PRACTICAL SYNTHESIS FOR ELECTRONIC MUSIC

VOLUME ONE

2ND EDITION

Roland

Contents

List of illustrations and Patch Diagrams

Introduction

The purpose of Volume 1 of *Practical Synthesis for Electronic Music* is to provide practical applications of the theory of synthesis for producing useful sounds on a voltage controlled synthesizer. The theory itself can be found in *A Foundation for Electronic Music* (published by Roland), but this is not essential for comprehension of the material presented here. Generally, terms and expressions which are not explained in the text can be understood from context, and/or from examples given later.

All voltage controlled synthesizers are basically alike but for the purpose of presenting experiments in the production of specific sounds it is necessary to relate to a specific system to show control settings and their variations. For this purpose we have chosen the Roland System 100M Synthesizer because it is small, compact, and economical, yet it is a modular system with all the potential of much larger professional systems such as the Roland System 700.

The System 100M, being modular, can be built up into configurations to match the particular needs of each individual musician. The fold-out at the back of this volume shows panel diagrams of many of the modules which are available. In this book, to make it easier to read patch diagrams, we will show only those portions of the modules which are actually used in each patch. For example, when we want to show a VCO, we will probably show only half of a Model 112 Dual VCO Module rather than the whole module or the Model 110 VCO/VCF/VCA Module. Except for minor points, these VCO's are the same.

In many places specific values or standards are mentioned. It should be remembered that these values and standards will be common to many other synthesizers, but not necessarily to all synthesizers. Most of the diagrams and settings shown in this book are, of course, adaptable to other synthesizers. It is usually only a matter of comparing the System 100M with the other synthesizer, particularly in regard to how much a control movement affects the sound.

I wish to express grateful appreciation to my fellow workers at Roland without whose help and patience this book would have been impossible.

Robin Donald Graham
Synthesizer Project Manager
Roland Corporation

Osaka, Japan
September, 1979

Chapter One:
The Basic Patches

1-1 Introduction

In electronic Music there are three basic approaches to synthesizing sounds:

1. Subtractive Synthesis
2. Additive Synthesis
3. Direct Synthesis

Subtractive Synthesis is by far the most common form of synthesis used with the voltage controlled synthesizer. In subtractive synthesis we take a waveform (sound) which is rich in harmonics and filter it to produce sound with the desired harmonic content.

Additive Synthesis involves the controlling and mixing of sine waves of various frequencies and amounts to produce sound with the desired harmonic content. Additive synthesis is used to a certain extent with voltage controlled synthesizers, but true additive synthesis (the addition of large numbers of sine waves) is less common due to its complexity.

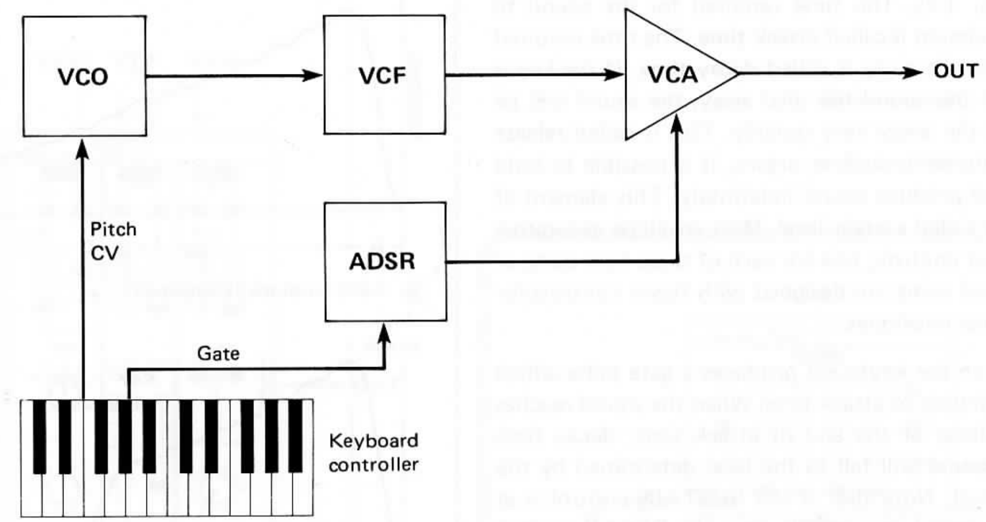
Direct Synthesis involves the generation of waveforms using data is stored in a computer memory. This form of synthesis is highly complex and requires extensive knowledge of higher mathematics and sophisticated computer programming. Nevertheless, it is possible, to a certain extent, to use the basic principle with more highly-developed voltage controlled synthesizer systems. For example, instead of using a sequencer for generating a series of voltages representing a melodic pattern, these voltages could be used for controlling a VCA to produce complex envelope patterns not possible with normal envelope generators. At any rate, the subject of direct synthesis is far beyond the scope of this book.

1-2 The Basic Patch for Subtractive Synthesis

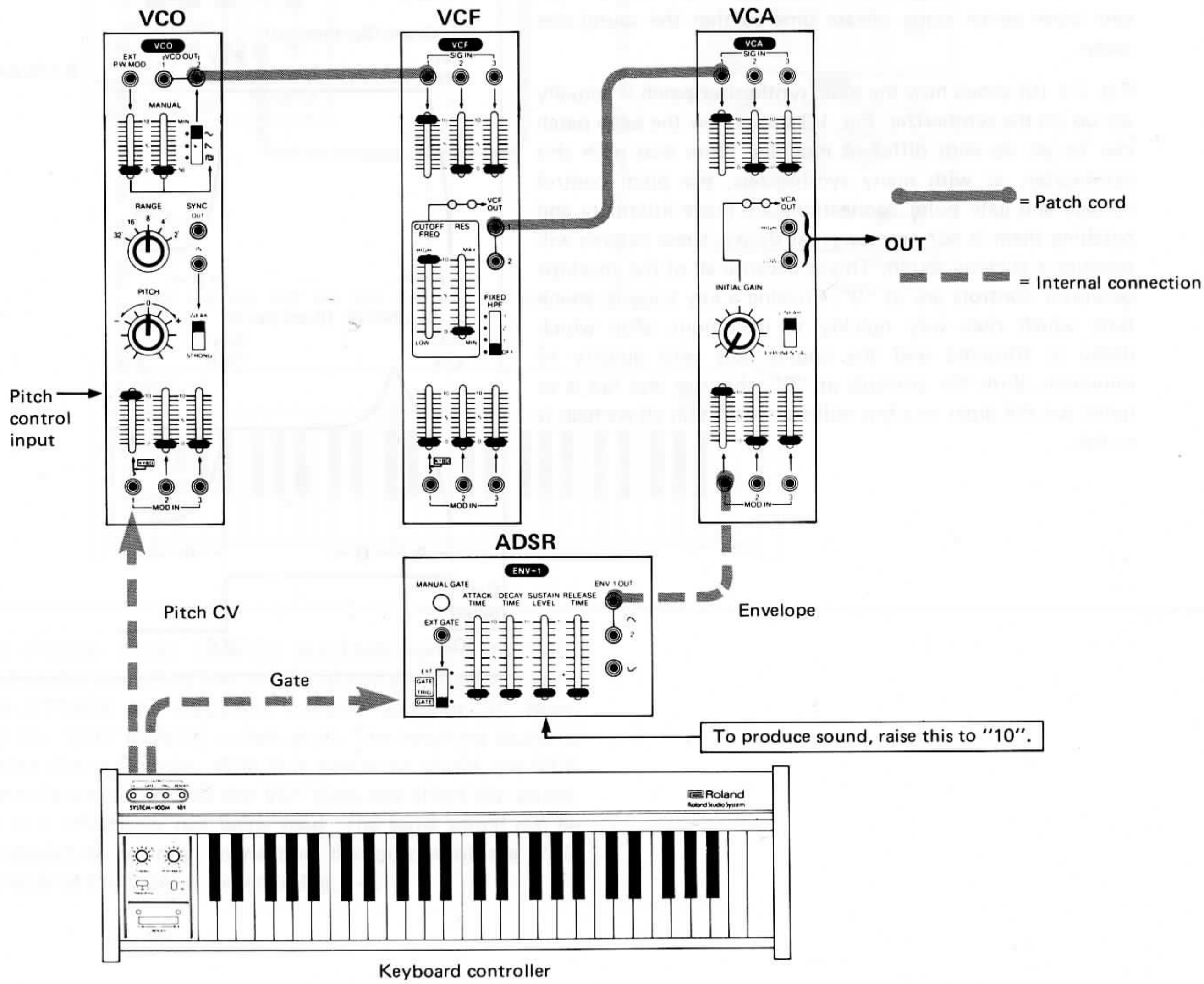
Fig. 1-1 (a) shows the block diagram for the basic synthesizer patch used in subtractive synthesis. To review how this patch works, the keyboard controller produces two outputs: a control voltage and a gate pulse. The level of the **control voltage** will correspond to the last key pressed. This control voltage output is most often used to control the frequency of a **Voltage Controlled Oscillator** (VCO), thus, when a key is pressed, the VCO will produce the pitch which corresponds to that key. The output of the VCO is a waveform (sound) rich in harmonics. **The Voltage Controlled Filter** (VCF) is used to filter and/or accent certain harmonics to give the final sound the desired tone color. The second output of the keyboard controller is a **gate pulse** which is a high level voltage (+10v) that appears at the GATE output whenever any key on the keyboard is in the depressed position. The gate pulse is most often used to trigger the **envelope generator** (also called **ADSR** for the control names) into operation. The control voltage output of the envelope generator "opens" the **Voltage Controlled Amplifier** (VCA) to let the waveform out so that when connected to an amplifier/speaker system, the synthesizer produces sound each time a key is pressed.

