a part of something wonderfully exciting.

Let's now look at some specific examples of how the character of melody is affected by the control of the musical aspects mentioned above. To begin, set the Odyssey controls as shown in Figure 2.1 and you'll be ready to examine melody in terms of rhythm.

Experiment 1: Rhythm and Melody

Using the patch shown in Figure 2.1, play the melody in Figure 2.2. Next, in every measure which contains four quarter-notes (measures 1, 5, 6, and 7) change every other quarter-note to a half-note, and play the melody again. Does the melody sound the same? The suggested change is not very great and the melody is probably still recognizable; however a radical change of time values in the melody of Figure 2.2 could eventually make the tune unfamiliar. Try this experiment with a friend: see who can make the simplest and most familiar melody unrecognizable by changing the rhythm.

By performing this experiment, you are actually demonstrating that melody and rhythm cannot be thought of as being mutually exclusive musical phenomena. Thus, the two fundamental qualities of musical sound — pitch and duration — form the raw material from which all melodies are fashioned.

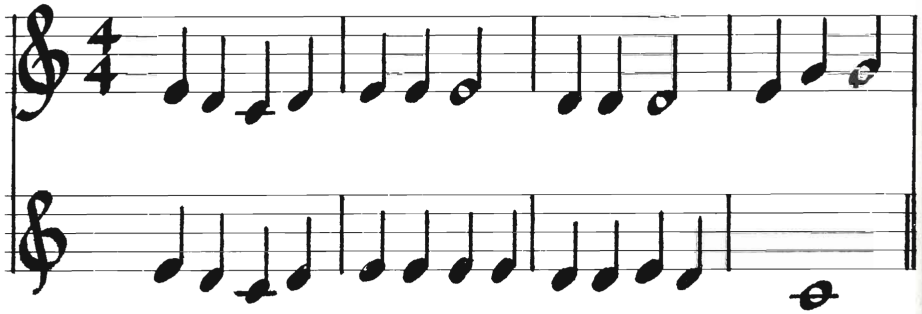


Figure 2.2. Merrily We Roll Along.

Experiment 2: Tempo and Melody

Another determining factor in how a melody will sound involves the speed/tempo at which the melody is played. Set the controls of the Odyssey as you did in the previous experiment and play the melody shown in Figure 2.3. Now play the melody as fast as you can at least two or three times, listening carefully as you play; then do just the opposite and play the music very slowly. Does the character of the melody change as the tempo changes?

If you wish, expand this concept of tempo and its effect on melody by playing a lovely ballad — choose one of your favorites — and play it at an unreasonably fast tempo. Do just the reverse with a composition normally performed rapidly — "Deck the Halls," for example — and play it very slowly. Regardless of how accurately the notes (pitches) are played, the change of tempo in instances such as these has a dramatic influence on how the melody is perceived.

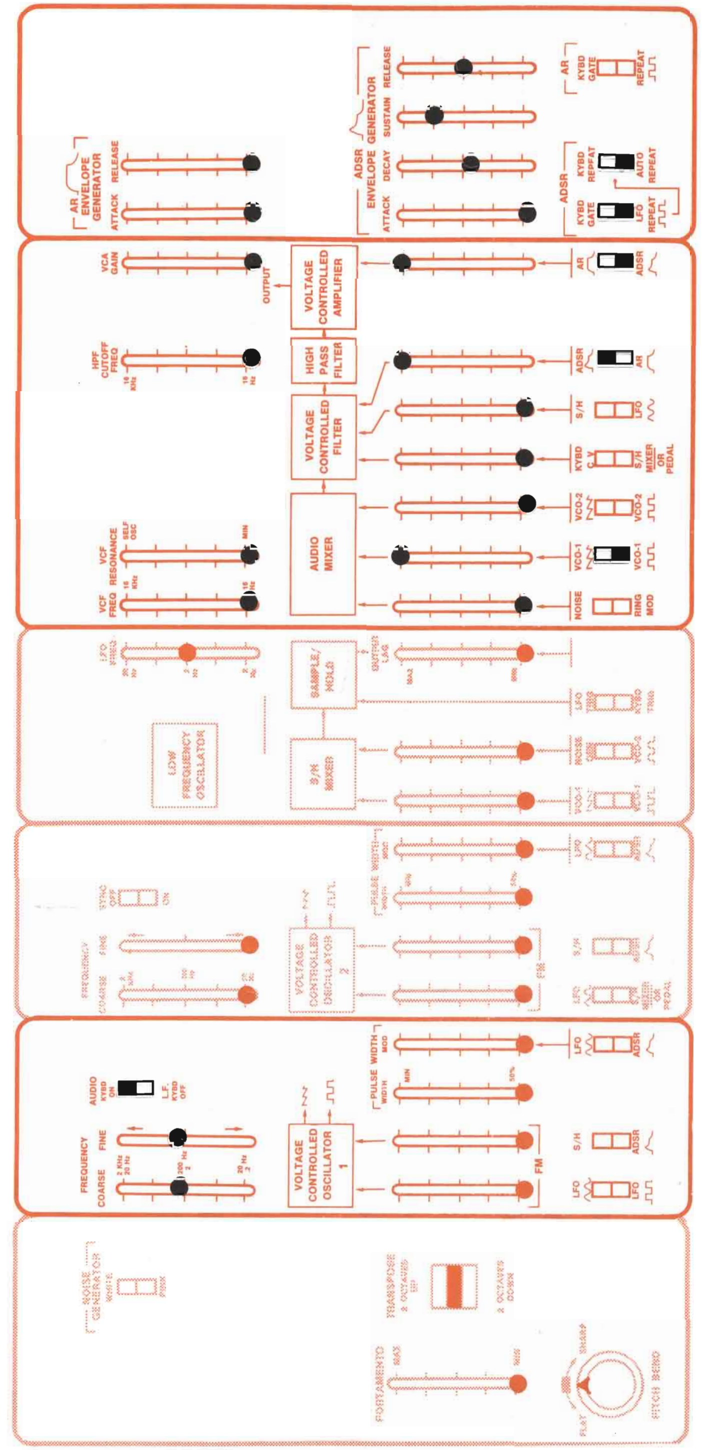


Figure 2.1. Basic patch for rhythm experiments.

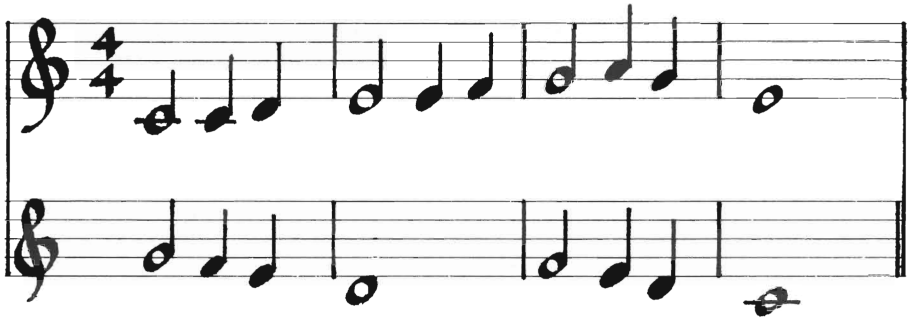


Figure 2.3. Long, Long Ago.

Experiment 3: Tonality and Melody

To the two aspects of melody already discussed, rhythm and tempo, let's add a third: *tonality.* The intervalic relationship of the series of pitches that make up a melody determines the prevailing tonality of that melody. This concept is more easily demonstrated than verbalized. To try altering the tonality of a familiar melody, first set up the patch shown in Figure 2.4.

The first musical example, shown in Figure 2.5, is the familiar

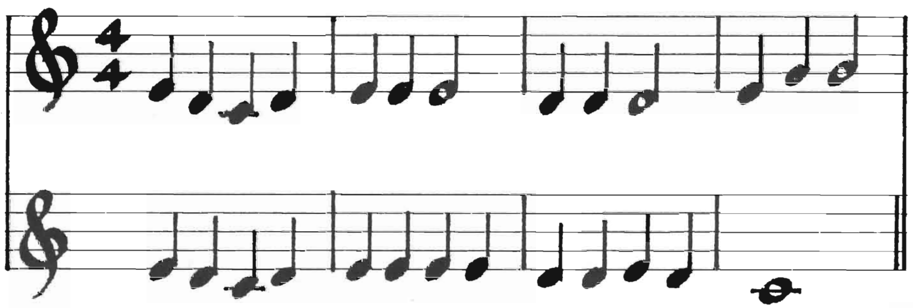


Figure 2.5. Merrily We Roll Along.

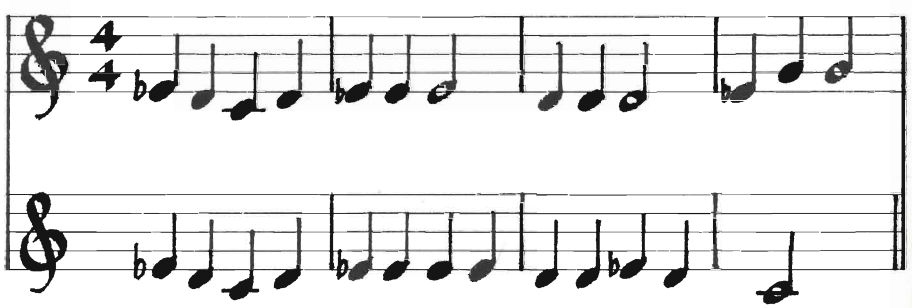



Figure 2.6. Minor-chord version of Figure 2.5.

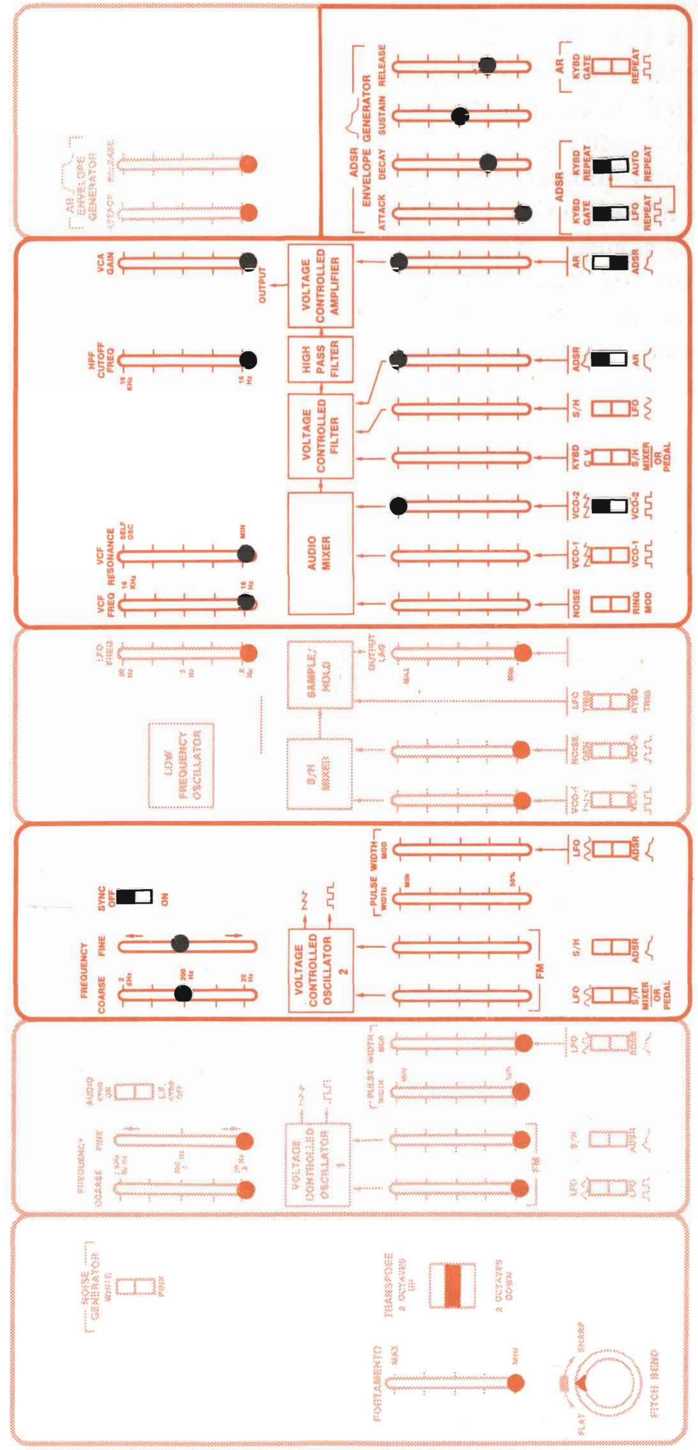


Figure 2.4. Basic patch for tonality experiments.

child's tune, "Merrily We Roll Along." You've already played this melody in Experiment 1, at which time you discovered the effect that changing the rhythm can have upon melody. This time play the melody as modified in Figure 2.6. As you play, note the difference that the change in tonality makes.

When you compare the two examples, you'll undoubtedly discover that the melody takes on a completely different character when the tonality is changed. If you have a knowledge of chords, you might try playing the first melody accompanied by major chords, the second by minor chords. This will augment the effect of the tonality change. Don't hesitate to experiment further by trying tonality changes in other familiar melodies.

Experiment 4: Timbre and Melody

The final experiment demonstrates how timbre interacts with melody to create a variety of different melodic effects. You can exercise almost total control over the timbre of a melody when playing it upon the Odyssey; this control — through the selection of waveforms, the use of the VCF and HPF, and the settings of the AR and/or ADSR generators — permits you to simulate traditional instrumental timbres or to go far beyond their bounds. The point is that you should think about playing expressively and making full use of the controllable synthesizer functions when playing the Odyssey, just as a trumpet player can add expression to a melody by permitting his tone to grow brighter during sustained notes. In each case, the expressive interpretation of the melody, whether on a horn or on the Odyssey, results in a more interesting musical experience for both the player and the listener.

To demonstrate the effects that a change in timbre (timbre being all those characteristics which distinguish any particular sound from another sound) set up the patch shown in Figure 2.7. Let's first try changing the timbre of this sound by modifying the attack and decay characteristics. Using the patch as shown, play the melody illustrated in Figure 2.8. Listen carefully as you play, so that you'll recall the basic effect of this melody as you play the next version.

Now, reset the ADSR controls to different positions, playing the melody again with each new setting. You'll find that raising the attack slider to its uppermost position makes it virtually impossible to play a composition with notes coming in rapid succession — certainly a change in the melodic character. Try each of the envelopes in Figure 2.9. Try experimenting with different settings of the VCF Resonance control and Portamento control, too.

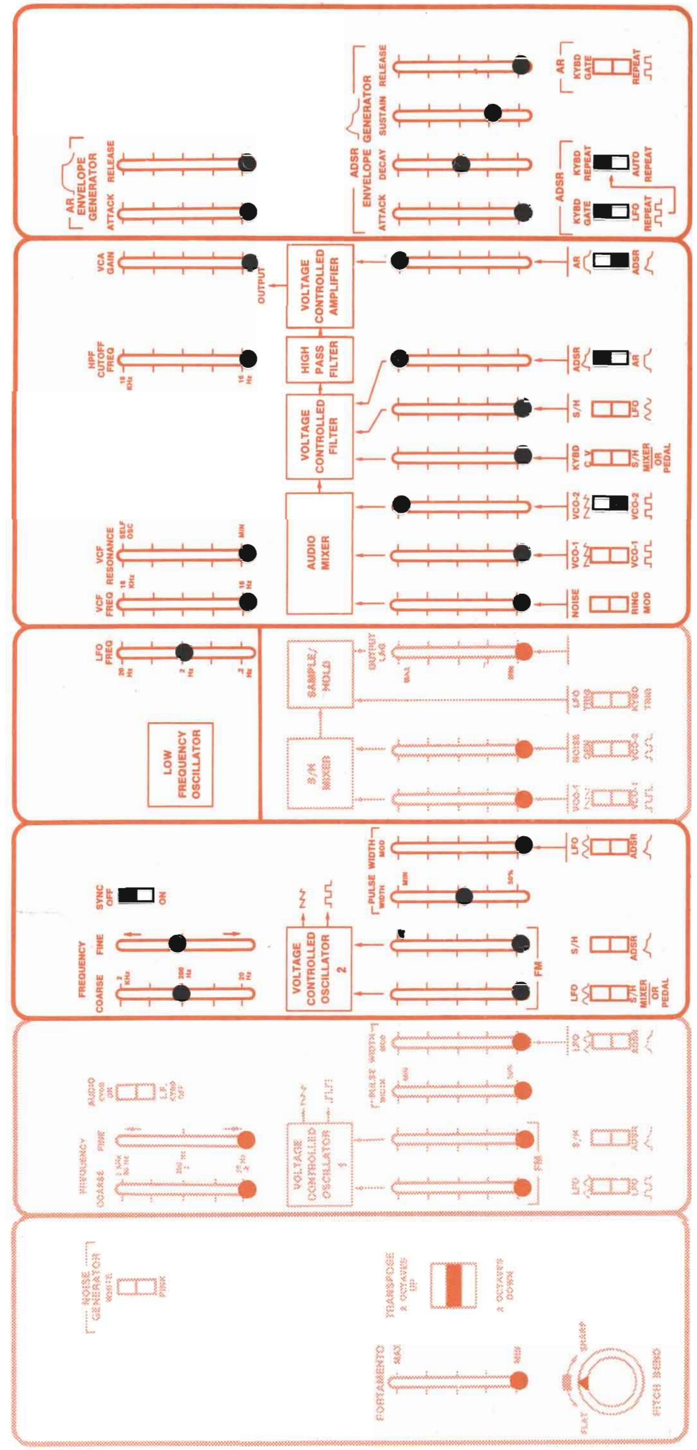


Figure 2.7. Basic patch for timbre experiments.