Fig. 1-1 Basic Patch for Subtractive Synthesis

ⓐ Block Diagram



ⓑ Patch Diagram (See alternate patch, Fig. 1-3)

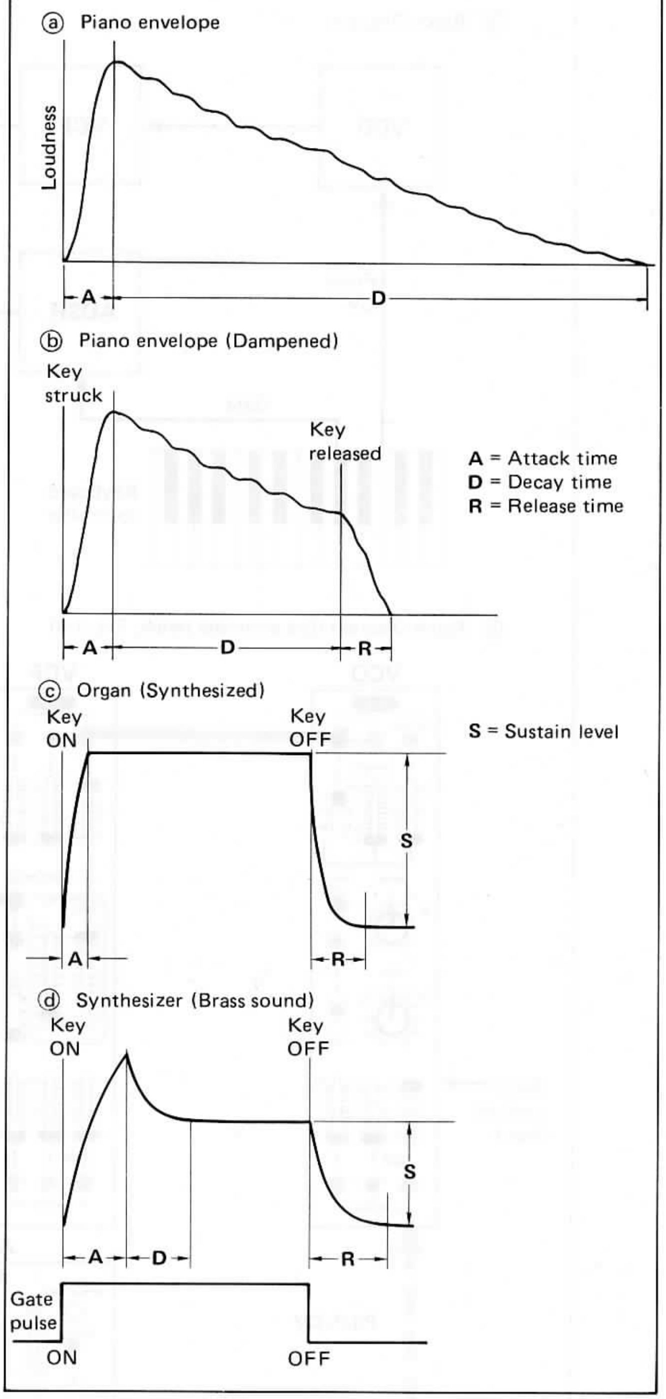


If a piano key is struck and held, the sound of the piano will jump up quickly to maximum loudness, then gradually die away. This loudness pattern is called the **envelope** of the sound (see Fig. 1-2). The time required for the sound to jump up to maximum is called **attack time.** The time required for the sound to die away is called **decay time.** If the key is released before the sound has died away, the sound will be dampened and die away very quickly. This is called **release time.** With instruments such as organs, it is possible to hold a key down and produce sound indefintely. This element of the envelope is called **sustain level.** Most envelope generators have at least four controls, one for each of these four parts of the envelope, but some are designed with fewer controls for producing simpler envelopes.

Pressing a key on the keyboard produces a **gate pulse** which triggers the beginning of attack time. When the sound reaches maximum loudness at the end of attack time, decay time starts and the sound will fall to the level determined by the SUSTAIN control. Note that if the SUSTAIN control is at maximum, the sound cannot fall, thus the DECAY control will have no effect. The sustain level will be held as long as the key remains depressed. Releasing the key cuts off the gate pulse which starts release time so that the sound dies away.

Fig. 1-1 (b) shows how the basic synthesizer patch is actually set up on the synthesizer. Fig. 1-3 shows how the same patch can be set up with different modules. Note that with this synthesizer, as with many synthesizers, the pitch control voltage and gate pulse connections are made internally and patching them is not necessary. As shown, these patches will produce a clicking sound. This is because all of the envelope generator controls are at "0". Pressing a key triggers attack time which rises very quickly to maximum, after which decay is triggered and the sound falls very quickly to minimum. With the controls at "0", this rise and fall is so quick (on the order of a few milliseconds[1]) that all we hear is a click.

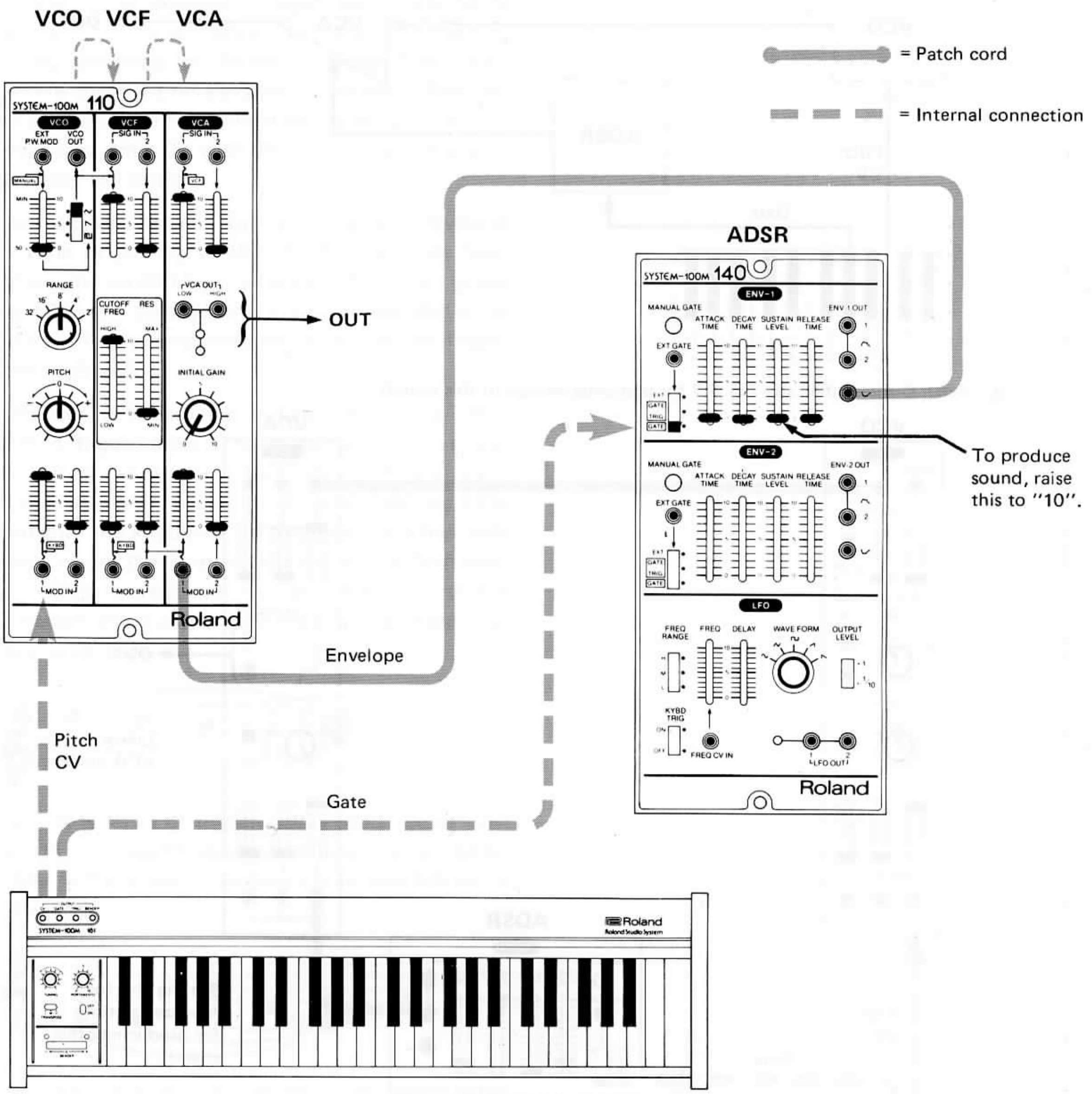
Fig. 1-2 Envelopes



1. 1 millisecond = 0.001 second

Fig. 1-3 Basic Patch Using Other Modules

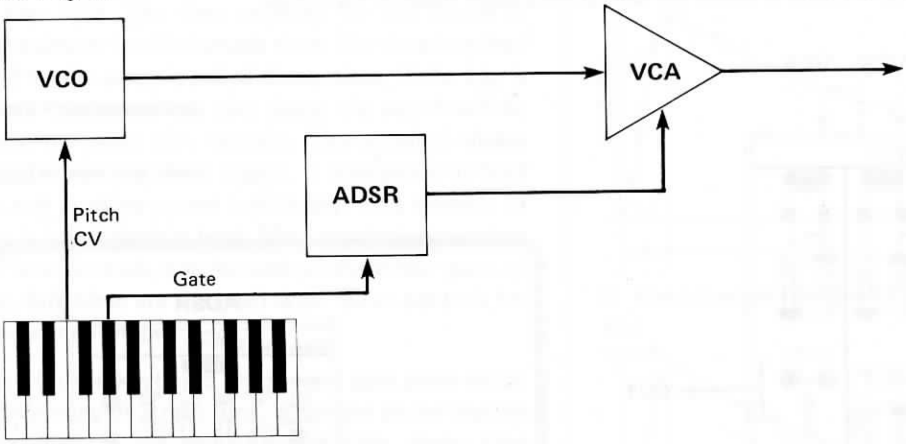
Electronically, this patch is exactly the same as the diagram shown in Fig. 1-1 ⓑ.



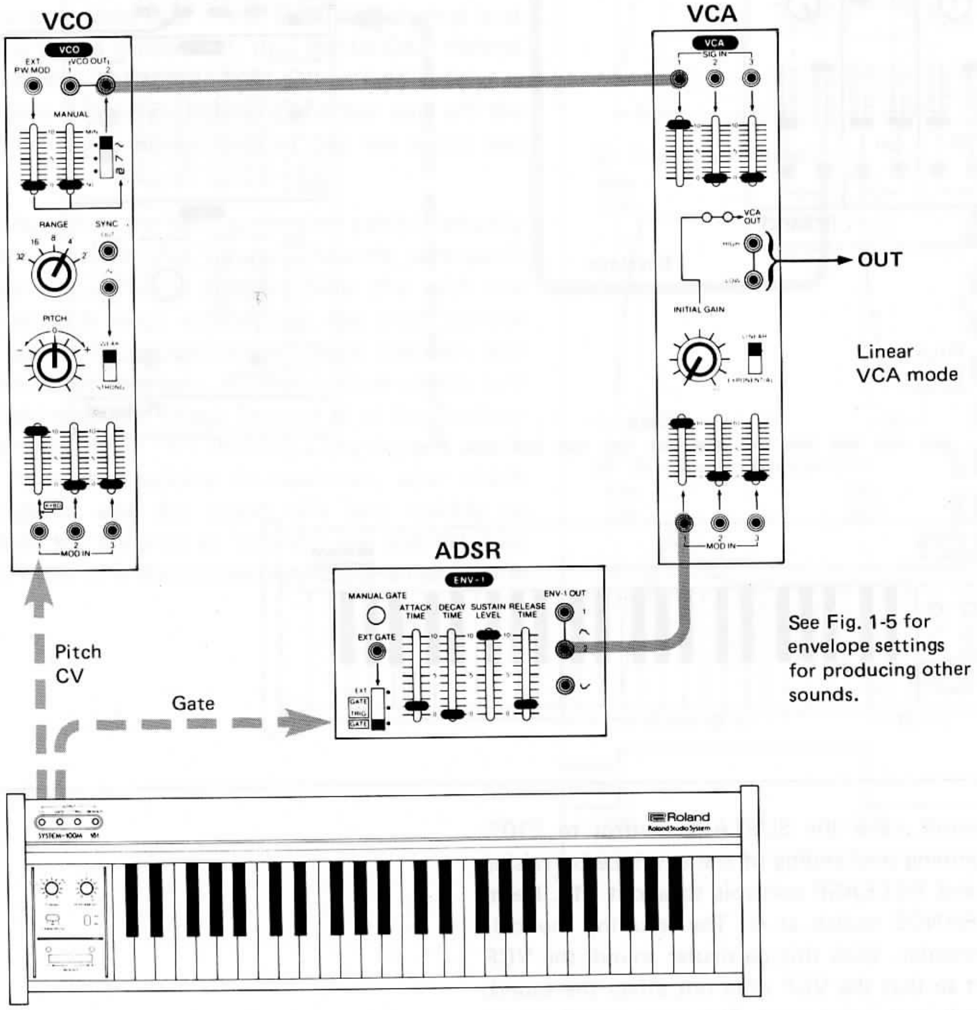
To produce sound, raise the SUSTAIN control to "10". Soften the beginning and ending of the envelope by raising the ATTACK and RELEASE controls to about "1". Next, set the VCO RANGE switch at 4′. The resulting sound is rather like a recorder. With this particular sound, the VCF controls are set so that the VCF does not affect the sound, so it is effectively not being used. The same sound can be produced by merely connecting the output of the VCO directly to the VCA as shown in Fig. 1-4.

Fig. 1-4 Simple **RECORDER**

ⓐ Block Diagram



ⓑ Patch Diagram (See also Fig. 1-7 for improved version of this sound).



Try Fig. 1-4 (b) with the ATTACK and SUSTAIN controls at "0", and the DECAY and RELEASE controls at about "5". This produces a celesta-like sound. Next try the DECAY and RELEASE at a point between "2" and "3" for a xylophone-like effect. This demonstrates how merely changing the envelope can considerably change the character of a sound.

Try playing a scale legatissimo with the celesta-like sound. Most musicians who are used to play the piano or organ will probably have a bit of trouble with this because in legato playing they tend to cling to the keys in order to produce as smooth a transition as possible between each notes being played. The result on the synthesizer keyboard controller is one single, long gate pulse for the entire passage. There is no one point where there are no keys being depressed. The first note will be loud with the following notes growing softer due to the fading decay time. Or, with the xylophone-like sound, only the first note will sound.

There are two ways to prevent this: The first is to develope a detached style of playing so that the first key has been released before the second key is pressed. This may prove quite difficult in rapid passages which are not intended to be played staccato. The second method is to use the trigger output of the keyboard.

Some synthesizer keyboard controllers produce a third output called a **trigger pulse**. The trigger pulse is a sharp, short pulse which occurs any time the keyboard pitch control voltage changes. To use this to advantage we must keep in mind that the keyboard (in this case) has a **low note priority,** which means that if more than one key is depressed, the control voltage output will represent the value of the lowest key pressed. If we play a scale moving downwards and hold each key as we press it:

the result will be a series of trigger pulses which occur each time a new key is pressed because each new key represents the lowest key in the group. If we play a scale upwards while holding all the keys:

the result will be the production of only one trigger pulse because the lowest note is held. If we release this low note, the release will produce a trigger pulse because the pitch of the lowest note has changed. In actual practice, this will probably not make much difference. Most musicians have a tendancy to develope a playing technique which produces the desired effect. When it is desired to have the envelope re-triggered for each pitch played, it will prove much easier to play if the GATE + TRIG mode of triggering the envelope generator is used. The GATE + TRIG mode is actually the more common mode, with the GATE mode being used in a special phrasing where it is desired not to re-trigger the envelope generator in a legato passage.

The sounds made by the above two envelopes may be made more percussive by using the exponential response mode of the VCA. In this case, the DECAY and RELEASE controls will have to be set a little higher than with the linear mode, as shown in Fig. 1-5 (b).

1-3 Control of Tone Color

To demonstrate the effect that the VCF has tone colors, set the basic patch shown in Fig. 1-1. Change the VCO waveform output to a sawtooth wave (|\ ) and lower the MOD IN slider which gives the VCO its pitch control voltage from the keyboard. Raise the VCA INITIAL GAIN control high enough to produce sound. With the VCF CUTOFF FREQ at "10" (HIGH), the VCO sawtooth wave passes through the VCF unchanged. If the VCF CUTOFF FREQ control is slowly lowered, the upper harmonics of the sawtooth wave will be gradually shaved off until "0" (LOW) is reached where all harmonics, including the fundamental, are filtered out.

With most sounds, the tone color of the sound will change during its production. Very often, this tone color change is closely related to the envelope of the sound. As an example, one of the factors which determines the harmonic content of the sound produced by a wind instrument is the intensity of the wind pressure exerted to produce the sound. When air is blown through a wind instrument, there is a slight delay as the wind pressure builds up before the air starts flowing through the instrument. This causes a delay in the entrance of the upper harmonics. Once the initial blast has left the bell of the instrument, the air flow and harmonic content settle down to a relatively steady state. When the air flow is stopped, there is a slight delay while the air remaining inside the instrument is expelled. As the pressure dies, the upper harmonics also die. This effect can be easily imitated by using the envelope generator to control the cutoff point of the VCF, as shown in Fig. 1-6 (a).

With slight variations, the settings shown in Fig. 1-6 (b) are very commonly used in producing brass sounds. The VCF CUTOFF FREQ control is set at its lowest point so that all the harmonics are filtered out of the sawtooth wave. When a key is pressed, the envelope generator is triggered which opens the VCA. At the same time, with the MOD IN control of the VCF at "8", the envelope control voltage from the envelope generator causes the VCF cutoff point to sweep quickly upwards (attack) then quickly down (decay) to the level set by the SUSTAIN control. This action causes a very quick but large change in the harmonic content of the sound at the beginning of each note played. When the key is released, another harmonic change takes place as the envelope sweeps the VCF cutoff point downwards (release).

Fig. 1-5 Effects of the Envelope on a Sound
(Try these envelopes with the patch in Fig. 1-4 ⓑ).