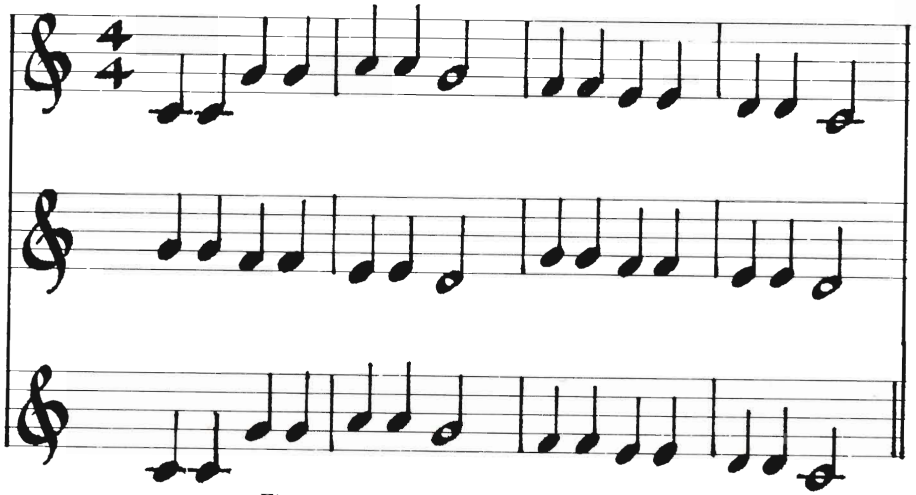



Figure 2.8. Twinkle, Twinkle, Little Star.

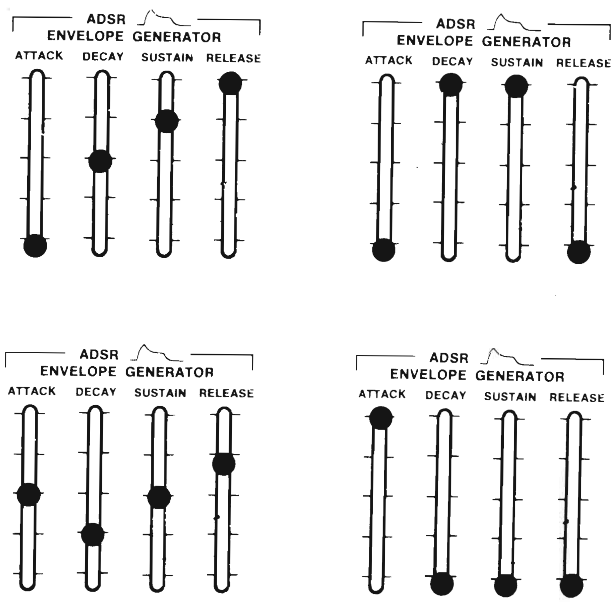


Figure 2.9. Variations on setting in Figure 2.7.

A similar experiment can be performed by recording a friend who plays the trumpet. Turn on the recorder and have him play a single sustained note, approximately five seconds in length. Then edit and splice the tape so that the attack and decay of the tone are eliminated — take only the central portion of the piece of tape, approximately two or three seconds. Play back the result and listen to the difference. If possible, try this experiment with other instruments — a clarinet, a violin, even a tuba. In some instances you'll find that the net effect is startling — the instrument is hardly recognizable.

At this point you may also wish to review the information provided in Part II on the envelope generators. This material will indirectly suggest many ways in which the character of the melody in Figure 2.8 may be altered. By all means try the ADSR control of the pulse width and ADSR-controlled frequency modulation. The dynamic-waveform concept and its musical usefulness in terms of melody will become readily apparent.

In addition to the above experiments, you should also try using several different pulse widths, from 10% to 50% (a square wave), as well as the sawtooth wave. Also try adding portamento and vibrato as you experiment (see Part II for patch settings, if necessary). The interaction of these effects and the effect upon the melody will come into clear focus as you experiment.

While this discussion has systematically examined a variety of ways to alter melody, the total musical effect of the melody often depends upon all of the above considerations. The performer has the final say as to the particular components to be highlighted, subject to the limitations of the instrument he plays.

Through this series of experiments, you have almost certainly come to realize that arrangers are very much aware of specific instruments and how they can be used most effectively to enrich melodic lines. This is *orchestration*. For instance, you'll rarely hear a tuba playing a love theme with a string orchestra; nor, on the other hand, would a violin play the foundational, or bass, part. Also, it's safe to assume that the "Minute Waltz" is not a favorite piece for trombone, while a pianist cannot produce a beautiful portamento effect on the piano. It follows that the timbral assets and limitations of an instrument are important reasons why an arranger chooses one instrument over another to play a particular part.

As a conclusion to this discussion of melody, try "orchestrating" a number of familiar melodies, using several of the instrumental patches that appeared in Section 1 of Part III. Decide which instrument(s) you find most appropriate for a lullaby, which for a pop tune, and which for a musical television commercial most students have heard. While you may not agree on one single instrument that sounds best, you'll almost undoubtedly agree on two or three that are totally inappropriate.

Section 3: Harmony

In contrast to melody, which we established as an ordered series of pitches making up the basic theme or "tune" of a song, *harmony* is present as musical tones which are sounded simultaneously. Therefore, whenever you play two or more notes of different pitches, harmony is automatically implied. As notes are sounded simultaneously to form various intervals (two notes) and chords (three or more notes), the resulting harmony further defines the tonality of a composition. More important, however, than any technical definition of harmony is its function: that is, to provide support for the melody, ranging from one simple counter-melodic line up to and including complex series of chords.

Because almost all music is dependent upon intervals and the relationships between intervals, this section of the text is devoted to a systematic study of intervals, how they contribute to harmony, what is meant by being "in tune" or "out of tune," and how you may improve your own ear for harmony and frequency relationships.

Experiment 1: Ear Training

The ability to distinguish minute differences in pitch is almost indispensable to any musician, regardless of the kind of musical activity in which he may be engaged. First, playing in tune is imperative for those persons who play tuneable instruments — strings, reeds, and brass instruments, for example. Musicians playing fixed-pitch instruments, such as the piano, also have need for a well-trained ear, however; accurately determining chord changes without music, rehearsing singers, and generally filling the role of musical director, as the keyboard player often does, all require attentiveness to intervalic and harmonic relationships.

The Odyssey is an excellent tool for providing ear-training experiences, as its infinitely variable, wide-range oscillators permit virtually any tuning. In addition, the VCF will be employed later in this section to "pick apart" the harmonics common to various tunings, dramatically demonstrating relationships between various intervals that could only be talked about theoretically without the use of a synthesizer.

Begin Experiment 1 by setting up the patch shown on Figure 3.1. Having now established a basic frequency setting on VCO-1, press the key on the Odyssey keyboard indicated by a circle in Figure 3.2. While this pitch continues to sound, raise the VCO-2 attenuator into the Audio Mixer and tune VCO-2 to the same pitch as VCO-1. To do so, you should first move the coarse-tuning slider to approximately the same position as the corresponding slider on VCO-1; then, continue

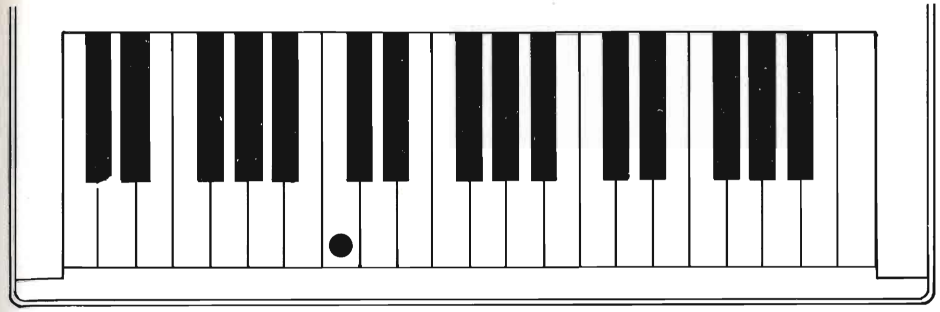


Figure 3.2. Tuning unisons.

tuning by adjusting the fine-tune slider on VCO-2. When the two tones are nearly in tune, you will hear a pronounced wavering sound, or "beat" — the tones will sound as if they are growing louder and softer. The speed of the beat depends upon how far out of tune the oscillators are. The closer they are to being in tune, the slower the beat will become. Continue to tune until the beats finally slow down and then stop; the pitches/frequencies of VCO-1 and VCO-2 are now identical, or in tune. In this instance, they are tuned in *unison*, or to the same pitch.

The reason that you hear beats when the oscillators are slightly out of tune is because two waveforms that are at slightly different frequencies produce an alternate cancelling effect, thus causing a cyclic change in amplitude, resulting in beats. This effect is a useful one, since it not only aids us in tuning unisons accurately, but also will permit the accurate tuning of other intervals containing common harmonics. While the fundamental frequencies of two tones may be different, if they have harmonics in common, these harmonics will beat as the interval being tuned is brought nearer and nearer to being in tune.

Now continue this exercise in ear training by tuning the oscillators to an interval of a major third. This commonly used musical interval is represented on the keyboard in Figure 3.3 by the circle and the triangle; the key upon which the triangle is placed is a major third above the basic unison pitch already tuned.

To tune this interval, listen first to the pitch produced when the key with the circle on it has been pressed down. Then press the triangle-coded key. Alternate between the two until, while listening to first pitch, you can easily hum the second pitch (that of the key with the triangle on it). When you can do this, tune VCO-2 to the pitch you are humming. *Note:* you should not attempt to do this with the coarse-tuning slider, as it will take you beyond the interval you want

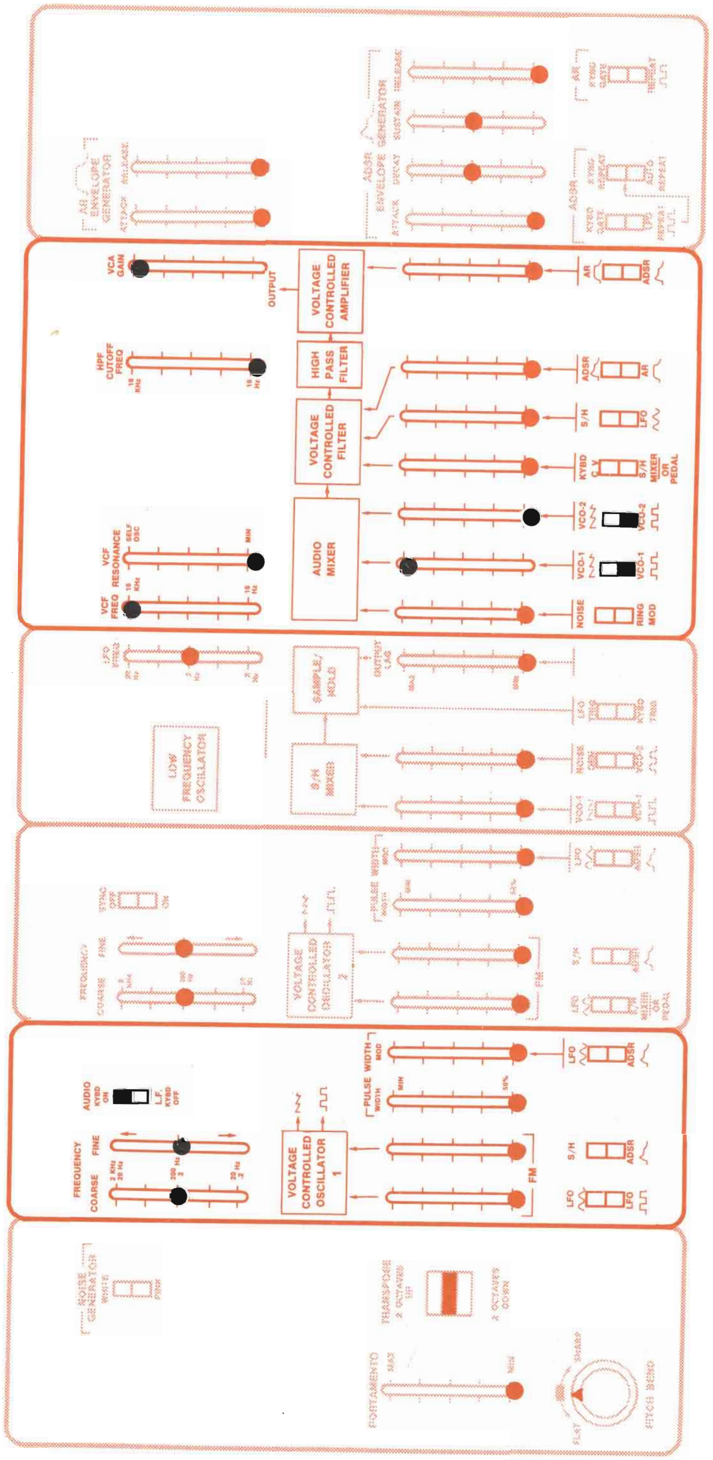


Figure 3.1. Basic patch for ear training experiments.

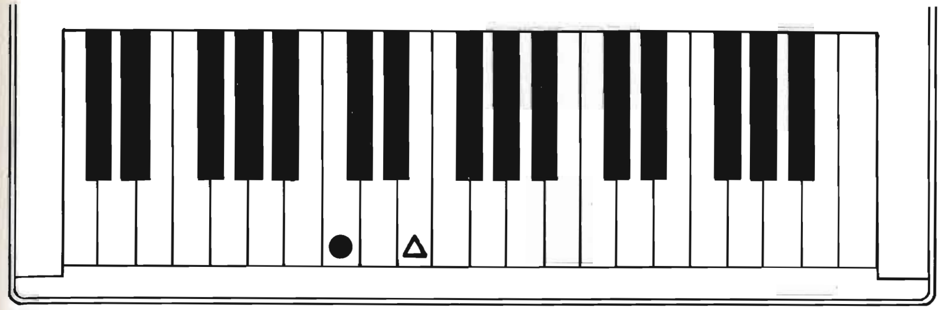


Figure 3.3. A major third.

to tune very quickly. Instead, use the fine-tune slider of VCO-2, raising it slowly until you hear the pitch of the major third come into tune.

Use the same procedure to tune the major fifth, using the keys indicated by the circle and the square in Figure 3.4. Listen for the beat of the harmonics the two frequencies have in common. You should also practice tuning two additional intervals: a fourth, indicated in Figure 3.5 by the circle and the star, and an octave, indicated by the first circle and last two circles in the same illustration.

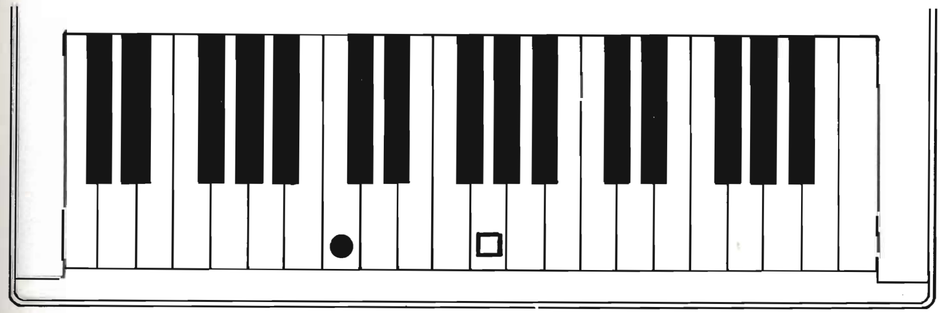


Figure 3.4. A major fifth.

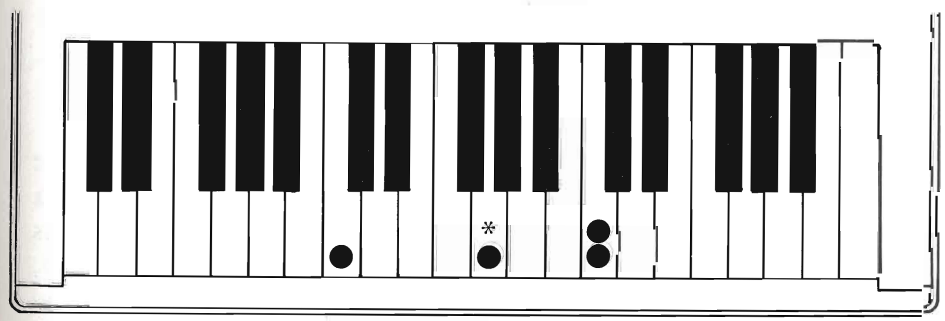


Figure 3.5. A fourth and an octave.