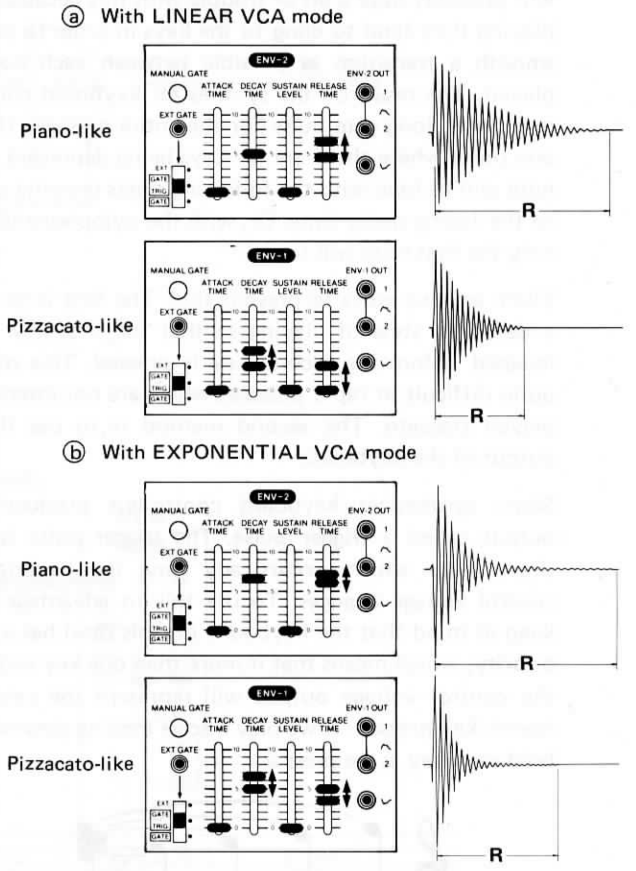
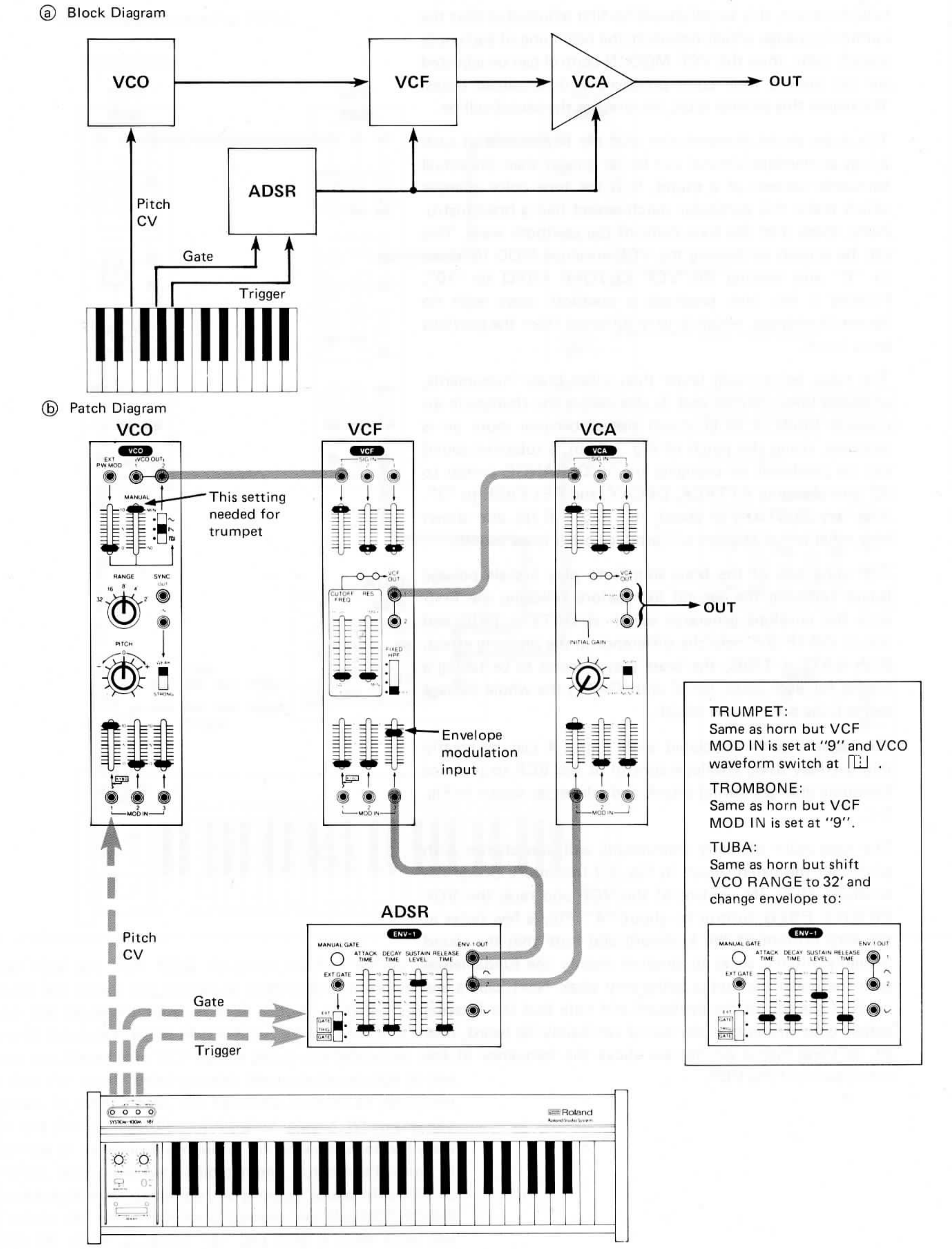


Fig. 1-6 HORN (also **TRUMPET, TROMBONE, TUBA**)



The level of the SUSTAIN control will determine the harmonic content of the brass sounds while the notes are held; however, this sound should be first adjusted so that the harmonic *change* which occurs at the beginning of each note sounds right, then the VCF MOD IN control can be adjusted for the desired tone color produced with sustained notes. The higher this control is set, the brassier the sound will be.

This brass sound demonstrates that the importance of such things as envelope control can be far greater than the actual harmonic content of a sound. It is the tone color *changes* which make this particular patch sound like a brass instrument, more than the tone color of the sawtooth wave. This can be proven by moving the VCF envelope MOD IN slider to "0" and moving the VCF CUTOFF FREQ to "10". Pressing a key now produces a sawtooth wave with no harmonic changes, which is very different from the previous brass sound.

The tuba, being much larger than other brass instruments, produces lower pitches and its size causes the changes in air pressure inside it to be much slower because more air is required. Using the patch of Fig. 1-6 (b), a tuba-like sound can be produced by changing the VCO RANGE switch to 32' and changing ATTACK, DECAY, and RELEASE to "2". Also, try SUSTAIN at about "6". Fig. 1-6 (b) also shows how other minor changes will produce other brass sounds.

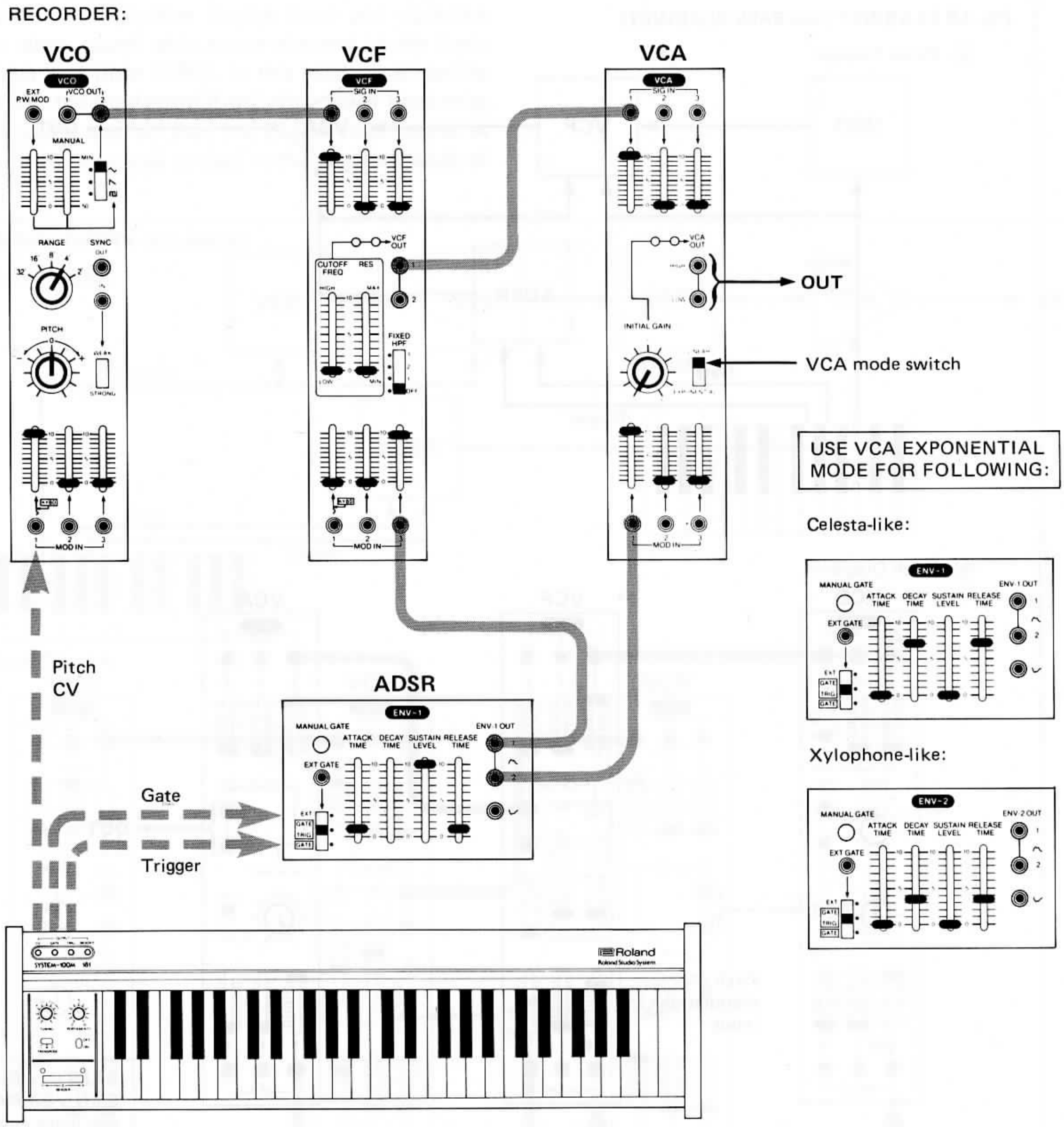
Try using one of the brass sounds to play a scale passage legato (pressing the second key before releasing the first) with the envelope generator switch at GATE + TRIG and also at GATE and note the difference in the phrasing effect. With GATE + TRIG, the brass player seems to be taking a breath for each note, while with GATE, the whole passage seems to be done in one breath.

The three sounds associated with Fig. 1-4 can be greatly improved by using envelope control of the VCF to produce harmonic changes during attack and release, as shown in Fig. 1-7.

The tone color of many instruments will also change with pitch. Set the patch shown in Fig. 1-7 (recorder). Lower the envelope MOD IN control of the VCF and raise the VCF CUTOFF FREQ control to about "4". Play a few notes at the very bottom of the keyboard and note that the sound is highly filtered so as to produce mostly the fundamental with the upper harmonics being very weak. Next, play a few notes at the top of the keyboard and note that the filtering action is so strong that the sound can barely be heard, if at all, as these higher pitches are above the frequency at the cutoff point of the VCF.

Fig. 1-7 RECORDER, CELESTA, XYLOPHONE

Block diagram same as Fig. 1-6 (a).



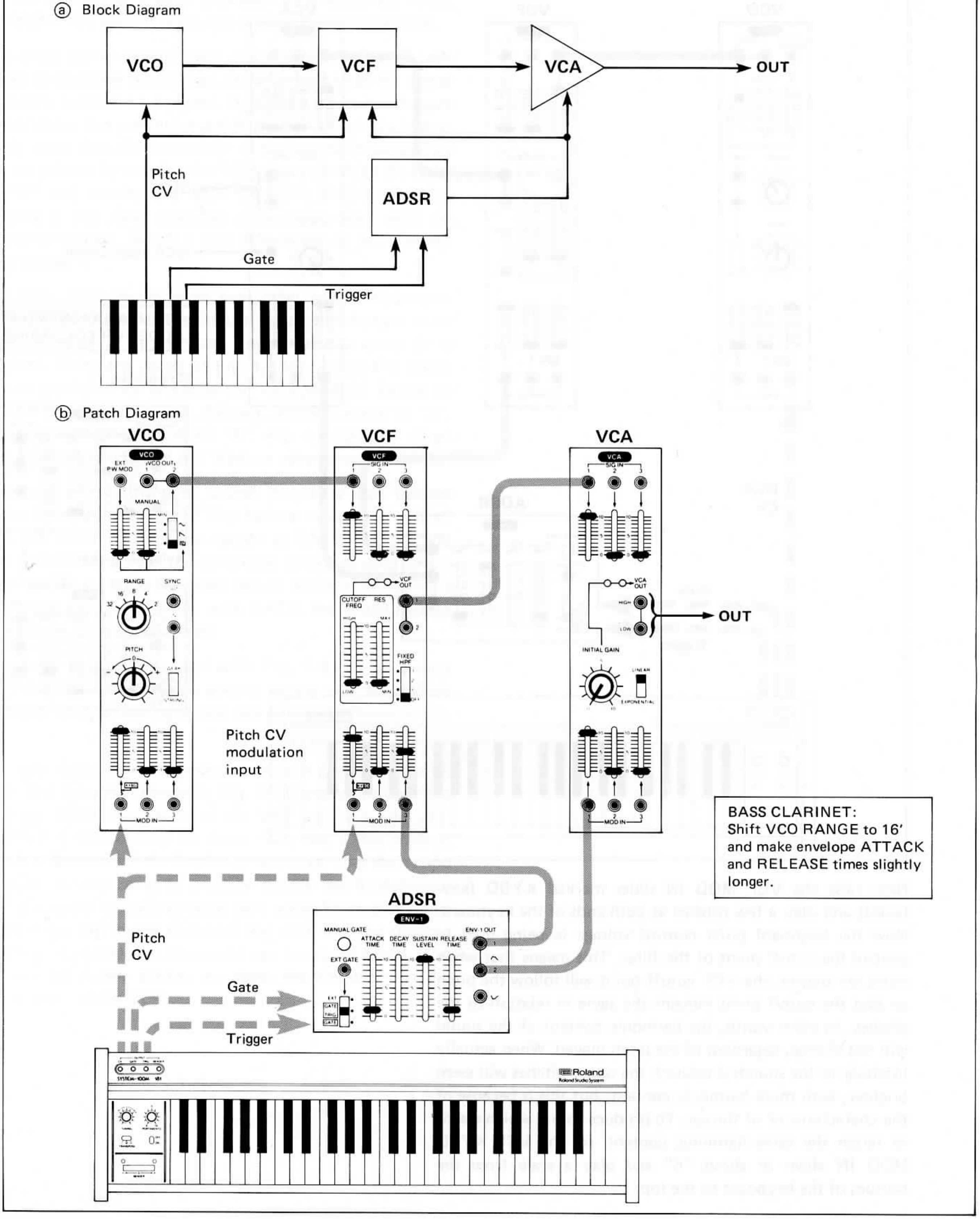
Next raise the VCF MOD IN slider marked KYBD (keyboard) and play a few pitches at both ends of the keyboard. Now the keyboard pitch control voltage is being used to control the cutoff point of the filter. This means that when notes are played, the VCF cutoff point will follow the pitch so that the cutoff point remains the same in relation to the pitches. In other words, the harmonic content of the sound will not change, regardless of the pitch played. When actually listening to the sounds produced, the upper pitches will seem brighter, with more harmonic content, but this is because of the characteristics of the ear. To produce sound which seems to retain the same harmonic content, set the VCF KYBD MOD IN slider at about "6" and play a scale from the bottom of the keyboard to the top.

Fig. 1-8 shows a clarinet-like sound in which the keyboard control voltage is added to the patch for producing a sound which seems to get brighter as the pitch moves up.

Fig. 1-8 CLARINET (also **BASS CLARINET**)

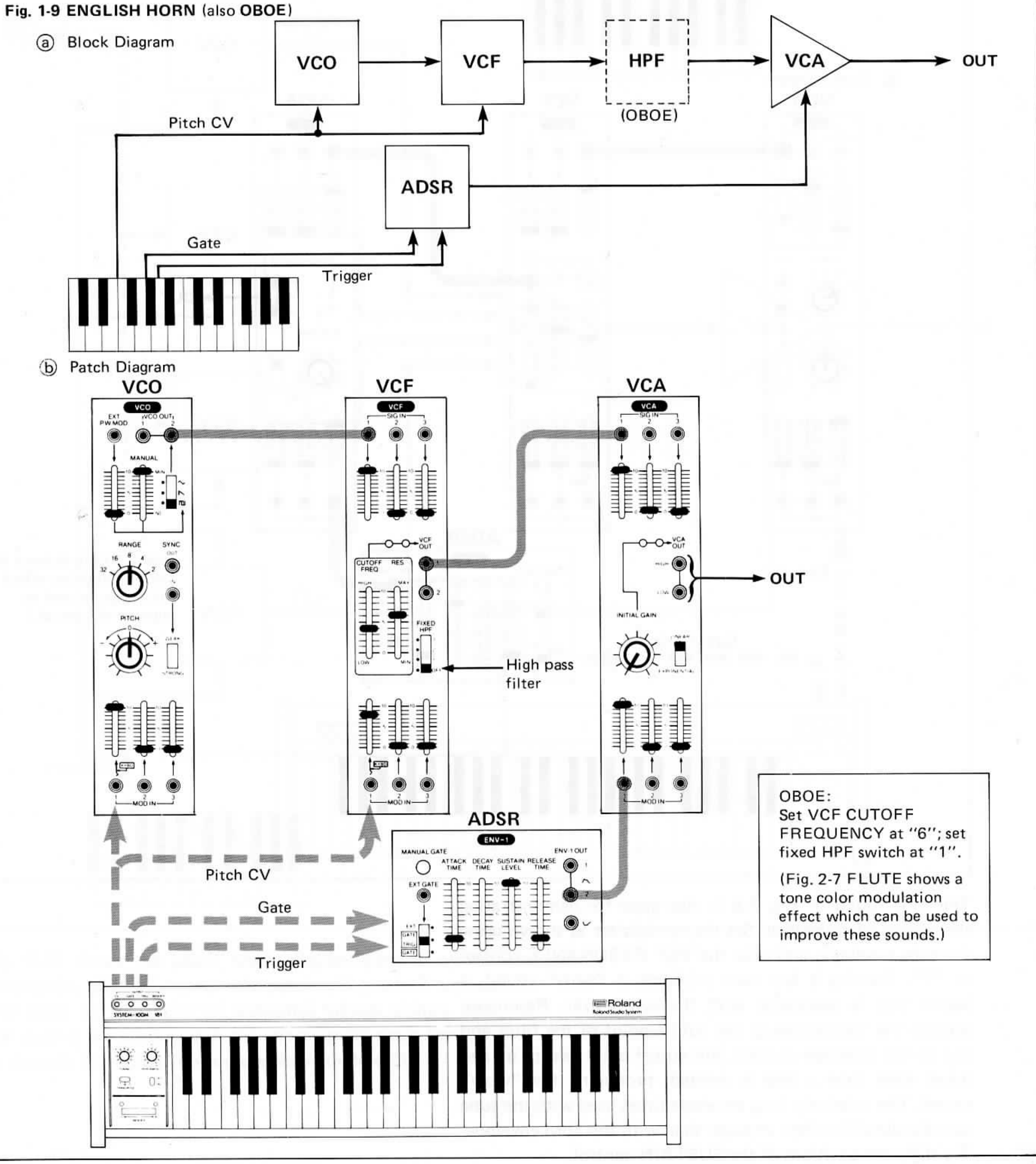
ⓐ Block Diagram

ⓑ Patch Diagram



The VCF RESONANCE control is used for accenting the frequencies at the cutoff point of the filter. It can be considered to add a nasal quality to the sound, like that produced by double reed instruments. Fig. 1-9 shows the use of VCF resonance to produce English horn- and oboe-like sounds. The oboe sound adds a new element to the basic patch: the **high pass filter** (HPF). In this patch, it is used to slightly suppress the fundamental and other lower harmonics to make the sound less full than the English horn sound. A higher amount of resonance is used in the bassoon patch of Fig. 1-10.

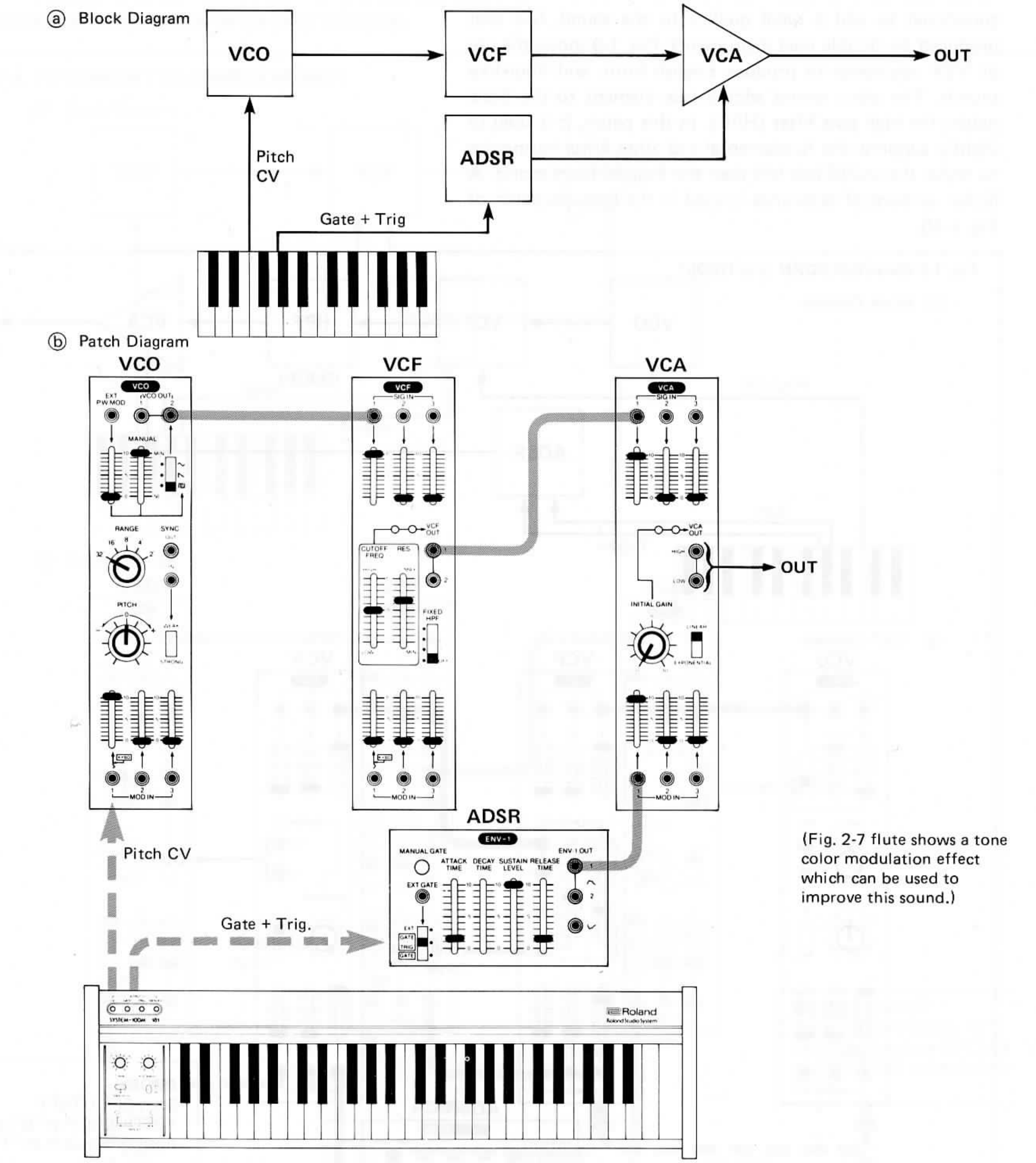
Fig. 1-9 ENGLISH HORN (also **OBOE**)