Experiment 2: Using the VCF to Listen to Specific Harmonics

Without the use of an electronic music synthesizer, the study of harmonics is one which has had to be taken pretty much on faith alone. Certainly, theory existed which seemed to demonstrate the presence of harmonics and their effect upon the timbre of the sounds you hear. Helmholtz first supplied the theoretical basis for such an understanding. To illustrate, let's use a vibrating string, as shown in Figure 3.6. It has been proven that a vibrating body, such as the string shown in this figure, does much more than simply vibrate back and

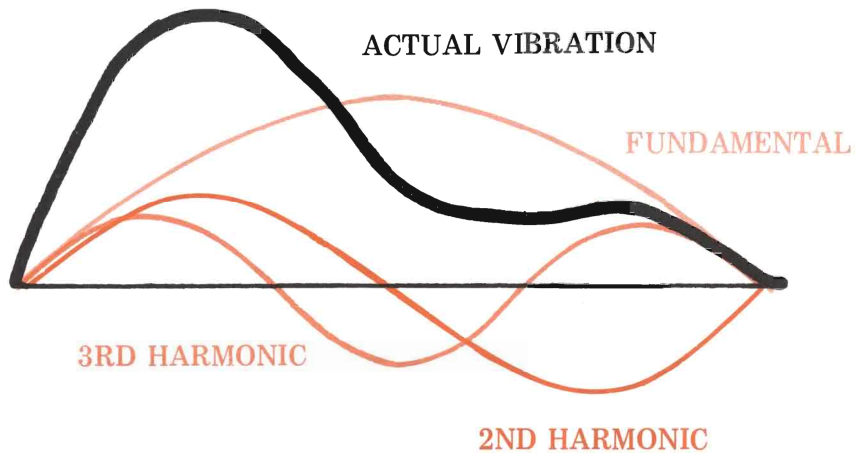


Figure 3.6. Vibrating string.

forth as a whole. Indeed, simultaneous but distinctly separate vibrations occur, so that the string is also vibrating in sections of one-half, one-third, one-fourth, and so on. Each of the vibrations, and the frequencies the smaller sections provide, becomes weaker as the divisions of the string become smaller. The pitch that is most prominent — that pitch resulting from the first and largest motion of vibration — has the greatest amplitude and is called the fundamental. This, of course, provides the basic pitch of the note that you hear. All of the frequencies produced by smaller divisions of the string are called overtones, or harmonics.

While this information has been known since the latter half of the 19th century, it has never been something that could be easily demonstrated. Through the use of the Odyssey's low-pass and high-pass filters, however, it is now possible to single out virtually any harmonic component of a waveform, dramatically demonstrating its presence. It is also possible, when tuning intervals other than unison, to sweep, or scan, the harmonics of both waves — picking out common harmonics having a potential beat (if the interval is slightly out of tune), thus demonstrating why you perceive a beat when tuning an interval such as a fifth. Since it is not possible for the fundamentals to beat, since they are two different frequencies, any beat perceived when tuning intervals other than unison must be because of harmonics held in common between the two notes. The following exercises will graphically demonstrate this fact.

First, set up the patch shown in Figure 3.7. Note that the resonance on the VCF is very high, in order to elicit a peaked response or band-pass effect around the narrow band of frequencies determined by the setting of the initial filter frequency slider. Now, by raising the initial filter frequency slider (marked VCF Freq), you will be able to scan, or pick out, the harmonic components of the waveform being sent to the filter by VCO-1. The pitches of the harmonics you will hear, from the fundamental up through the ninth harmonic, are shown on the staff in Figure 3.8.

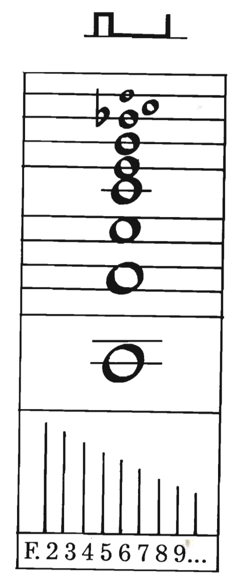


Figure 3.8. First nine harmonics of a pulse wave.

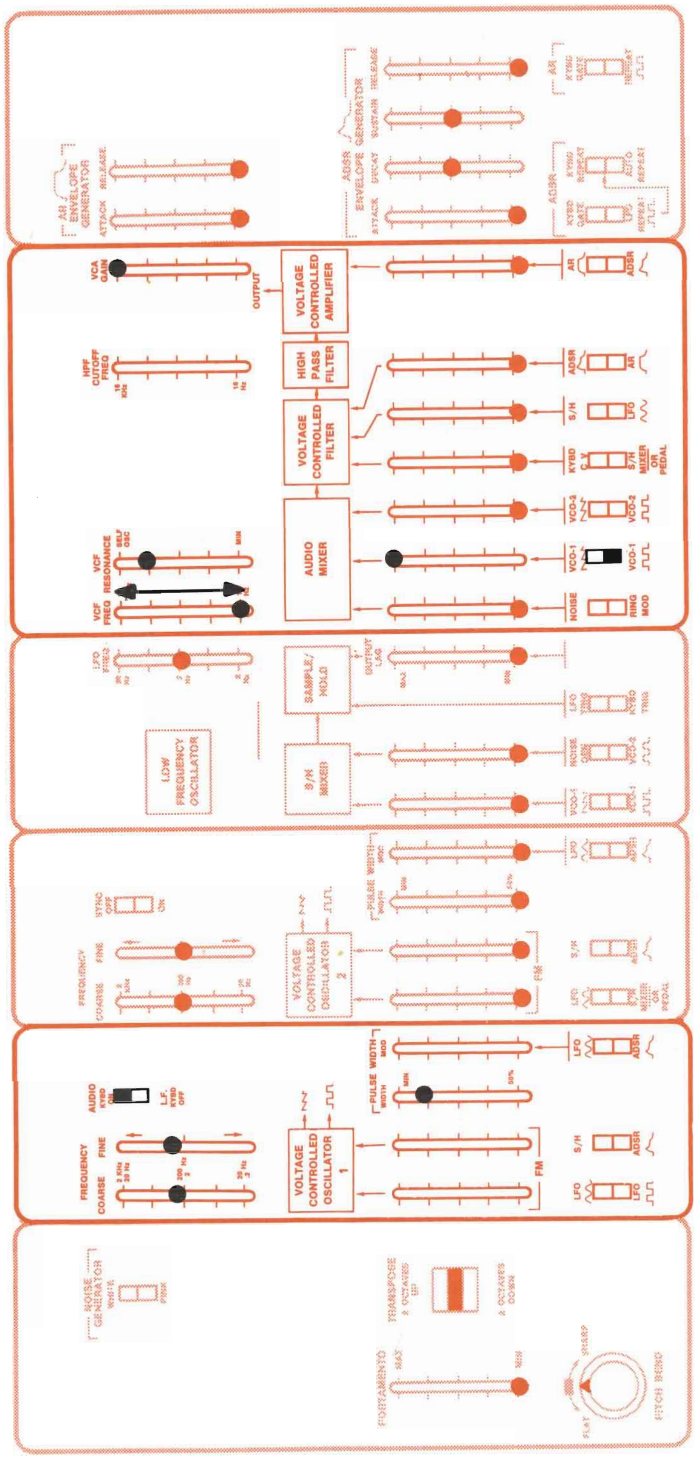


Figure 3.7. Basic patch for harmonic scanning experiments.

Having now heard the fundamental and all of the harmonics of the basic waveform, tune VCO-2 to unison. When you have achieved a good unison tuning, again gradually raise the VCF Freq slider, picking out the fundamental, then the second harmonic, then the third harmonic, and so on. If you hear a beat at any point, this means that a perfect unison has not been achieved and you should sharpen your tuning.

After scanning the harmonics of the unison waveforms with the oscillators in tune, then slightly detune VCO-2 by moving the fine-tuning slider up just slightly. With the VCF Freq slider open most of the way, you should be able to detune easily to get a perceptible beat of one or two cycles per second. With the oscillators detuned in this manner, scan the harmonics again. You'll note that with unison tuning, the fundamental and every harmonic will beat. Logically, you will understand that this is as it should be, since two fundamentals of exactly the same frequency will have the same harmonics. Figure 3.9 illustrates this.

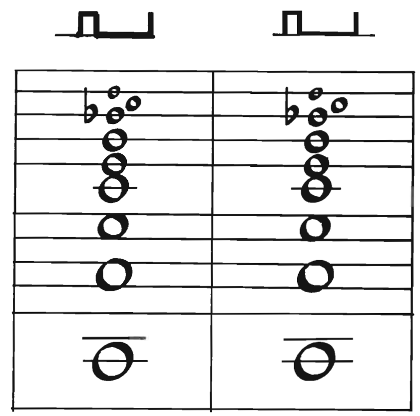


Figure 3.9. Harmonics in unison tuning.

Now tune VCO-2 a fifth above VCO-1, as you did in Experiment 1 of this section. When you have achieved a good tuning, again detune slightly so that you get a perceptible beat. As discussed earlier, the beat you are now hearing is the result of the harmonics that the two frequencies have in common (Figure 3.10).

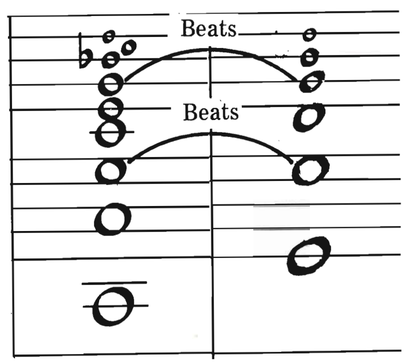


Figure 3.10. Harmonics when VCO-1 and VCO-2 are a fifth apart.

If you now scan the harmonics, again using the low-pass VCF, you will find that every harmonic does not beat, as it did when the oscillators were tuned to unison. This is because with a fundamental frequency of VCO-1 being different from the fundamental frequency of VCO-2, all of the resulting harmonics are not common. As you can demonstrate by scanning with the VCF, every third harmonic of VCO-1 will beat with every second harmonic of VCO-2. Figure 3.11 indicates why this is so; if a note appears on the same line for both VCO-1 and VCO-2, a potential beat exists between those harmonics if the interval is not perfectly tuned. On the other hand, on those lines and spaces of the staff where VCO-1 and VCO-2 do not *both* have a note, indicating a harmonic, there is no beat (potential or otherwise) and you should hear a steady, even harmonic pitch.

The chart in Figure 3.11 illustrates the same concept, but uses arbitrary frequencies of 100 Hz (VCO-1) and 150 Hz (VCO-2) for the purposes of numerical simplicity; the relationship between these two round numbers, however, is the same as that of any particular pitch and that pitch a fifth above it. Therefore, the harmonic relationships expressed in the chart will hold true whether you tune the first oscillator to middle C, A-440, or any other frequency.

VCO - 1 (100 Hz) Potential Harmonics	100	200	300	400	500	600	
		Beats			Beats		etcetera.
VCO - 2 (150 Hz) Potential Harmonics	150	300	450	600	750	900	

Figure 3.11. Frequencies of harmonics of two pitches a fifth apart.

Experiment 3: Using the High-Pass Filter to Eliminate Certain Frequencies

A kind of reverse experiment can be performed by using the Odyssey's HPF (High-Pass Filter). In this experiment, you will raise the HPF Cutoff Freq slider in order to eliminate the harmonics from "the bottom up." Set up the patch shown in Figure 3.12. You'll hear a continuous tone, the sound of the narrow pulse wave of VCO-1. Now, gradually raise the HPF slider, listening carefully as each of the harmonics drops out. The first to go, of course, should be the fundamental — the basic pitch itself.

The musical example shown in Figure 3.13 illustrates what is happening in terms of the fundamental and all harmonics shown on a staff. Note that an interesting effect occurs after you have filtered out the fundamental frequency; despite the fact that you have filtered the frequency which provides/provided the basic pitch reference, it is

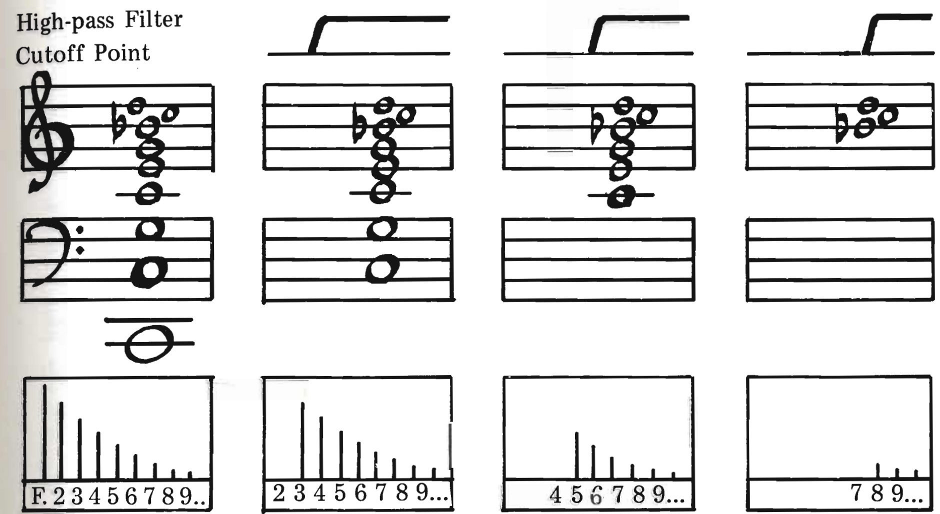


Figure 3.13. What happens when harmonics are filtered.

still implied — you are still hearing the original pitch determined by the setting of the oscillator and the last note pressed on the keyboard. You can go even further and remove many of the lower harmonics without altering this. This effect is called *residual pitch.* If this tone is interrupted, however, and then resumed — as it might be if you closed the VCA, and then returned to the Odyssey five minutes later and reopened it — the sensation of pitch will be completely altered. Instead of hearing the residual pitch based upon the original fundamental frequency, you will hear the *formant pitch,* which will lie approximately in the region of the strongest remaining harmonics.

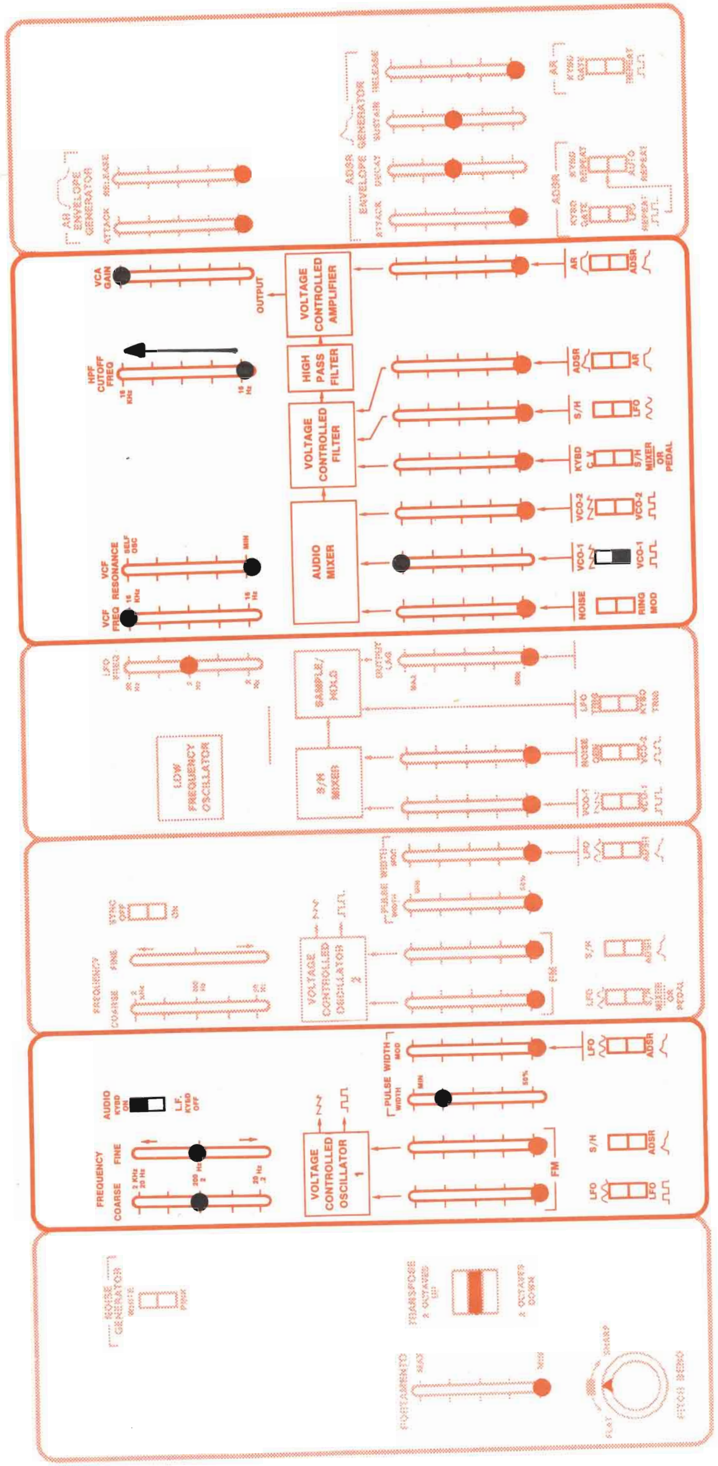


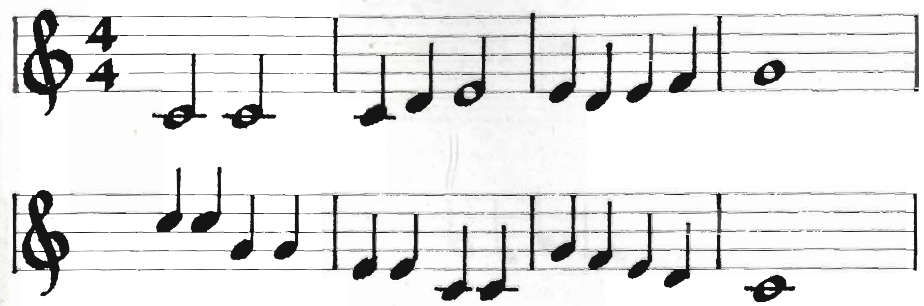
Figure 3.12. Basic patch for high-pass filtering experiment.