OBOE:
Set VCF CUTOFF FREQUENCY at "6"; set fixed HPF switch at "1".

(Fig. 2-7 FLUTE shows a tone color modulation effect which can be used to improve these sounds.)

Fig. 1-10 BASSOON



(Fig. 2-7 flute shows a tone color modulation effect which can be used to improve this sound.)

The tuba sound of Fig. 1-6 is also good for demonstrating the effects of resonance. Set the synthesizer as shown using the tuba envelope, and raise the VCF RESONANCE control to "7". Pressing a key now produces a "wow" sound, a sound that is associated with the synthesizer. Resonance accents the frequencies at the cutoff point of the filter and due to the envelope control, this cutoff point sweeps up and down each time a key is pressed, producing the "wow" sound. The relatively long envelope times used with the tuba sound makes this effect stronger than with the horn envelope. Try different positions of the SUSTAIN control.

1-4 The Basic Patch for Additive Synthesis

Fig. 1-11 shows a simplified block diagram for additive synthesis. The mixer is used for mixing the sine wave outputs of the VCO's in the proportions desired. Since the harmonic content of most sounds changes often during production, the ideal patch would include one ADSR/VCA combination for each VCO output. Also, most sounds have far more harmonics than the four sine waves shown.

Fig. 1-11 Additive Synthesis Block Diagram

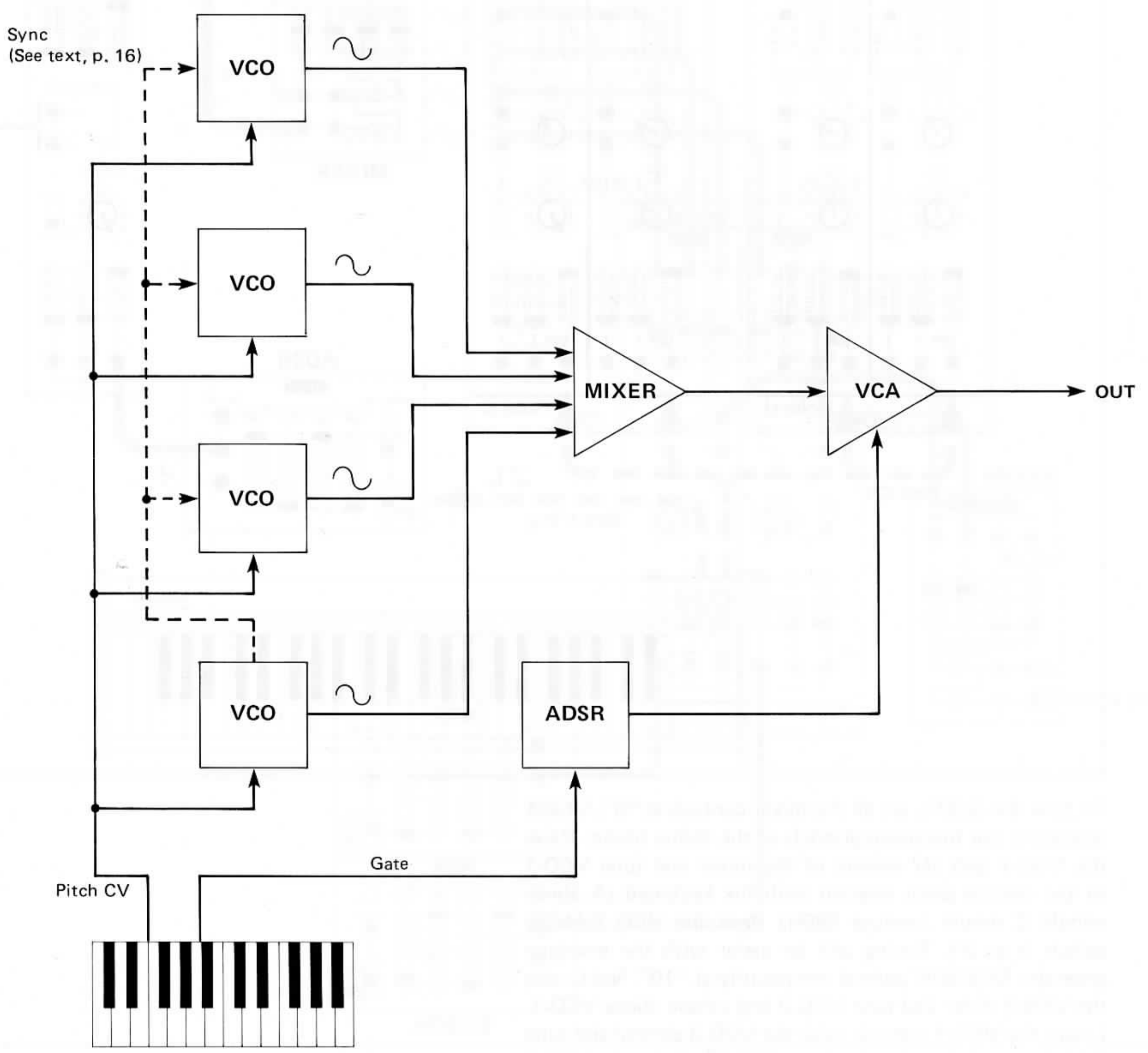


Fig. 1-12 shows the above block diagram patched on the synthesizer. Set as shown, this patch will produce sounds very close to the sound of orchestral tubular chimes. The best sounds are produced in the approximate range between C_4 (middle C) and C_5 (one octave above middle C).

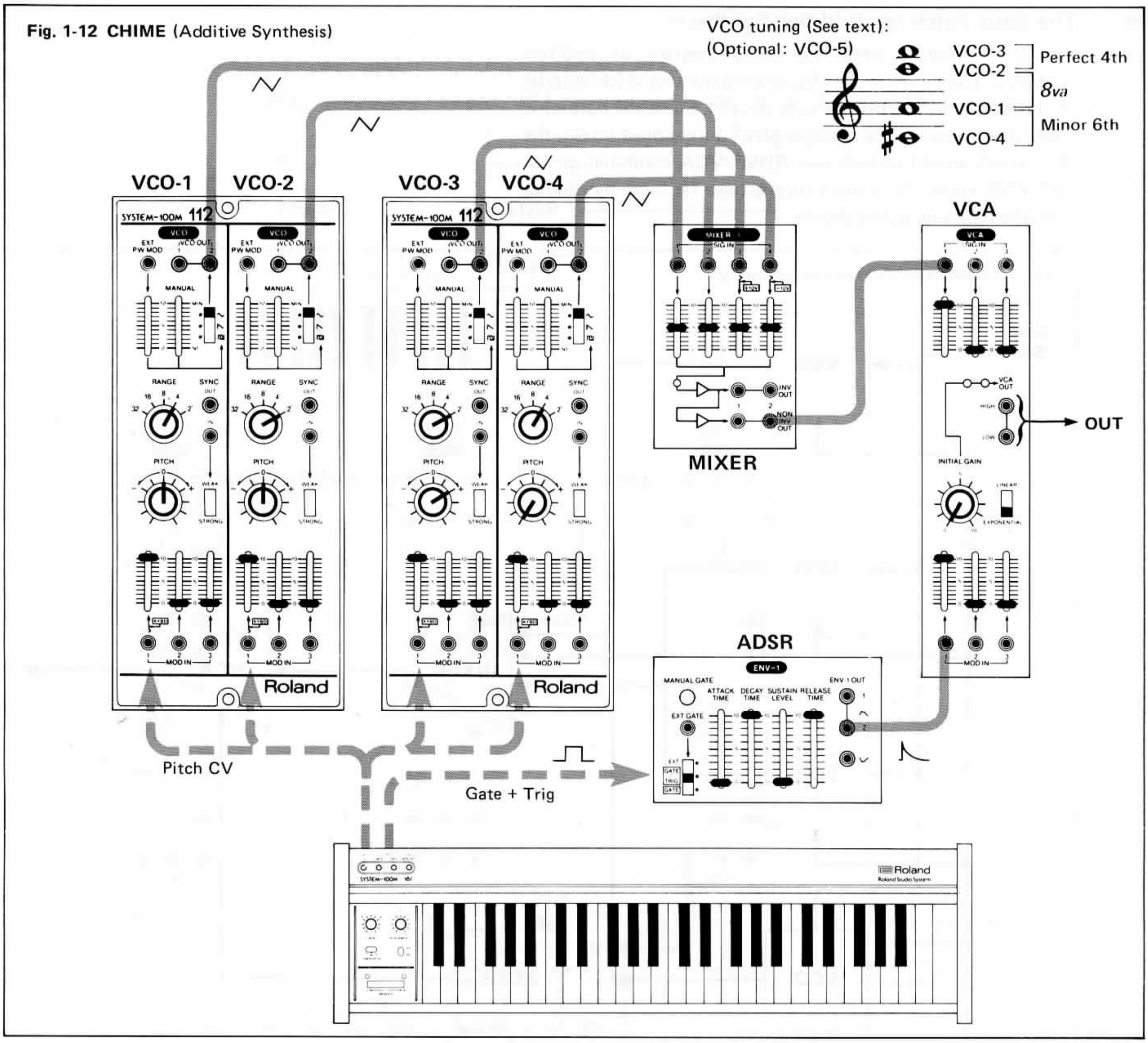


Fig. 1-12 CHIME (Additive Synthesis)