Experiment 4: Counter-Melody

You'll recall that in the beginning of this section we said that harmony is implied any time that two or more notes are sounded. Counter-melody is an excellent demonstration of this fact, providing the simplest of exercises in two-note harmony. Many of the "rounds" that you sang as a child depend upon this kind of effect.

The round shown in Figure 3.14 will provide an adequate demonstration. This music may be performed in two ways: if you are using the Odyssey with a tape recorder, record the first part (Part A); then, as the recorded portion is being played back, play Part B along with the tape. If you are not using tape equipment, you can simply have one student play Part A in the lowest octave of the Odyssey's keyboard while a second plays Part B in the highest octave.

PART A



PART B

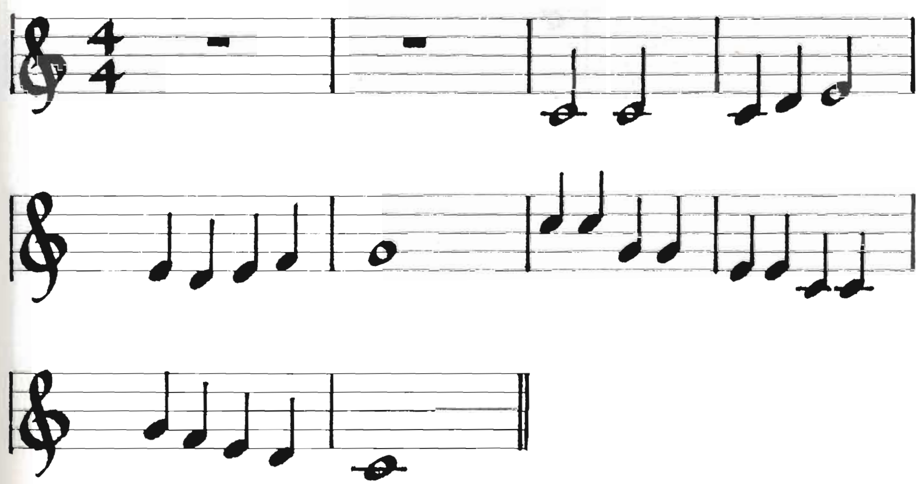



Figure 3.14. Row, Row, Row Your Boat.

Experiment 5: Two-Note Chords with a Recorded Melody

The final experiment in this section will provide an experience in harmonizing a simple melody with two-note chords played on the Odyssey. You will find that, despite the fact the melody is the same in both instances, the interrelation of melody and harmony is such that if the tonality of the harmony is changed, the net effect of the music upon the listener is greatly altered.

If you are using tape equipment, you should first record the melody (upper line) shown in the two examples; this is exactly the same in Version A and in Version B. Then, as the recorded portion is played back, play the harmony part (lower two-note part) indicated in Version A. After you have had a chance to hear how this harmony sounds, play the part indicated in Version B.

VERSION A

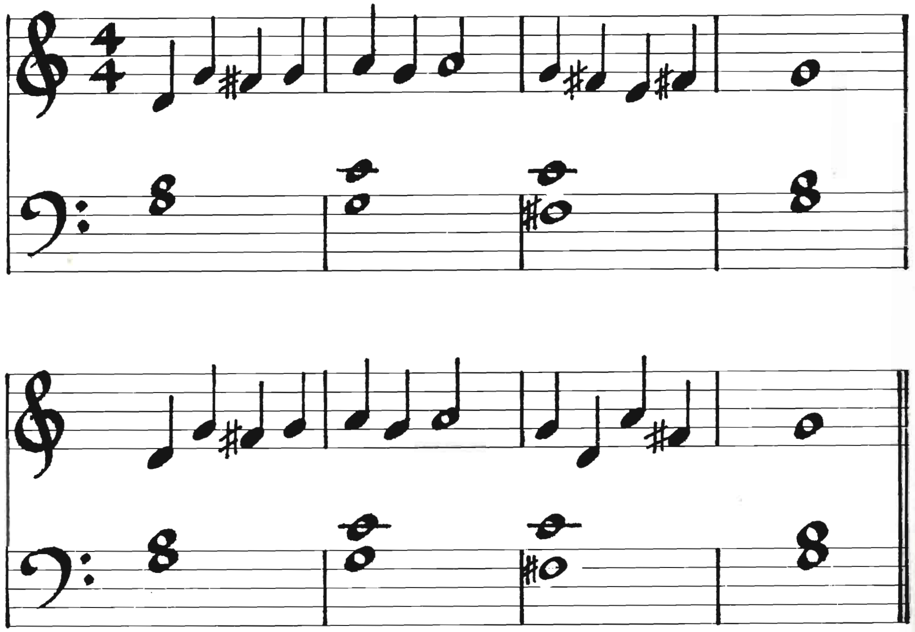



Figure 3.15. Aura Lee.

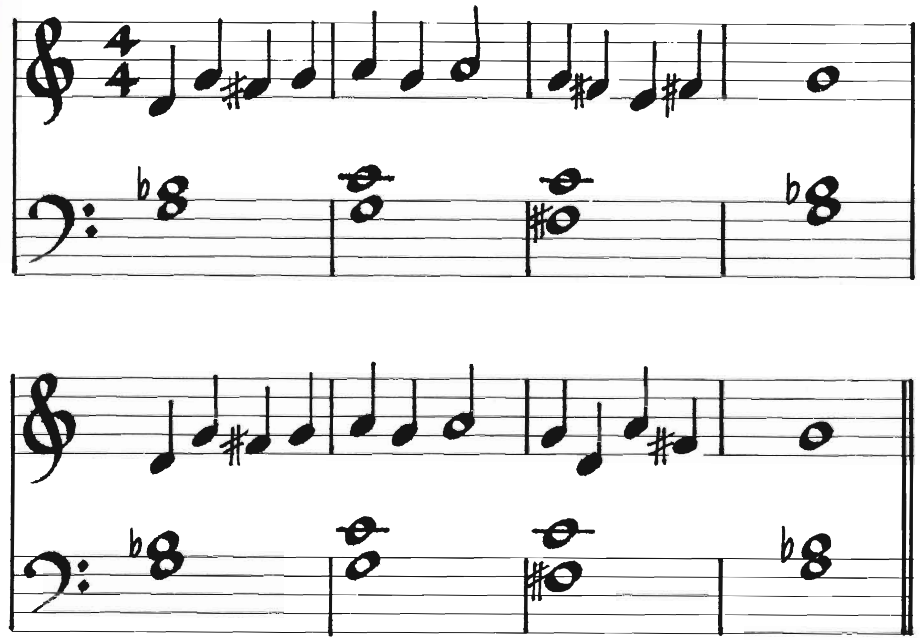



Figure 3.16. Minor-key version of Figure 3.15.

When you compare the two examples, you'll probably discover that the melody takes on a completely different character when the tonality of the harmony is changed. The first example is harmonized in a *major* key; the second is in a *minor* key. The minor chords providing the harmonic background in Version B typically create an effect best described as "dark" in nature, and tending to convey feelings of sadness. Major keys, on the other hand, tend to provide a "lightness" characteristic of Version A. While these two general descriptions are arbitrary, and probably stem from our experience — associating minor keys with funerals and blues — they are set forth here simply to provide an easy "handle" by which to describe the effect that the harmony has upon the impact of the musical whole. Don't hesitate to try reharmonizing other melodies that members of the class know well.

Section 4: Transposition

There are many ways of classifying musical instruments. One such classification, mentioned in Part II, divides them into two general categories: (1) concert-pitch instruments, and (2) transposing instruments. The concert-pitch instruments provide the same sound notated on the music. In other words, a particular pitch/note is produced by a key which has the same name. On the other hand, a transposing instrument is one in which the player reads one note and the instrument sounds another. Therefore, musical notation for a transposing instrument has to be written in another key; its pitch will then match those instruments tuned to A-440, or concert pitch. As you can well understand, this basic difference in the pitch produced by concert-pitch and transposing instruments can and has caused certain problems. For example, if a clarinetist (B-flat instrument) wanted to play a flute part (C instrument), one of two things would be required: (1) transposing the music by rewriting the part, or (2) mentally reading the notes in another key.

With a few exceptions, most brass and woodwind instruments are *not* tuned to concert pitch, while keyboard instruments — piano, organ, accordion, and celeste — *are* tuned to concert pitch. This pitch difference in the two kinds of instruments makes it impossible for them to be played together unless one of the above-mentioned considerations is met. In the normal course of events, the first consideration — written transposition — is automatically met in the scoring of the music by the arranger, thus eliminating this step by the player. The transpositions in the score, however, would only cover those normally needed and still leave the clarinetist in our example relying on personal devices to play the flute part. Keyboard players desiring to play parts written for transposing instruments could also expect to deal with the same problems. As can easily be seen, the interchanging of one instrument for another has important limitations other than just the differences in sound/tone color which two instruments produce.

By now you may have wondered why all instruments are not tuned in the same key — a question that many an individual struggling to transpose a part has thought about. The reason can be explained by very briefly examining the history of the early development of musical instruments.

Originally, the musical staff notation used by an individual to play a particular instrument was determined by the range of the instrument — in this case the lowest note. If the lowest clear tone that an instrument could produce was a B flat — using today's pitch standards — then the instrument was said to be pitched in B flat and became known as a B-flat instrument. These early instruments did not have the benefit of valves and slides and therefore could not duplicate the range of their modern-day equivalents. Although valves and slides have extended the capability of instruments so that they can play effectively in almost any key, transposed notation has survived and become a tradition.

The result of these differences in instrumental pitch is a musical score which by necessity contains various key signatures to accomodate these pitch differences. Examine Figure 4.1 and locate two instruments with unlike key signatures.

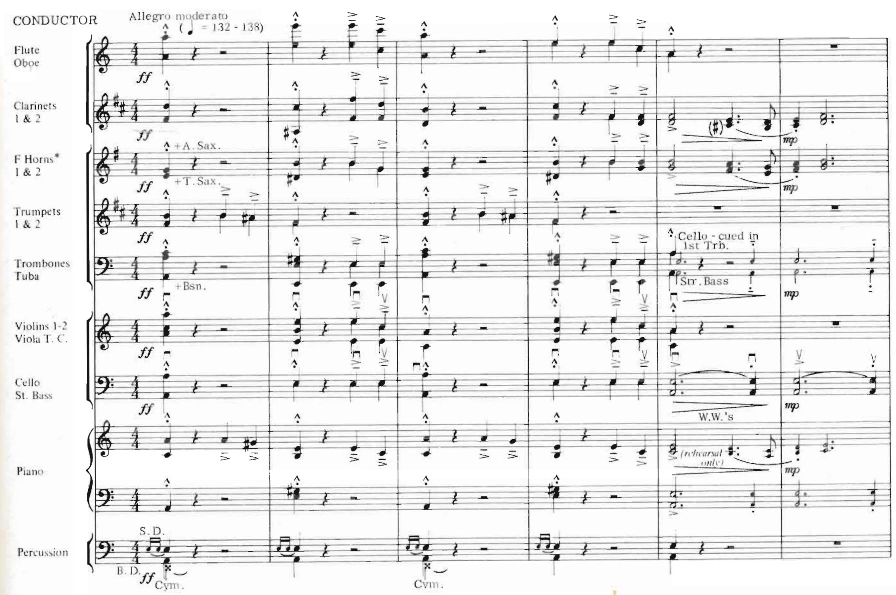



Figure 4.1. Example of an orchestral score.

Trumpet

Find a simple melody and transpose the music for the trumpet. Remember, the notes must be written one whole step higher.

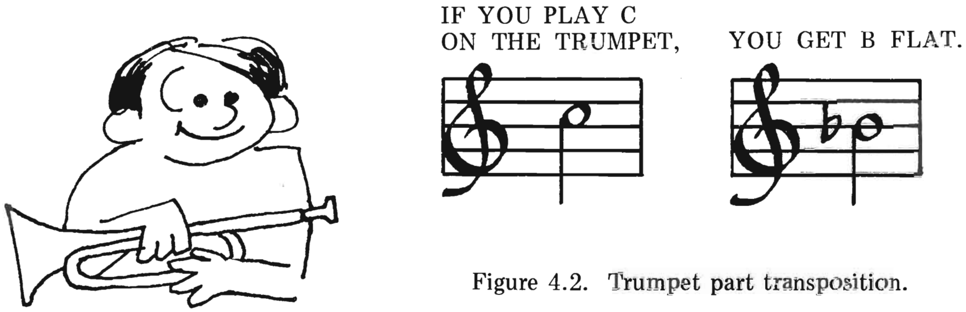


Figure 4.2. Trumpet part transposition.

Alto Saxophone

The alto saxophone is another transposing instrument. Unlike the trumpet, the alto saxophone is an E-flat horn producing a note 4½ steps lower than the written notation (Figure 4.3). Just a moment's glance at Figure 4.1 provides an immediate indication of one area in which a conductor must be extremely proficient — transposition. It's readily apparent that the study of musical scores remains a difficult task for many musicians, the beginners and the experienced alike. To rehearse and conduct an orchestra competently demands a thorough knowledge of transposition on the director's part, and the concepts of transposition continue to challenge most musicians.

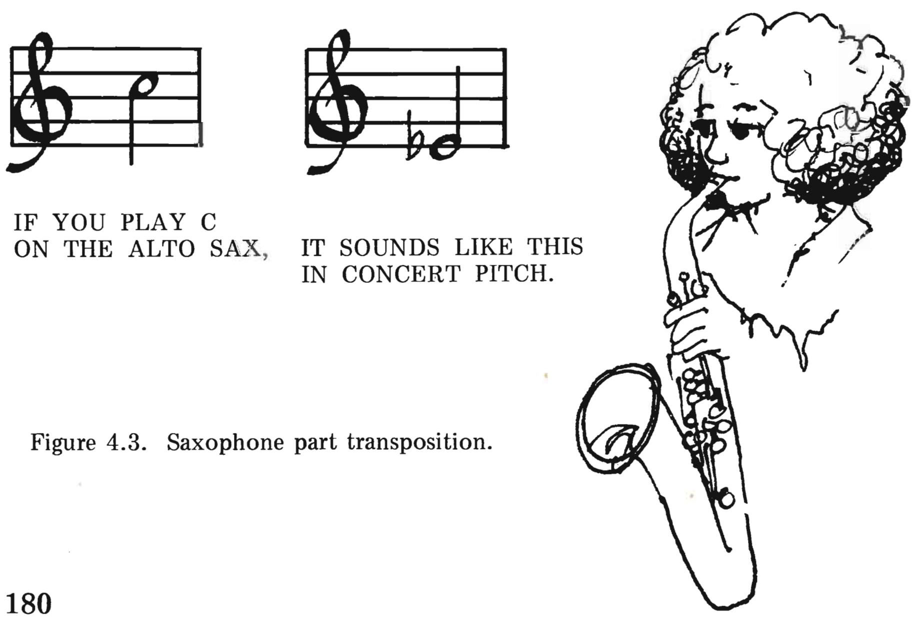


Figure 4.3. Saxophone part transposition.

Transposing and Concert-Pitch Instruments

Now that you have acquired a general idea of transposition and its over-all importance, let's examine some specific instruments in terms of the pitch they produce.

Trumpet

The trumpet depends on transposition to produce notes in the concert key. It is pitched in B flat, which means that the trumpet part must be written one whole step higher to facilitate the correct adjustment in pitch (Figure 4.2). Transpose a simple melody for the alto saxophone. You may find this transposition more difficult due to the wider interval. Check the notes carefully.

Trombone

The trombone is an instrument that does *not* need to be transposed — it produces the notated pitch. Figure 4.4 shows trombone notation.

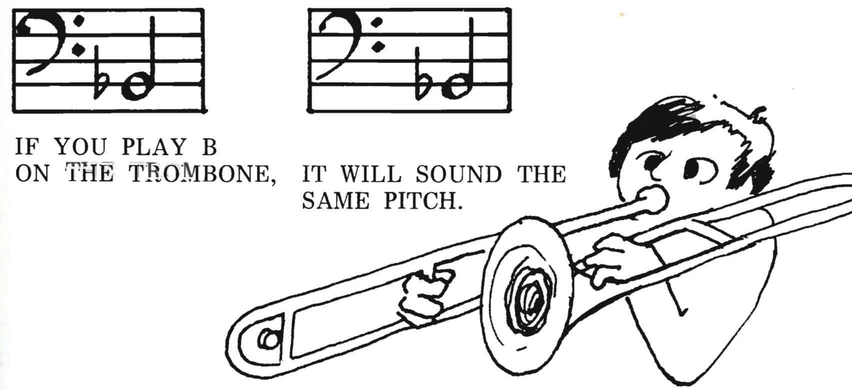


Figure 4.4. Trombone part example.

Flute

Another example of a concert-pitch instrument is the flute. Figure 4.5 illustrates flute notation.

At this juncture you've learned that pitch differences between instruments, which made the concept of transposition necessary, was brought about by the limitations of early instruments and continued because of tradition. Since it doesn't appear that our notation system

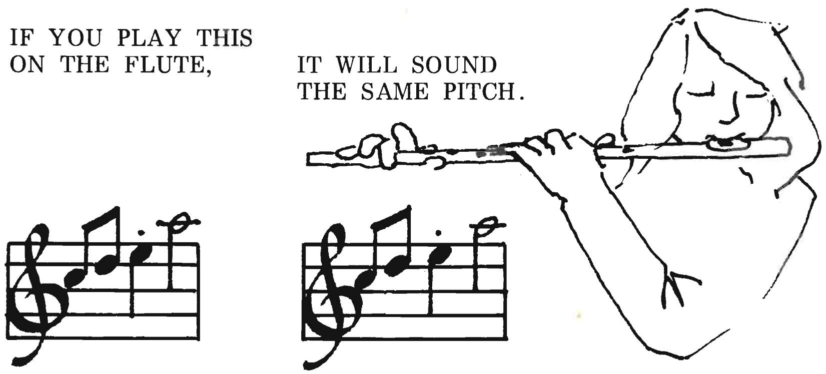


Figure 4.5. Flute part example.

is going to change, pitch alterations by transposing will undoubtedly remain the usual practice. Therefore, all the skills needed to make this system operational will also be required. There is an instrument with the capability of instantly mastering any transpositional situation, however. Imagine playing an instrument which functions in a manner so sophisticated that it immediately allows you to transpose to the key of your choice *without* fulfilling any of the usual steps. This instrument is, of course, the Odyssey (Figure 4.6).

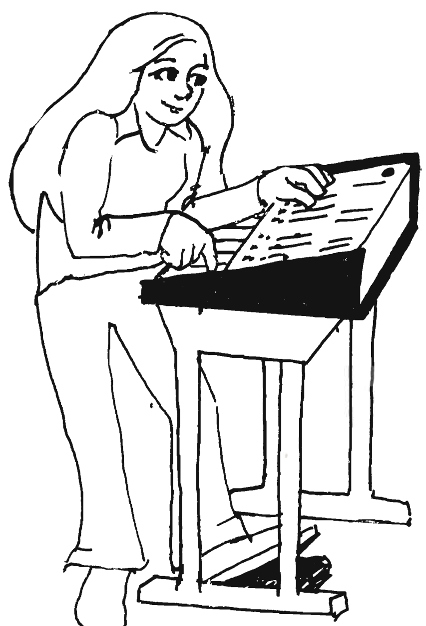


Figure 4.6. The ARP Odyssey.

Now, let's take an example — the clarinet — and perform the necessary steps to allow a band or orchestra member to assume the responsibility for playing a clarinet part on the Odyssey.

The first step is to set the controls to create the sound of the desired instrument. Figure 4.7 shows the patch for the clarinet sound. Second, determine the interval difference between the note read and the note produced. Figure 4.8 follows through with the clarinet example. Use

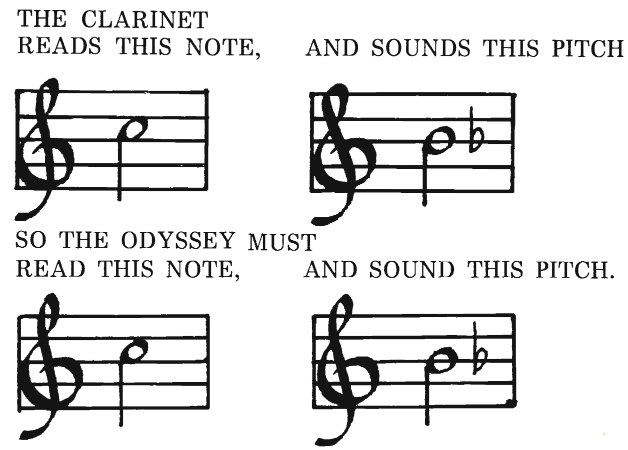


Figure 4.8. Clarinet transposition.

this same format for assessing the pitch differential between other instruments. Initially, if the Odyssey is tuned to concert pitch (A-440), then A played on the Odyssey keyboard will produce the same pitch — A. Figure 4.9 illustrates.

The second step in Figure 4.9 clearly shows that the concert-pitch tuning will not be acceptable. Therefore, move the fine-tuning slider of VCO-2 until the pitch becomes one whole step lower. The Odyssey

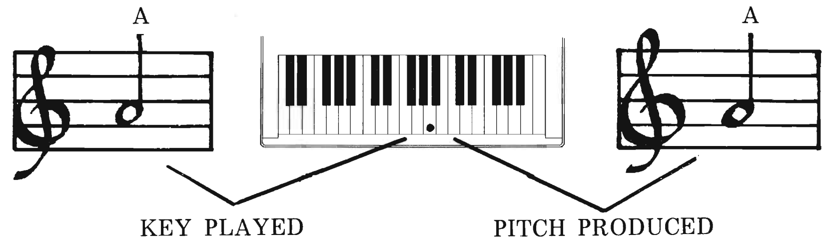


Figure 4.9. Concert-pitch tuning.

is now *transposed* to the Key of B flat — playing C on the keyboard produces a pitch one step lower or B flat. Figure 4.10 shows this relationship. Double check your tuning by trying the following example:

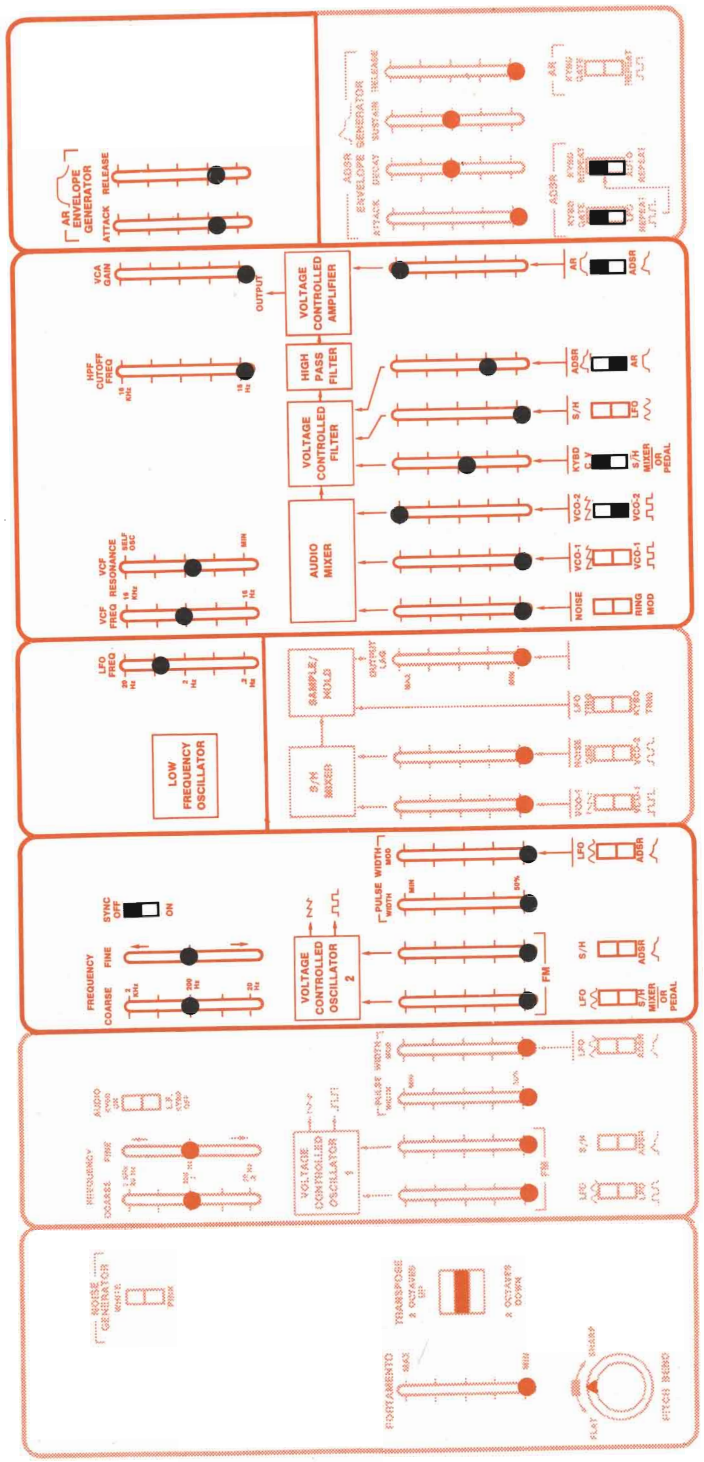
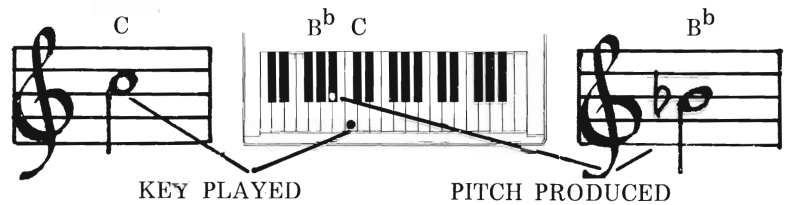


Figure 4.7. Basic clarinet patch.

Play G on the keyboard. What note should this key produce? Figure 4.11 answers the question.



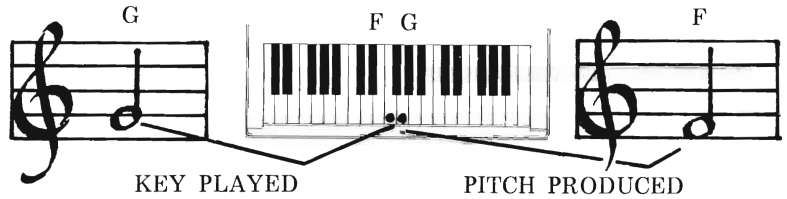


Figure 4.10, Figure 4.11. Transposed tuning.

In addition to adjusting the keyboard pitch to facilitate playing various instrumental parts, another important consideration enters the picture — the range in which an instrument plays. The Odyssey control employed to satisfy the particular range requirements of each instrument — the Transpose switch — is shown in Figure 4.12. To demonstrate the range capability of the Odyssey, first press the circled key shown in Figure 4.13. Now play the key indicated with a triangle two octaves lower. Remember the sound.

Play the first key (circled letter) again. This time move the transpose switch to the "down" position, and play the same key. Did you hear the identical pitch produced by the key labeled with the triangle? Perform a similar experiment, this time starting at the lower end of the keyboard, and produce a note two octaves higher. Exploring the total range possibilities of the Odyssey will yield seven complete octaves — ample flexibility for the creation of almost any instrumental sound.

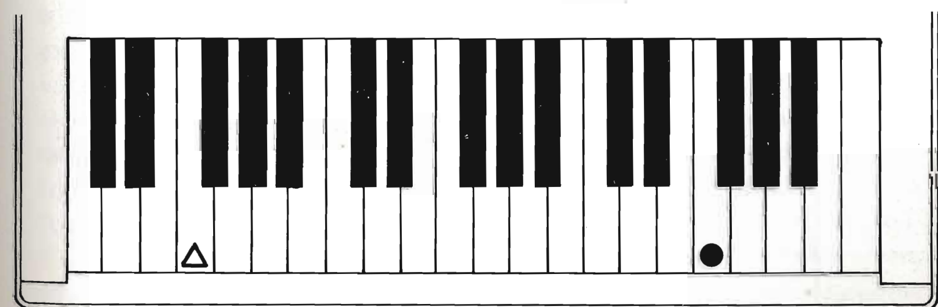


Figure 4.13. Two-octave transposition.

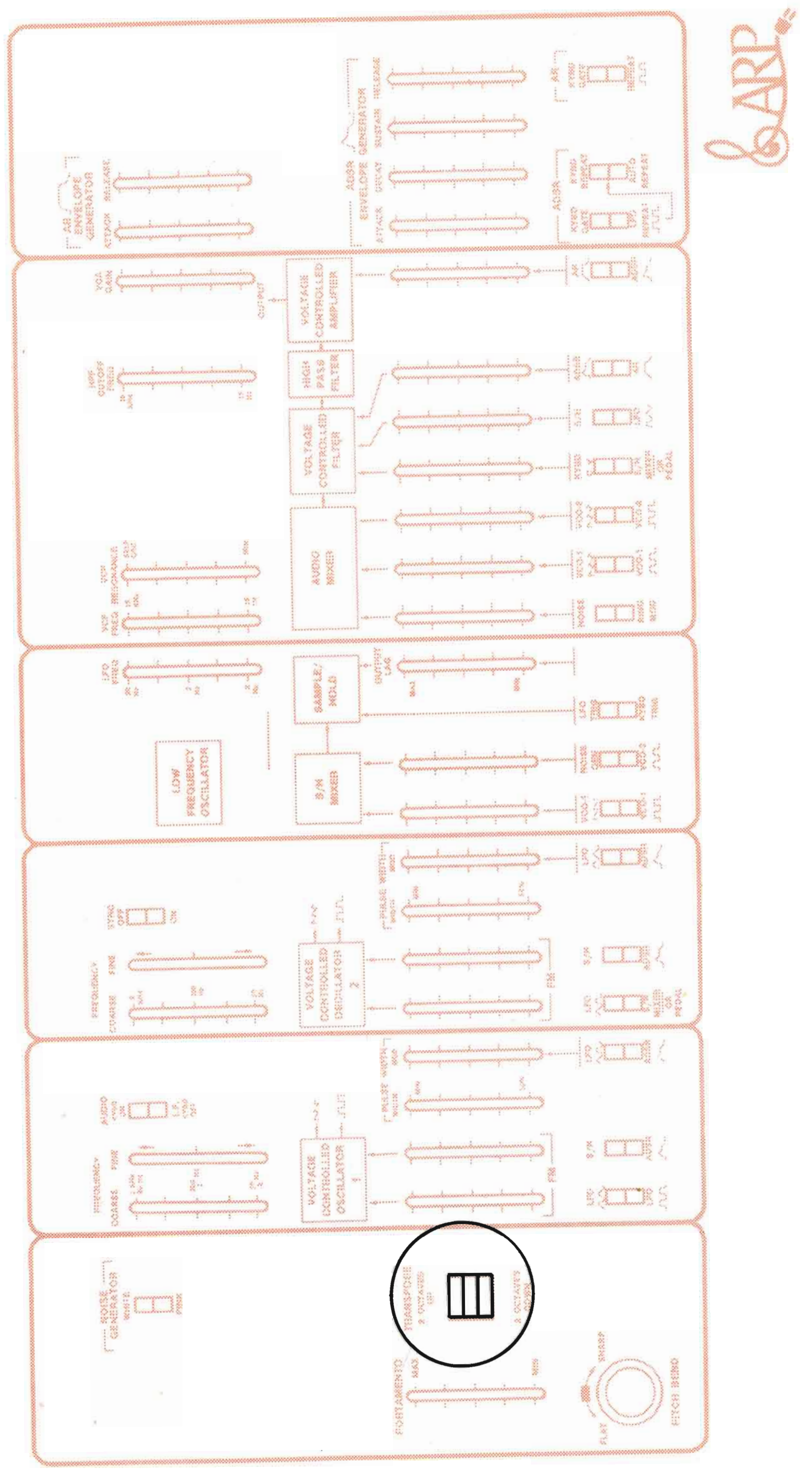


Figure 4.12. Location of Transpose switch.

Tape Music Techniques

The rapid proliferation of electronic music synthesizers has only served to further accelerate the already impressive growth of contemporary recording arts. Electronic music studios have popped up as natural additions to long-established recording studios; new demands, in turn brought about by the increasing number of composers working in this medium, have inspired exciting new recording techniques and equipment. Equally important, these composers have encouraged all of us concerned with, and involved in, the making of music to look at tape recording in a new light. The situation where the performer goes into a studio to record the work written and orchestrated by the composer has changed; instead, it is likely to be the composer who goes into the studio and performs his own work, meticulously building layer upon layer of sound until, at the end of the session, the composer-performer is able to walk out of the studio with his composition-performance on a reel of tape in his hand. Composition and performance have been fused into one act, through the use of electronic music synthesizers and sophisticated recording techniques.

The reasons for the emergence of the composer-performer are probably as numerous as the motivations which cause men to write music. Certainly, the old dichotomy of the composer's intentions versus the performer's interpretation is neatly resolved by this method. More significant, however, is the fact that the composer is no longer concerned with just melody, harmony, and rhythm; a fourth element has been added — timbre. Electronic music synthesizers have opened the door to infinitely broad timbral resources, resources which contemporary composers are seeking to exploit to the fullest. No longer is the composer bound by the timbres which could be produced by conventional musical instruments. With that freedom, however, has come the concomitant problem — how does one notate the subtle changes in timbre, much less the all-encompassing audio fabric, that is woven from single strands of synthesized sound? Music notation, which has changed in the past to accommodate new forms of musical expression, undoubtedly will change in the future to include the advances we view as radical today. Nevertheless, until that time comes, the composer must almost inevitably become the performer, and not surprisingly, many are going to be reluctant to relinquish this newfound role when the moment does arrive. Therefore, it seems logical to expect that fusion of the roles of composer and performer is likely to herald a new era of musical endeavor that will not be totally reversed, despite the ultimate development of adequate notation systems.