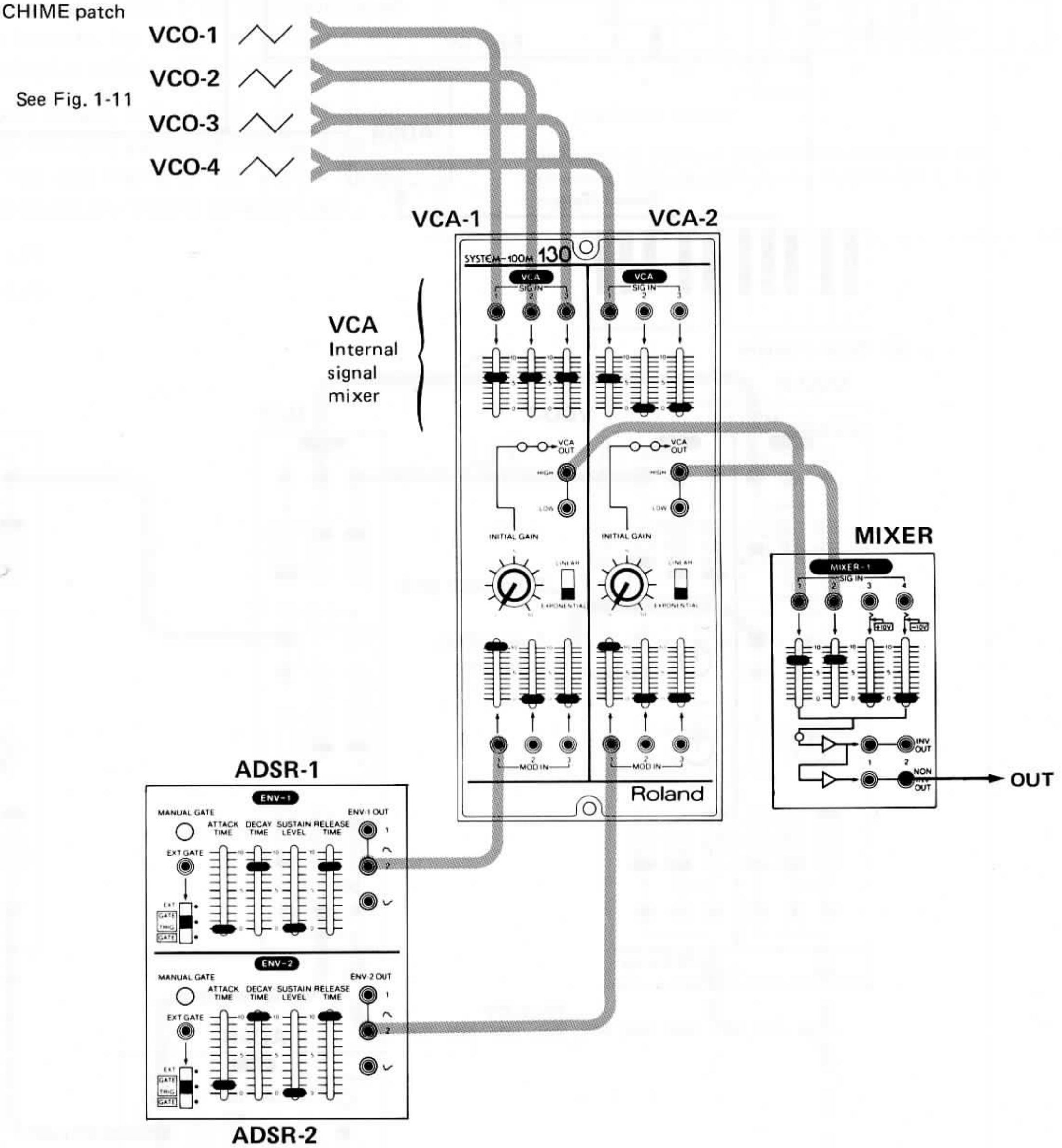
To tune the VCO's, set all the mixer controls at "0". VCO-1 represents the fundamental pitch of the chime sound. Raise the VCO-1 SIG IN control of the mixer and tune VCO-1 to the desired pitch relation with the keyboard (A above middle C should produce 880Hz since the VCO RANGE switch is at 4'). Tuning will be easier with the envelope generator SUSTAIN control temporarily at "10". Next, raise the VCO-2 slider and tune VCO-2 one octave above VCO-1. Lower the VCO-1 control; raise the VCO-3 control and tune VCO-3 to a perfect fourth above VCO-2[2]. Raise the VCO-4 and VCO-1 controls and tune VCO-4 to a minor sixth below VCO-1[3]. This sound can be improved by adding a fifth VCO tuned a perfect fourth below VCO-2.

2. To tune: With only the VCO-2 control raised, press F and note the pitch; lower the VCO-2 control and raise the VCO-3 control; press C one fourth below the previous F and tune VCO-3 to the F pitch noted before; with both VCO-2 and VCO-3 controls raised, fine tune VCO-3 so as to eliminate the beat.

3. Tune VCO-4 so that it produces the pitch of C when the E a minor sixth below is pressed.

Fig. 1-13 shows a variation of the chime patch. The lowest pitch in the patch is fed to its own separate VCA, controlled by a second envelope generator (ADSR-2). The ATTACK control of this envelope generator is set at about "1", thus, when a key is struck, the lower pitch is slightly late in entering the total sound. Also, with ADSR-1 DECAY and RELEASE at "8" instead of "10", the higher pitches die out slightly faster than the lower pitch. The ideal patch would contain an ADSR/VCA combination for each VCO output.

Fig. 1-13 Variation of CHIME patch



Regardless of how perfectly the VCO's are tuned, they will eventually drift off-pitch, causing beat frequencies to appear. In some sounds, such as the chimes above, this beating is not undesireable but actually greatly enhances the sound; in other sounds, the beating would be intolerable. The VCO **sync** function can be used to lock the frequency of one or more VCO's to some musical interval in relation to a controlling VCO. In Fig. 1-11, the controlling VCO is the bottom one.

Fig. 1-14 CLARINET (Additive synthesis)

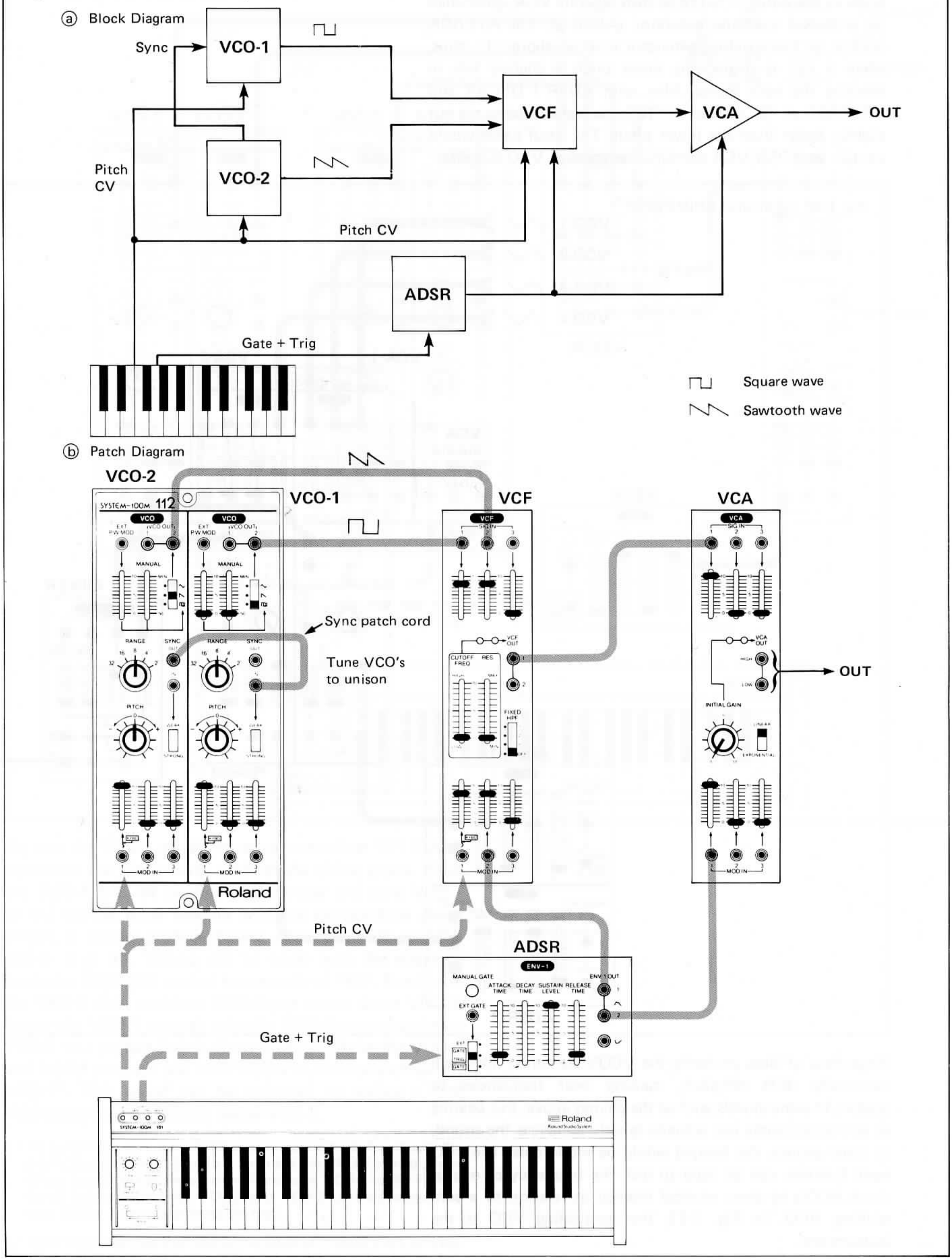
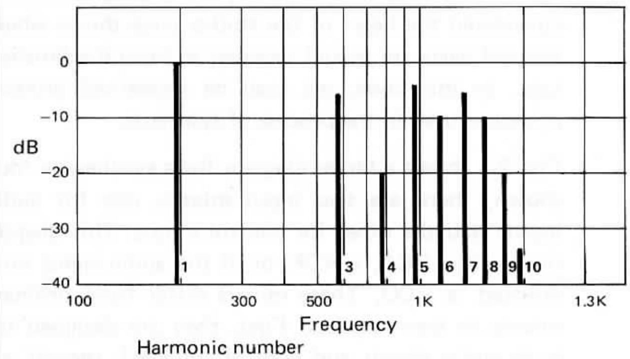