It is for this reason that throughout this text, mention is made of the use of tape equipment. Not only is the joint role of composer-performer a pleasurable one, it is an almost essential one — particularly if you are to preserve the musical expressions you create when using the Odyssey. Therefore, it is suggested that whenever possible, the electronic music studio that is developed around the synthesizer should include at least one good-quality tape recorder. If your school already has such a machine, this section of the text will provide suggestions as to how you can make the most of it; if you do not presently have access to a recorder, use the guidelines in this section as the basis for making a decision when the opportunity to acquire such a machine arises.

Section 5: Setting up an Electronic Music Studio

An electronic music studio need not require thousands of dollars worth of expensive electronic equipment. Indeed, the ARP Odyssey will provide the basis for a sound, well-equipped studio, and all other equipment can be added as the budget permits. Most studios are begun in whatever room is available — a portion of the music room, a practice room, or an unused office. The following basic list of equipment may be helpful.

Electrical Outlets. All that is necessary here is to make sure that the number of outlets is sufficient to handle the synthesizer and whatever auxilary equipment is to be used — recorders, amplifiers, and so forth. Outlets should be of the three-pronged type, to insure that each electrical unit is adequately grounded.

Tables. The studio should accommodate a sufficient number of tables for the synthesizer(s), recorder(s), and other equipment that will be used. Naturally, all tables should be as sturdy as possible to prevent damage to any equipment. Shelving for storing tapes, etc., may also be installed after the tables have been placed.

Bulletin Board. A bulletin board will be useful in maintaining a schedule for use of the studio and/or the equipment contained therein. Also listed on the board should be any checklists and procedures for equipment operation. One word of caution, however: do not use chalkboards in the room; the chalk dust can cause expensive problems if allowed to accumulate on recording equipment. Use a board with pushpins for attaching notes and checklists, or secure one of the new types of message boards that utilize a wax crayon on a plastic finish. These will wipe clean with no dust.

Tape Recorders. While not essential, a recorder will add greatly to the enjoyment and utilization of the Odyssey. More detail will be provided as to the most useful type of recorder later in this section.

An Amplifier/Speaker System. The Odyssey contains no speakers of its own, but is fully compatible with almost any equipment that is available. Frequently, a portable public address system used for assemblies can be used; a small electric guitar amplifier can also be employed. Many schools also own either electronic pianos or an electronic organ into which the Odyssey can be connected. You may also be able to connect the Odyssey to the external input of a recorder or a record player (assuming that the recorder or phonograph contains speakers of its own). Figure 5.1 shows a number of simple connections; depending upon what type of equipment is being paired to the Odyssey, a notation has been made on the back panel of the Odyssey as to which output to use — the high-level or the low-level output. Specific applications and special tape techniques may necessitate altering these basic connections.

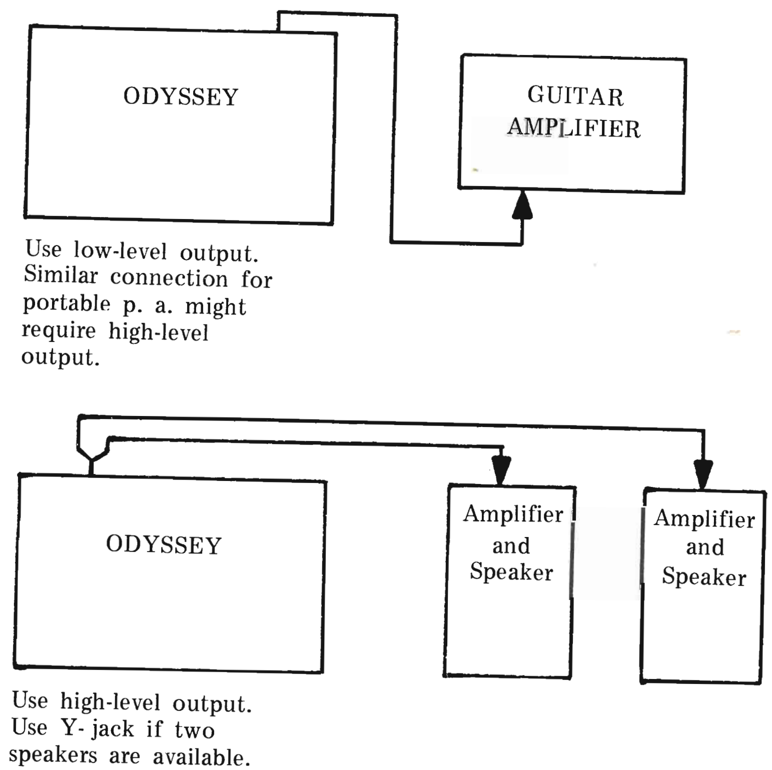
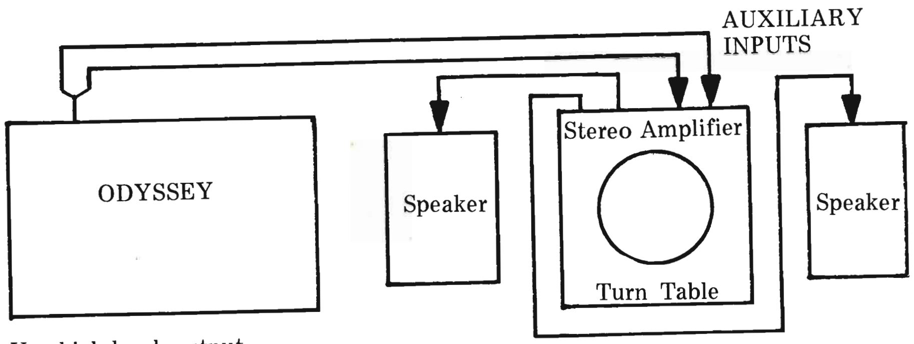
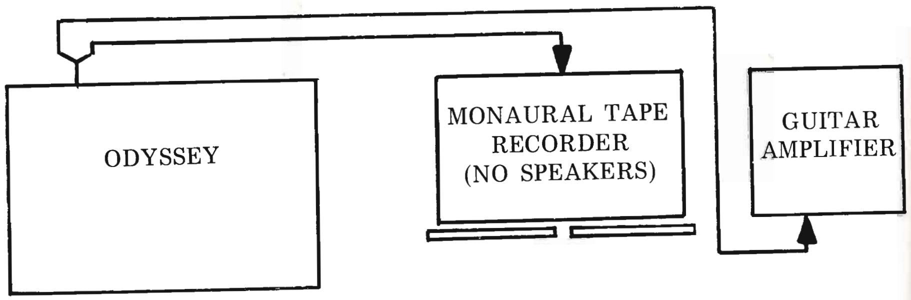


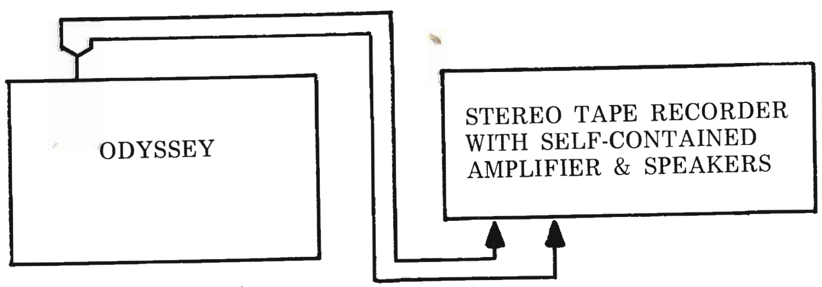
Figure 5.1. Five possible connections for use with the Odyssey.



Use high-level output.
Use Y- jack to provide
signal to both channels.



Use low-level output
to amplifier. Use high-
level direct to tape
recorder, with no
microphone.



Use high-level output
with Y- jack to provide
sound for both channels.

Figure 5.1, continued. Five possible connections.

In the event that you will be using a tape recorder, the following supplies should also be available.

Tape. Fresh, unused tape should always be available for serious studio work; practice and experimental work can be performed on tape that has been used and erased. Avoid extra-long reels of tape; the longer the tape that is wound upon a reel, the thinner it must be to get the extra playing time on that same reel. Use 1½ mil tape; it will be easier to handle and less prone to stretching and accidental breaks.

Splicer. The use of a tape splicer is both easier, and safer, than the use of a razor blade. Do not use magnetized scissors to cut tape; not only will they do a sloppy job but the magnetized blades will create a "pop" on the tape.

Cleaning: Recorders need periodic care in order to operate at their best. However, the cleaning and demagnetizing of recorder heads should be performed only by those persons with appropriate training and experience.

The one item not yet mentioned, of course, is the synthesizer itself. The ARP Odyssey is just one of a full line of electronic music synthesizers made available by ARP. All ARP equipment is compatible, and the studio that begins with an Odyssey can later add its big brother, the ARP 2600, or, bigger yet, the ARP 2500 modular system. Product information on each of these instruments is available directly from ARP Instruments, Inc., Newton, Massachusetts.

Using the Odyssey with a Recorder

As already mentioned, use of a tape recorder will greatly enhance the value and enjoyment of your experience with electronic music. Even the simplest monaural recorder offers at least the opportunity to record one part and then play it back while performing a second part on the Odyssey. The following information details how you can get the maximum use from whatever equipment is available, from a monaural machine up to and including a four-channel recorder.

Monaural Machines.

With a mono recorder, the possibilities are somewhat limited; for this reason, if you are about to acquire a new machine, it is suggested that something with greater flexibility be considered. Nevertheless, a monaural recorder can be used to create a basic library of taped sounds that will provide an audible record of experiments and patches, as well as the beginning of a sounds-and-effects library that may be used to contribute to the soundtrack accompanying school drama events.

With a mono machine, you have two options: first, you can record and simply listen; this would include recording and storing those tapes of the best patches, effects, and experiments. Second, you can record one part, a melody, for example, and then play the recording while adding a second part, perhaps a counter-melody. Use of a Y-jack on the input of the machine will permit you to record both the Odyssey and the signal from a microphone, phonograph, or another recorder. In the end, however, you will find yourself constantly improvising connections in an effort to make the machine perform functions for which it was not designed.

Stereo Tape Recorders

The availability of a stereo recorder is a big plus for an electronic music studio. While not the ultimate machine, it is easily within the reach of most budgets, and a variety of good recorders are readily available. Most stereo machines have a "sound-on-sound" provision that will allow you to record one channel, then play back that channel while at the same time recording a second part — in effect, adding a second sound upon the first sound. This provision will open the door to multitracking, in a basic sort of way, permitting actual composition using the Odyssey and the recorder.

Most recorders of this type will also permit you to go beyond just two tracks by using a "bounce" technique. As the name implies, you record on one channel and then bounce the signal from that channel, along with a new part, onto the other channel. You can then bounce the net result, along with yet another part performed on the Odyssey, back onto the original channel. Figure 5.2 illustrates.

Part Being Performed	Recorded on Channel
No. 1 — Bass Part	A
No. 2 — Rhythmic Effect, plus recorded bass from Channel A	B
No. 3 — Chord Accompaniment, plus recorded bass and rhythm from Channel B	A

Figure 5.2. Diagram of "bounce" technique.

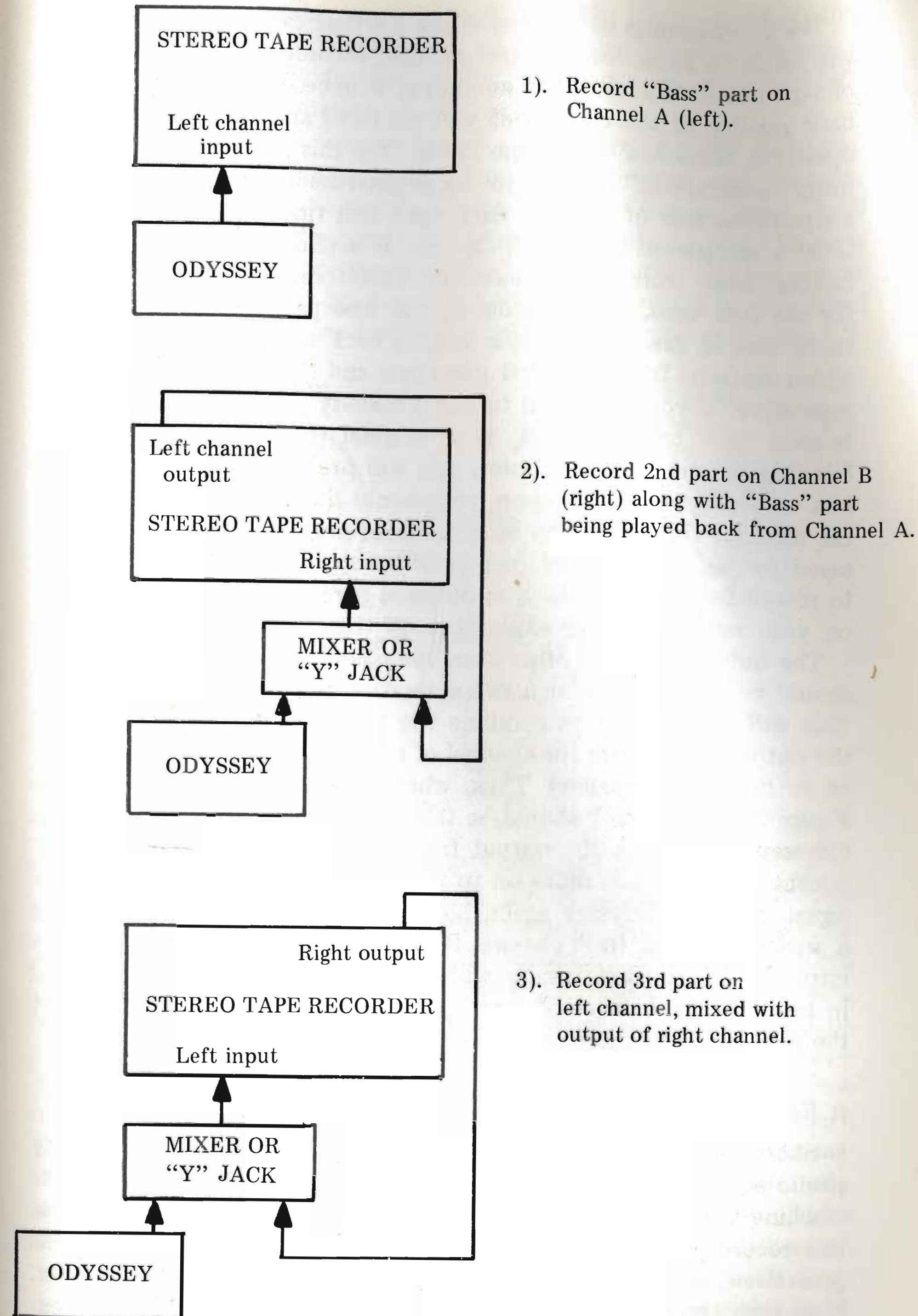


Figure 5.2, continued. Connections for "bouncing" between tracks.

The shortcomings of this procedure are that each successive bounce will result in some loss of signal quality, accompanied by an increase of tape noise. As a result, the number of bounces will be limited by the basic quality of the equipment you are using and the cleanliness and operating condition of its tape heads. Try this technique to see how many "generations" away from the original track you can go without a significant loss of fidelity. Each time that first bit of recorded material is transferred to another channel, it is said to be one generation further away from first generation. Therefore, when it is bounced for the first time, onto channel B, the bass part in Figure 5.2 is already second generation. When it goes back to channel A, with the added material from recorded parts two and three, it is into its third generation. If you were then to add a melody part, through one more bounce back to channel B, your original track would be into its fourth generation. At that point, you will probably notice an obvious loss of quality. For this reason, you should always record the less critical material first, with the most delicate or exacting material to be saved for last. If you find that your stereo recorder will permit you to record four separate lines, as outlined here, you will indeed be well on your way toward the exploration of the fun of tape composition.

The only equipment other than the recorder and the Odyssey that should be required for such an exploration is a Y-jack or patch cord. This will permit you to combine the signal from the Odyssey with the output signal from the channel of the recorder that is being bounced to the second channel. Thus, when performing part number 2 in Figure 5.2, the Y-jack should be taking the rhythmic effect from the Odyssey, along with the output from channel A, and feeding it into channel B. When you move on to part number 3, you'll be taking the signal from the Odyssey again, but this time you will be combining it with the output from channel B and feeding the Y-combined signals into channel A. Best results will require some practice, particularly in terms of the balance of the volume levels of the various parts, but the results will be more than worthwhile.

A Four-Channel Recorder

Short of acquiring prohibitively expensive, professional-quality studio equipment, a good-quality four-channel recorder is the ultimate machine for a basic electronic music studio. As the name implies, this recorder permits you to record four separate channels, all first generation, resulting in high-quality, multiple-track tape composition. Moreover, the more advanced machines, such as the TEAC 3340, enable you to go beyond four channels to at least seven tracks, none of

which will be more than second generation. The key to the flexibility of such machines is a "record head" that can also be used for monitoring. Before going any further, let's define what a "head" is and look at the normal configuration of tape heads.

The tape heads are simply the parts of the machine that make contact with the recording tape, encoding (recording) and decoding (playback) the magnetic signal. Normally, there is also an "erase" head that precedes both the record and playback heads, in order to erase unwanted signals before the record head applies new magnetic signals to the tape. Looking head-on at a typical machine, the uncovered heads might be positioned something like the diagram in Figure 5.3.

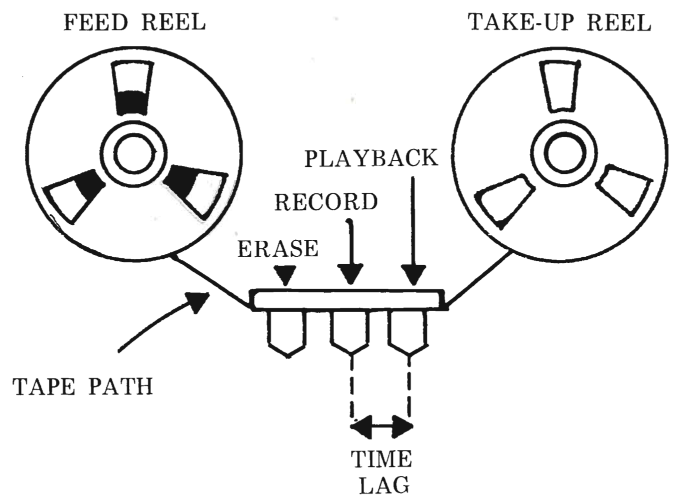


Figure 5.3. Standard tape recorder.

On the less sophisticated recorders, the record head is not capable of monitoring tracks already recorded while at the same time recording those tracks still open. Instead, the recorded tracks are monitored by the playback head, resulting in a time lag between the signal being recorded on the second track(s) and the signal being monitored on the first track(s). Remember, the tape is moving across the heads at a particular speed measured in inches per second; the slower the tape moves, the more time that distance between the record and playback heads represents. At 3¾ ips, which is a common speed on stereo machines, the time lag between heads can be as much as a quarter of a second. Imagine being a quarter of a second out of meter when playing with a group; if you can mentally hear the rhythmic "fight" that would result, you can begin to imagine the effect that would result from trying to record a new track in perfect synchronization with a track you're hearing a quarter of a second later than you should.

For this reason, most modern quarter-track machines have the synchronization feature mentioned earlier: a record head that can also be used for monitoring, thus eliminating the time lag between heads. All legitimate four-channel machines also have all four channels running in the same direction. Most stereo machines actually record four separate channels, but can only record or play back two at a time. Such machines are called "quarter-track stereo" recorders. When the reels are switched and the tape is reversed on such machines, stereo recording continues in the other direction. This is potentially useful for getting more stereo material on a single reel of tape, but is not particularly advantageous in an electronic music studio (Figure 5.4).

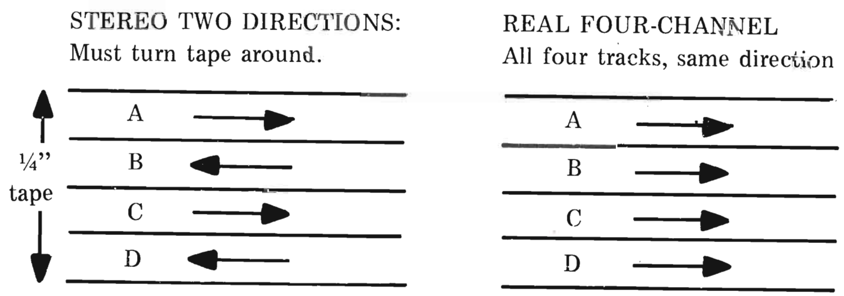


Figure 5.4. Comparison of two types of four-track recorders.

A four-channel, synchronized machine, then, permits the recording of four individual tracks with ease. One simply records track 1, then monitors 1 while recording 2, monitors 1 and 2 while recording 3, and finally, monitors 1, 2, and 3 while recording 4. See Figure 5.5.

Musical Track	Channel
1	A
2	B
3	C
4	D

Figure 5.5. Four-track recording.

It is also possible, however, to get six or seven tracks, with no track being more than second generation (Figure 5.6).

Musical Track	Channel
1	A
2	B
3	C
Mix A+B+C	D = 1+2+3
4	A = 4
5	B = 5
6	C = 6

All 6 tracks

Figure 5.6. Procedure for six-track recording.

To produce six tracks (three first generation and three second generation), simply record tracks 1, 2, and 3, then mix and record them on D. This frees channels A, B, and C for musical tracks 4, 5, and 6.

To achieve seven tracks, follow the procedure shown in Figure 5.7.

Musical Track	Channel
1	A
2	B
3	C
A+B+C	D = 1+2+3
4	A
5	B
A+B	C = 4+5
6	A = 6
7	B = 7

All 7 tracks

Figure 5.7. Seven-track recording.

Now your four-channel tape contains seven separate musical tracks:

A = 6 B = 7 C = 4+5 D = 1+2+3	Mix = 1+2+3+4+5+6+7

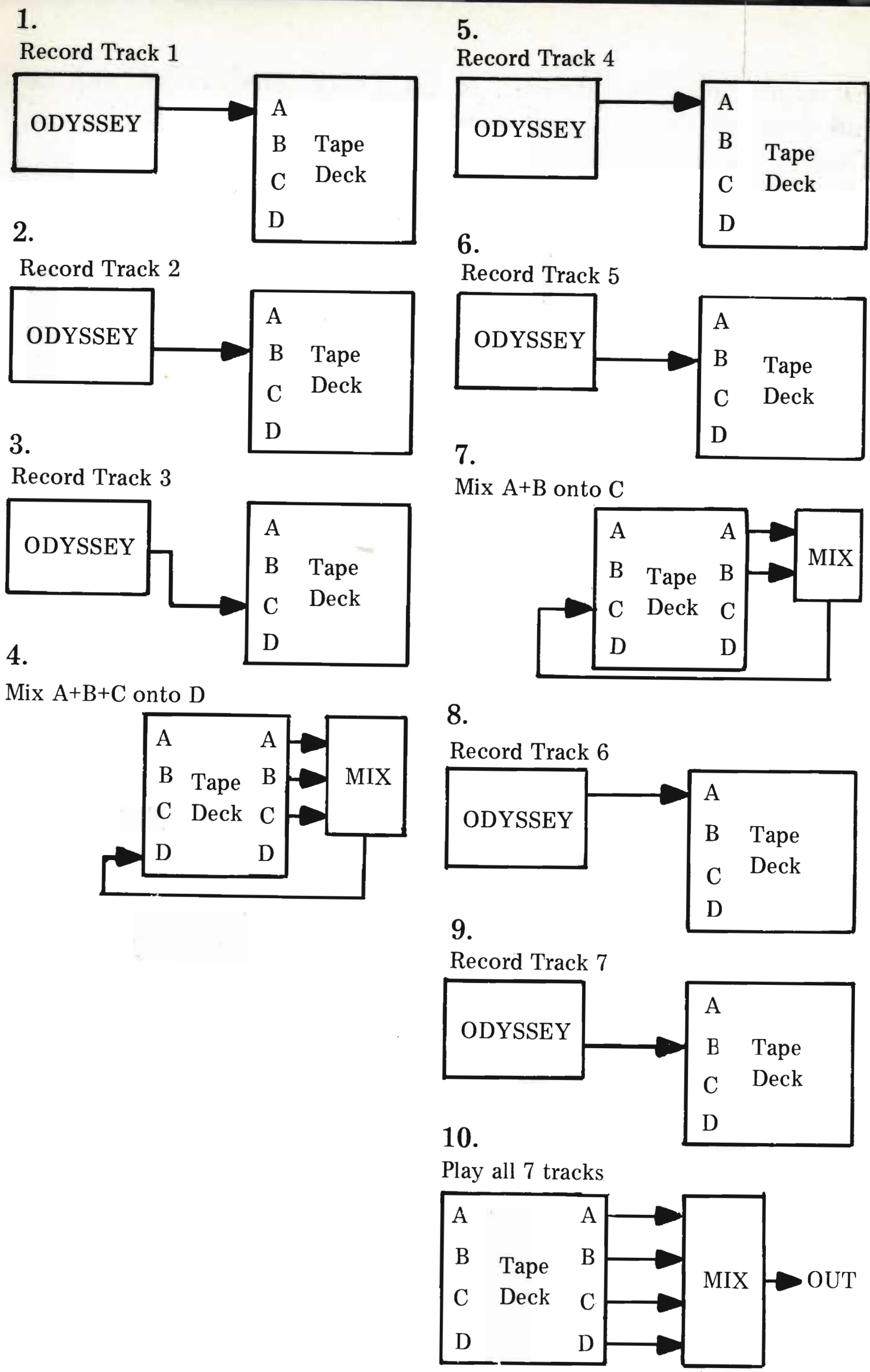


Figure 5.7, continued. Connections for 7-track recording.

The result shown in Figure 5.7 is seven separate tracks, five second generation and two first generation, resulting in extremely high musical quality and ample possibilities for exploration of tape composition.

Our constant concern for the quality of the recorded signal is reflected in the development of eight-, sixteen-, and twenty-four-track professional machines which are now in use in studios around the world. These machines, such as the one pictured in the studio layout shown in Figure 5.8, offer as many as twenty-four individual tracks and provide virtually unlimited multitracking potential without going beyond first generation on any track. This is particularly useful in recording large orchestras, since it is not necessary to devote a separate channel to each instrument; rather, sections of instruments may be covered with one or two microphones, while other difficult-to-record instruments (such as a drum set) get one or more mikes of their own.

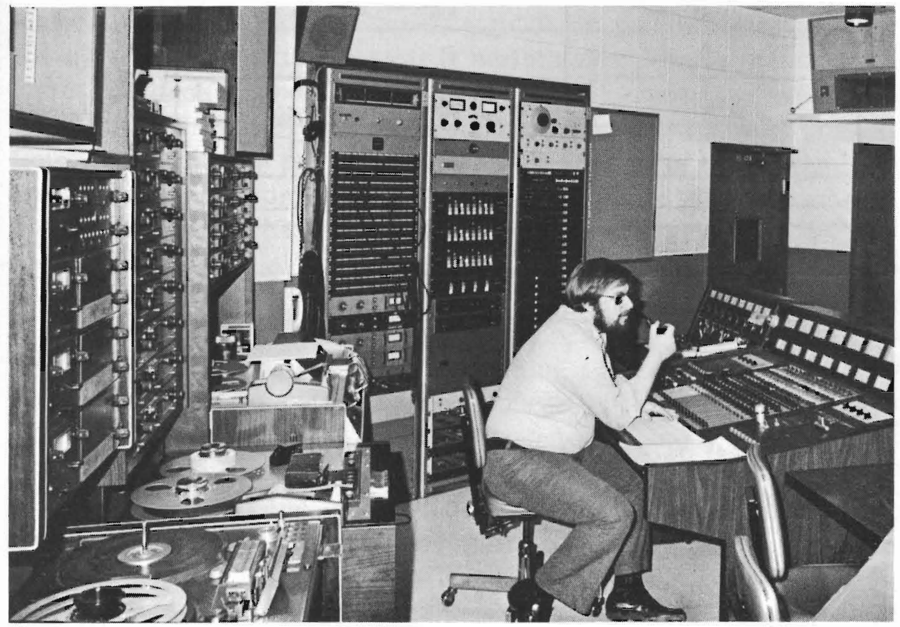


Figure 5.8.

If your electronic music studio includes a four-channel tape machine, you'll find that its seven-track potential permits all but the most complex tape compositions to be recorded. In the event you need more than seven tracks, you may get up to eleven tracks on a four-channel recording by bouncing from track to track as shown in Figure 5.9.

Musical Track	Channel	
1	A	
2	B	
3	C	
A+B+C	D = 1+2+3	
4	A	
5	B	
A+B	C = 4+5	
6	B	
B+C+D	A = 1+2+3+4+5+6	
7	B	
8	C	
B+C	D = 7+8	1 - 11
9	C	
C+D	B = 7+8+9	
10	C	
11	D	

Musical tracks 1, 2, 3, 4, 5, 7, 8 will be 3rd generation.
Musical tracks 6, 9 will be 2nd generation.
Musical tracks 10,11 will be 1st generation.

Figure 5.9. Eleven-track recording.

As in the case of the stereo machine, be certain to record less critical parts first. As you mix down a number of tracks onto a single channel, be careful to watch the volume levels of the individual tracks to be sure that you are getting the balance you want between the parts. Experimentation and lots of practice will prove to be your best guide.

Tape Editing

Up to this point, this section has been primarily concerned with the kind of equipment that you may have available and how to maximize its potential in terms of the number of individual tracks that you may record. We have spent a considerable amount of time on this subject because it so directly affects the potential complexity of the tape compositions that you will be able to create. Nevertheless, there are a number of subjects that will prove to be of equal importance as you begin incorporating a recorder into the activities of the electronic music studio. One such subject is tape editing.

Virtually every contemporary recording you hear is not the result of a single continuous "take," but rather is the best parts of several tries, or takes. The reason that finished recordings are assembled from the best pieces of several takes is largely because of the convenience to the performer and the lower cost in time and labor. While a single performer might eventually record an entire piece without making a single mistake, even that is difficult; you can imagine, then, the likelihood of getting a perfect performance from a group in one take. The likelihood diminishes rapidly as the number of players in the group goes up. Therefore, in order to produce the best possible finished recording, a number of *intercuts* — splices of tape taken from several different recorded takes — may ultimately be joined to produce the finished piece. You can use this technique as effectively as professional recording engineers; it simply requires a bit of basic information about splicing, accompanied by a fair amount of practice.

As is implied by the foregoing discussion, splicing is the art of cutting the recording tape and then rejoining two separate pieces. The cutting should be done with a tape splicer. Such a unit will provide the tape cut at the correct angle, illustrated in Figure 5.10. The splice should always be made diagonally in order to avoid getting a "pop" in the recording at the point where the splice is made. In addition, you should be sure to use actual splicing tape — not just any kind of transparent tape. Not only will most transparent tape eventually slip and/or stretch, but the adhesive backing will sometimes ooze out and come into contact with delicate parts such as the tape heads.

Where to splice is the part that will require practice. A good tape editor is an artist — an artist who is able to catch a singer between breaths, or a rock group between beats — and when the two segments are spliced together, you cannot tell that the material was not done in one take. To find the correct spot to splice, listen as the tape runs until it comes to the part you want to cut out. As the tape reaches that portion of the take, press the "pause" or "edit" button of your machine. This should leave the heads in the playback configuration, permitting you to turn the reels back and forth by hand until you are able to determine accurately the precise spot where you want to make the first cut. Repeat the procedure to find the spot for the second cut. This process will be made easier by removing the protective cover over the heads. In this manner, you will be able to actually see the playback head, and determine more accurately where the splice should be made.

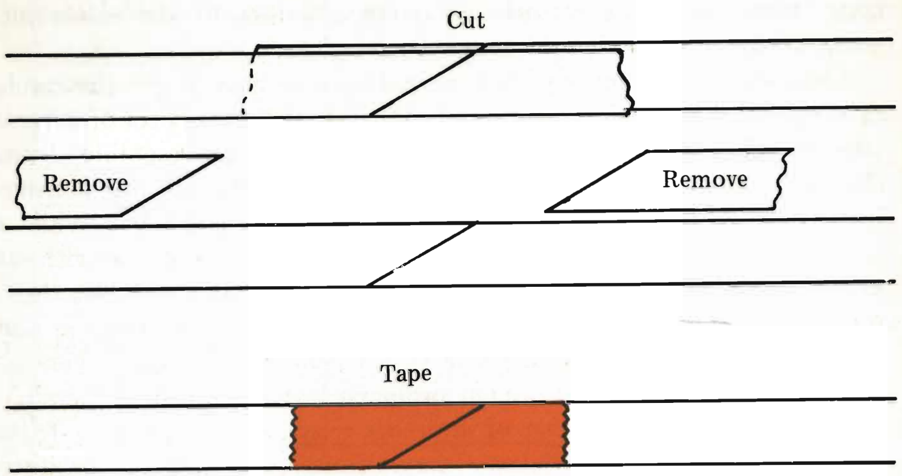


Figure 5.10. Correct tape-splice angle.

When you have located the spots, mark them on the tape with a wax pencil or crayon. Be careful not to get wax on the heads! Then gently pull the tape away from the heads, so that you have a semicircular loop including both marks between your right and your left hands. Put this loop in the splicer, positioning the first mark so that the first cut will be made at that point. Do the same for the second. After making the second cut *but before moving* either end of the recording tape, secure your splice with the special splicing tape. Assuming that all of the foregoing proceeds smoothly, you should then move the reels a sufficient amount to take up the slack in the tape and then rewind to a point just before the splice. This can usually be done by hand-turning the feed reel in a counterclockwise direction. Then disengage the pause control and listen to your splice. If it's a good one, you won't hear any superfluous noise, nor will the rhythm skip a beat or the singer miss a word. It should sound as if the original performance had continued perfectly, without interruption.

As you can imagine, this is not so easy at first, but it is not really that difficult and is extremely rewarding once mastered. One obvious hint: such splices will be easier to make if the material is recorded at the *highest* possible tape speed. Whereas a recording made at 1 7/8 ips would require extremely careful editing, the tape only having moved less than two inches in one second, a recording made at 7½ ips will allow some latitude for less-than-perfect cuts — four times the amount of tape will have passed over the record head in the same period of time. For this reason, most professional studios record at no less than 15 ips, and never at less than 7½. These speeds also have the residual effect (particularly on less expensive machines) of eliminating the unintended waver, or vibratolike effect, created by a variance in the tape speed as it moves across the heads.

Experiment 1: Editing Techniques

Before going on, practice making an intercut. This experiment can be performed with either a monaural machine or a four-channel. Record a simple melody, such as "Row, Row Your Boat" or "Merrily We Roll Along." Play the verse three times through. Then, to practice editing, go back and cut out the second verse, joining the first to the third. To do this effectively, you will now have to be concerned not only with editing techniques but also with the recording considerations involved. The third verse must be at the same *tempo* as the first; if you have not maintained the same tempo, the chances are that the difference between verses 1 and 3, when verse 2 is edited out, will be immediately noticeable. You can see that this is a particular problem when splicing material together from several different takes which may not have been made on the same day.

You should also be careful to watch the *volume level* when recording. If verse 3 is significantly louder than verse 1, the abrupt change in volume where they are spliced together provides a sure tip to the listener that a splice has been made. Again, this concern is even more critical if you will be splicing pieces of cuts made over a series of days, or even weeks. Watch the level meters (V.U. meters) of your recorder carefully; try to maintain a consistent practice when recording so that the general levels of any two sessions will always work out to be pretty much the same.

Finally, you must be attentive to the general *acoustical properties* of any two cuts that you are splicing together. The chances are that, in this experiment, verses 1 and 3 will not differ significantly — unless you deliberately add reverb or change the basic timbre of the sound during verse two. When editing takes from several sessions, however, you will not enjoy this luxury; just as you cannot splice together one cut made in your living room with a second cut made in your bathroom, you cannot splice together two different cuts exhibiting different amounts of electronic reverberation. It would be as obvious as splicing the "live," reverberation-saturated cut made in a bathroom together with the muffled, deadened cut recorded in a heavily carpeted, well-draped living room.

Tape Manipulation Techniques

Thus far, we have dealt with fairly common, though extremely useful, recording techniques. As a conclusion to this section of the text, a number of more exotic tape techniques will be presented. While the Odyssey alone permits almost total control of every parameter of sound, the use of a tape recorder will extend your control just that much further, expanding your creative and technical facilities.

One such technique is the relatively simple process of recording a track at one speed for playback or mixdown at another speed. This technique is most useful when you are recording a musical passage that is difficult for you to play at the speed required. To achieve the effect that you want, simply play the passage one octave lower than written and at half the speed you are ultimately seeking. If you record this octave-lower/half-speed passage at 3¾, by playing it back at 7½ you will exactly double the speed and raise the pitch by one octave — producing exactly the effect you sought. Similarly, the same effect would be created by recording it at 7½ and playing it back at 15 ips. While it is less useful musically, you can reverse the procedure to create a number of interesting effects. A cymbal crash at half-speed, for example, is an effect that can be incorporated into certain electronic compositions.

The same basic technique can be used to record and modify natural sounds, such as voices, footsteps, or running water. The act of recording, modifying, and assembling such sounds is the basis for *musique concrête* (classical electronic music made by splicing together unrelated phrases, notes, and sounds to achieve new effects). You should not ignore the possibility of mixing sounds of this nature with the electronic effects produced by the synthesizer in order to assemble a compositional whole.

A second technique you may wish to try is the creation of a tape loop. To do this, set the Odyssey controls as shown in the patch that appears in Figure 5.11. This patch will provide a random, percussive effect. Record two or three minutes of this effect.

After you have a length of tape containing nothing but the percussive sounds, cut a portion of this tape (about thirty inches long) from the center of the tape. You do not necessarily have to be concerned about where the cut made begins and ends. After removing this section of tape, splice its two ends together to form a continuous loop. If possible, you should now turn your tape machine on its side so that gravity will tend to hold the loop in position as it is played. Be certain that the dull side of the tape is touching the heads of the recorder, since this is the side that the sound is recorded upon. Let the tape hang over the edge of the table, as shown in Figure 5.12.

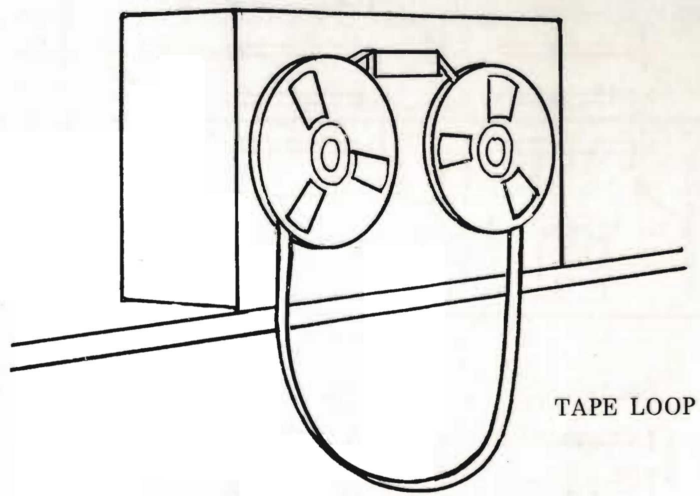


Figure 5.12. Tape loop in recorder.

The repeating pattern that is created by the loop will often be one of sufficient interest that it can be used as a background accompaniment to an electronic composition. You should also try taking this one step further by constructing deliberately repetitive patterns — perhaps a bass pattern that can be used as a foundation for a particular composition. Simply record the pattern you want; then edit it carefully from the main reel, leaving an extra length of tape after the final beat of the pattern. This length of tape will affect the meter of the loop when the ends are joined. It's better to start with too much tape there and cut some out than to leave yourself too little and be forced to splice in minute sections until the timing is just right.

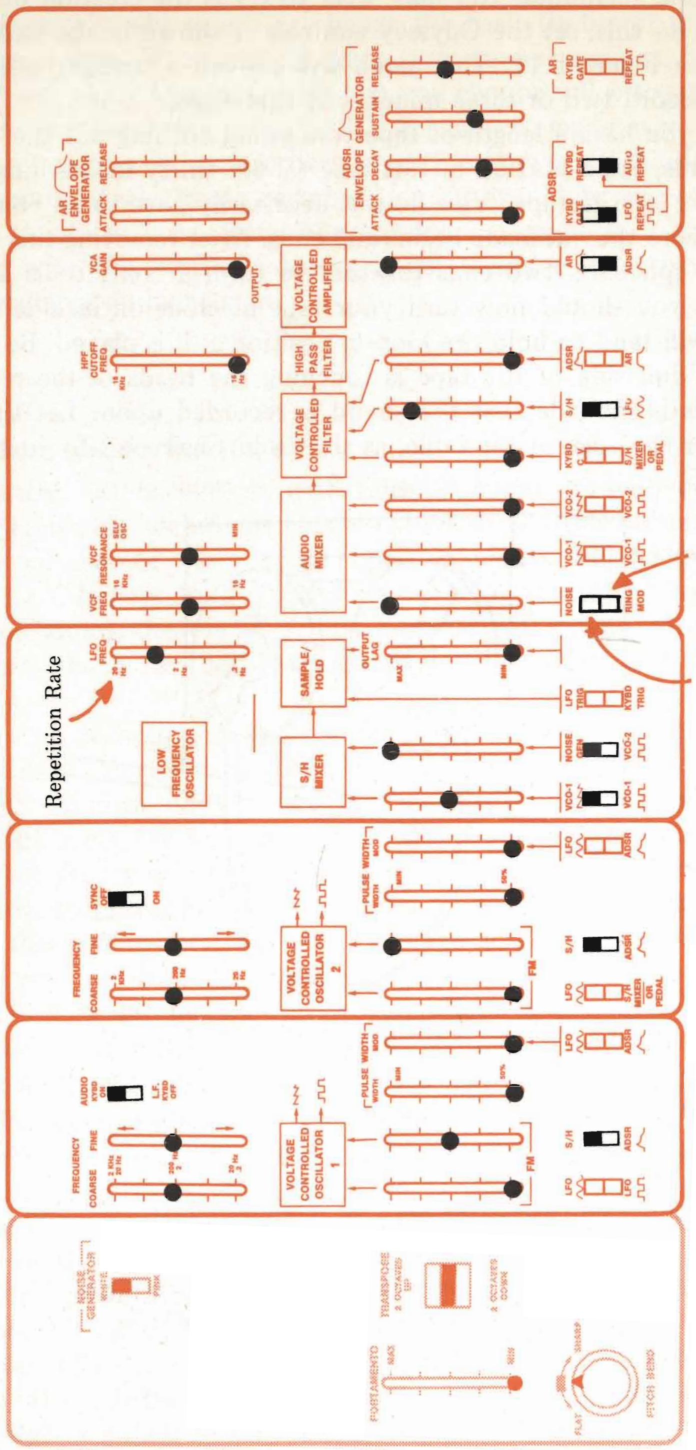


Figure 5.11. Basic patch for creating a random percussion effect.

An additional technique that you may wish to try is that of backward recording. This technique has been used with great success in the albums of a number of contemporary groups. The attraction of the technique is that it totally reverses the attack and decay characteristics of every sound. This effect is particularly noticeable when recording percussion instruments; try a rhythmic series of cymbal strokes. Record the pattern just as the drummer plays it. Then flip the reels over as illustrated in Figure 5.13.

Remember that you have now turned the tape upside down; therefore, on a four-channel machine, the track order will now be reversed: track 1 will be track 4, track 2 will be track 3, track 3 will have become track 2, and track 4 becomes the new track. Consequently, this technique will not work on quarter-track stereo recorders. With the cymbal track now playing backward, you can add other tracks to this unusual percussive background. The result is one track playing backward, the others forward, both at the same time and perfectly synchronized.

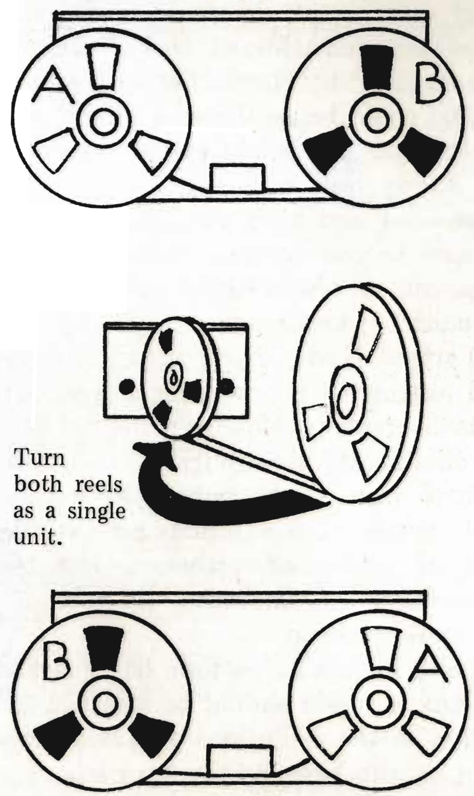


Figure 5.13. Reversing tape reels to record backward.

Other effects, such as the deliberate use of the time delay produced by the separation of the record and payback heads, can be created on certain machines. The principle of this effect, of course, is to feed the signal from the output of a particular channel (taken from the playback head) back into the input for the same channel, thus creating an echo effect by returning the signal produced by the playback head to the record head. Consult the owner's manual for your machine to determine if this is possible; a number of factors that vary widely from one machine to another will have a pronounced effect upon the quality and duration of the echo. These factors include: (1) the distance between the record and playback heads, (2) tape speed, and (3) the level of the signal. The louder the signal, the slower the tape speed; the wider the distance between the two heads, the *longer* the echo. If your machine can be operated in this way, experiment to see if musically useful results can be produced.

Conclusion: Listening for Electronic Music

While it has been the primary purpose of this text to get you involved in the actual creation of electronic music, it is hoped that a secondary goal has also been achieved: that of making you more aware of and more curious about the sounds you hear around you every day. Every natural sound could be synthesized, given the proper electronic equipment. Imagine how you might create patches for many of the sounds you hear during your daily activities; listen for other sounds that might be recorded and then modified for use in an electronic composition. Be alert to your audio-environment!

Your awareness will be rewarded by an ever-increasing amount of good electronic music — music being played by the top rock, jazz, and even classical artists. Nearly every performer of major stature who plays a keyboard instrument is now using a synthesizer, either in live performance or in his records. Moreover, the use of such instruments extends beyond all limiting classifications; musicians performing all kinds of music have realized the potentially expanded capacity for expression that electronic music synthesis can provide. Even television commercials, station identification themes, and the drama soundtracks are incorporating an increasing amount of pure electronic effects into their audio makeup.

It's not a fad, and it's not an esoteric fancy intended for a chosen few. Instead, electronic music should be regarded as what it is — the logical continuation in the evolutionary development of musical expression. Those of us who have already experienced, even in a limited way, the boundless potential of this medium are convinced that in these possibilities lie the beginnings of an exciting and significant evolutionary step in the development of the music of the world. Enjoy it.

GLOSSARY

Additive synthesis: adding sine waveforms together to create new waveforms.

Amplifier: an electronic circuit which increases the power of an electrical signal.

Amplitude: amount of a waveform's deviation from center. When used to describe sound, amplitude means volume.

Amplitude modulation: a periodic change in the amplitude of a sound; for instance, tremolo.

Aperiodic waveform: irregular, nonrepeating waveform.

Attack: beginning of a sound.

Attenuator: controls amount of signal passing through it.

Audio range: range of pitches you can hear: Roughly 20-20,000 Hz.

Band-pass filter: passes one frequency band.

Band-reject filter: rejects one frequency band.

Cutoff frequency: used when describing the characteristics of high-pass or low-pass filters to indicate the specific frequency beyond which the filter is supposed to attenuate all frequencies.

Decay: initial fading of sound (after attack).

Envelope: attack and decay of a sound.

Envelope generator: produces transient voltages useful in creating attacks and decays and special effects.

Filter: changes tone color (timbre) by removing selected harmonics.

Frequency: rate at which a waveform repeats. Expressed in cycles per second or Hertz (Hz).

Frequency modulation: a periodic change in the pitch of a sound; for instance, vibrato.

Fundamental: usually the lowest frequency component in simple waveforms, perceived as the "pitch" of the sound.

Gate: on/off signal indicating beginning, duration, and end of an event.

Harmonics: overtones that give a tone a particular sound or timbre. Harmonic frequencies are always exact multiples of the fundamental.

Harmony: two or more simultaneous tones, implying a tonality.

Hertz (Hz): term for cycles per second.

High-pass filter: passes high frequencies, cuts out low frequencies.

Low-pass filter: passes low frequencies, cuts out high frequencies.

Low-frequency oscillator: an oscillator which is designed specifically to operate at subsonic frequencies.

Mixer: combines signals.

Modulation: any periodic change in a waveform.

Noise: random signals which contain all audio frequencies.

Oscillator: generates tone or low-frequency periodic waveform.

Overtones: frequency components of a sound. May be in any mathematical relationship to the fundamental.

Patch: connection of two or more functions.

Periodic waveform: repeating wave pattern.

Phase: relationship between waveforms at any moment in time.

Phase-synchronization: forcing a fixed phase relationship between two waveforms.

Pink noise: noise which is musically balanced; high and low frequencies sound equally loud.

Pitch: perceived frequency of a sound.

Pitch bend: changing the frequency of a pitch while played.

Portamento: sliding between notes.

Pulse wave: family of waveforms with square corners.

Release: ending of a signal.

Resonance: amplifies a band of overtones. A resonance which moves in frequency can create a "wow" sound.

Ring modulator: produces a complex output from two simple input signals.

Rolloff: the effectiveness with which a filter eliminates signals which it is not supposed to pass.

Sample and hold: a circuit which can be used to store, or hold, an input voltage.

Sawtooth wave: sounds rich, full, brassy.

Signature: signs placed at the beginning of a composition indicating the key.

Sine wave: sounds smooth and pure; has no harmonics.

Square wave: sounds hollow and reedy. Square wave is a special kind of pulse wave.

Subsonic: below human hearing range; usually lower than 20 Hz.

Subtractive synthesis: filtering out certain frequencies to create new sounds.

Sustain: in synthesis, describes the level of the held part of a note in relation to the attack.

Tremolo: amplitude modulation.

Tempo: speed.

Timbre: all those qualities of a sound that make it distinctive.

Tonality: relationship of the pitches defining a "key."

Transposition: changing from one key signature to another.

Trigger: electronic impulse used most often to activate envelope generators.

Vibrato: frequency modulation.

Voltage: electrical potential.

Voltage control: a process whereby one electrical circuit is used to control the function of some other electrical circuit.

Voltage-controlled amplifier: an amplifier whose gain can be controlled by an external voltage.

Voltage-controlled filter: a filter whose cutoff frequency can be controlled by an external voltage.

Voltage-controlled oscillator: an oscillator whose operating frequency can be controlled by an external voltage.

Waveform: characteristic shape of a wave; helps determine timbre.

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