Fig. 1-14 shows an improved clarinet sound which is a better example of how the principles of additive synthesis are applied to the voltage controlled synthesizer. The sawtooth wave ( ) contains all harmonics, while the square wave ( ) contains only the odd-numbered harmonics. When these are mixed together at the VCF input, the odd-numbered harmonics of the square wave reinforce the odd-numbered harmonics in the sawtooth wave. In the total sound, the odd-numbered harmonics are very strong and the even-numbered harmonics are rather weak, which agrees roughly with the spectrum of the clarinet shown in Fig. 1-15. This addition of harmonics takes place because, by means of the sync, the two VCO's are phase locked at unison with each other.

Since the interval used is unison, either VCO could be used as the controlling VCO. The sync patch cord of Fig. 1-14 (b) could be reversed so that one end is in the VCO-1 SYNC OUT jack and the other end in the VCO-2 SYNC IN jack.

Fig. 1-15 Spectrom for the Clarinet

Pitch = G_3



The length of the lines indicates the strength of the harmonics. Note that the second harmonic is missing from the sound.

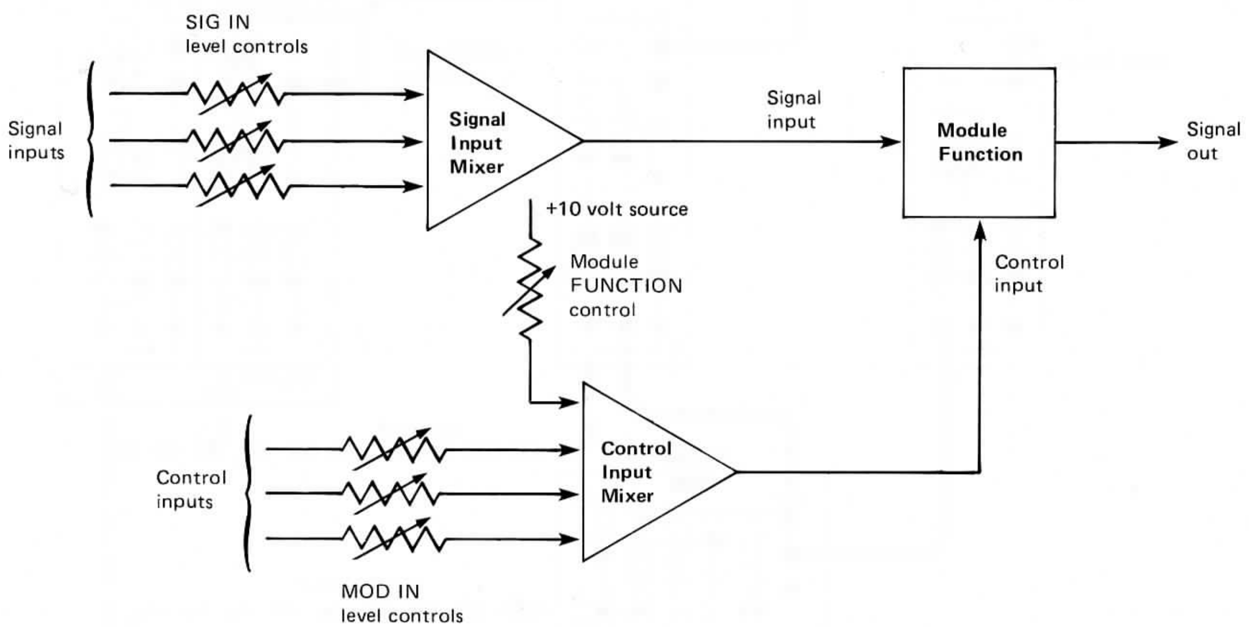
Chapter Two:
Expansion of the Basic Patches

2-1 Introduction

As the name implies, **mixers** are used for mixing individual signals together. In the studio, the mixing console might be considered the heart of the studio since this is where all the musical parts are mixed together to form the finished master tape. In this book, we shall be concerned primarily with mixers as used in the process of synthesis.

Fig. 2-1 shows a block diagram for a synthesizer module. As shown, there are two input mixers, one for audio signal inputs and the other for control inputs. This diagram could represent a VCA, a VCF, or, if the audio signal mixer were omitted, a VCO. These mixers differ from ordinary audio mixers in two respects: First, they are designed to handle both audio signals and control voltages[1]; second, and more important, these mixers are summing mixers. This means that the output will be the algebraic sum of the inputs at any given instant. For example, if the control inputs consist of two control voltages, one +5 volts and the other −2 volts, the output will be (+5) + (−2) = +3 volts. This is important because manual control of the module function is derived by means of varying a voltage source input to the control input mixer, as can be seen in Fig. 2-1. In the VCF, the control marked "function control" would be the VCF CUTOFF FREQ control; in the VCA, the VCA INITIAL GAIN control; and in the VCO, a combination of the PITCH and RANGE controls.

Fig. 2-1 Block Diagram for Synthesizer Module



1. In the case of the audio signal input mixers, the fact that they pass control voltages is of little use since the modules themselves will not pass fixed voltages through their signal inputs.

There is one major limitation to this summing process. The mixer can produce maximum output levels of slightly over 10 volts (either "+" or "–"). This means that if three inputs of +10 volts are used, the summed output will be a little over +10 volts rather than +30 volts. This point should be kept in mind because it represents a distortion of what the true output should be. With control inputs, this can usually be recognized easily. When the module function is driven to its maximum limit, a further increase of control voltage will have no effect on the function. With signal mixers, the result is distortion of the sound, which will usually be indicated by the red LED's in the VCF and VCA. Distortion can be avoided by keeping the level controls somewhat lower than maximum when there are more than one or two high-level inputs.

In the case of the VCF, if the CUTOFF FREQ control is raised to maximum, the result is a +10 volt control voltage input to the VCF. Since this represents the maximum high for the cutoff point, it can be seen that additional control voltages at the control inputs will have no effect on the cutoff point of the filter unless they are negative voltages. As another example, if the VCF produces the desired tone color, and it is later decided to add a control voltage input from a source such as an envelope generator, the CUTOFF FREQ control will have to be lowered an appropriate amount to compensate because the actual cutoff point of the filter is decided by the sum of the control inputs.

2-2 Frequency Modulation

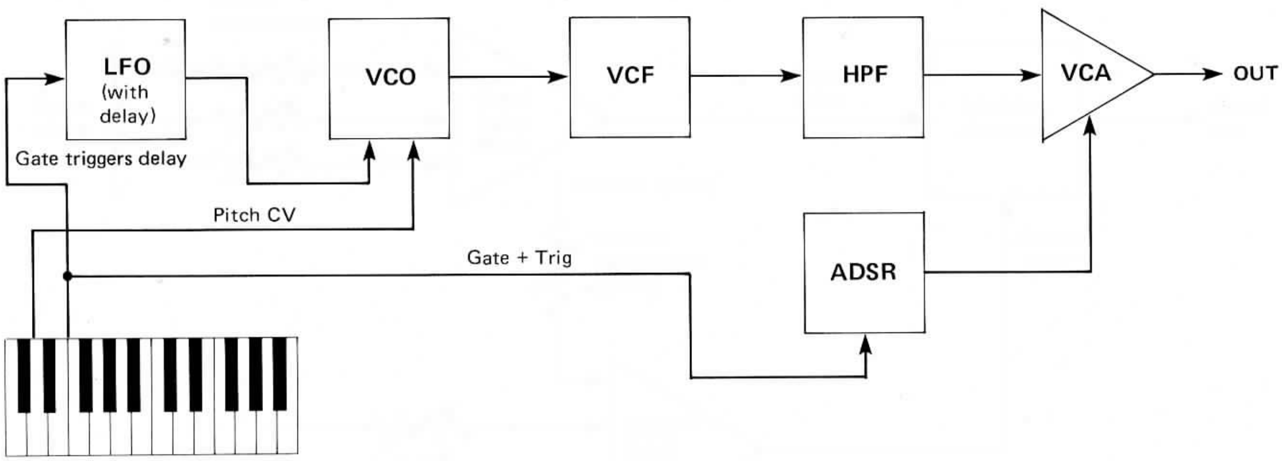
Modulation refers to the control of one parameter by another. **Frequency modulation** is the control of a frequency by means of some parameter, such as a control voltage, as when controlling pitch with the keyboard controller. Another form of frequency modulation commonly used in synthesis is the control of frequency by means of a low frequency waveform generated by a **Low Frequency Oscillator** (LFO). Using a low frequency sine wave to modulate the pitch of a VCO produces a wavering of pitch called **vibrato.**

Fig. 2-2 shows the block diagram and patch diagram for a violin sound which uses vibrato. Also shown are variations needed to produce other string instrument sounds. Try the violin sound with the LFO MOD IN to the VCO at "0" and notice that it does not sound very much like a violin at all. Next, lower the LFO DELAY control and raise the LFO MOD IN control to "2" or "3" and note the sound. Last raise the LFO DELAY control back up to about "3". With the delayed entrance of the vibrato effect produced by the built-in LFO delay, the violin sound becomes very natural.

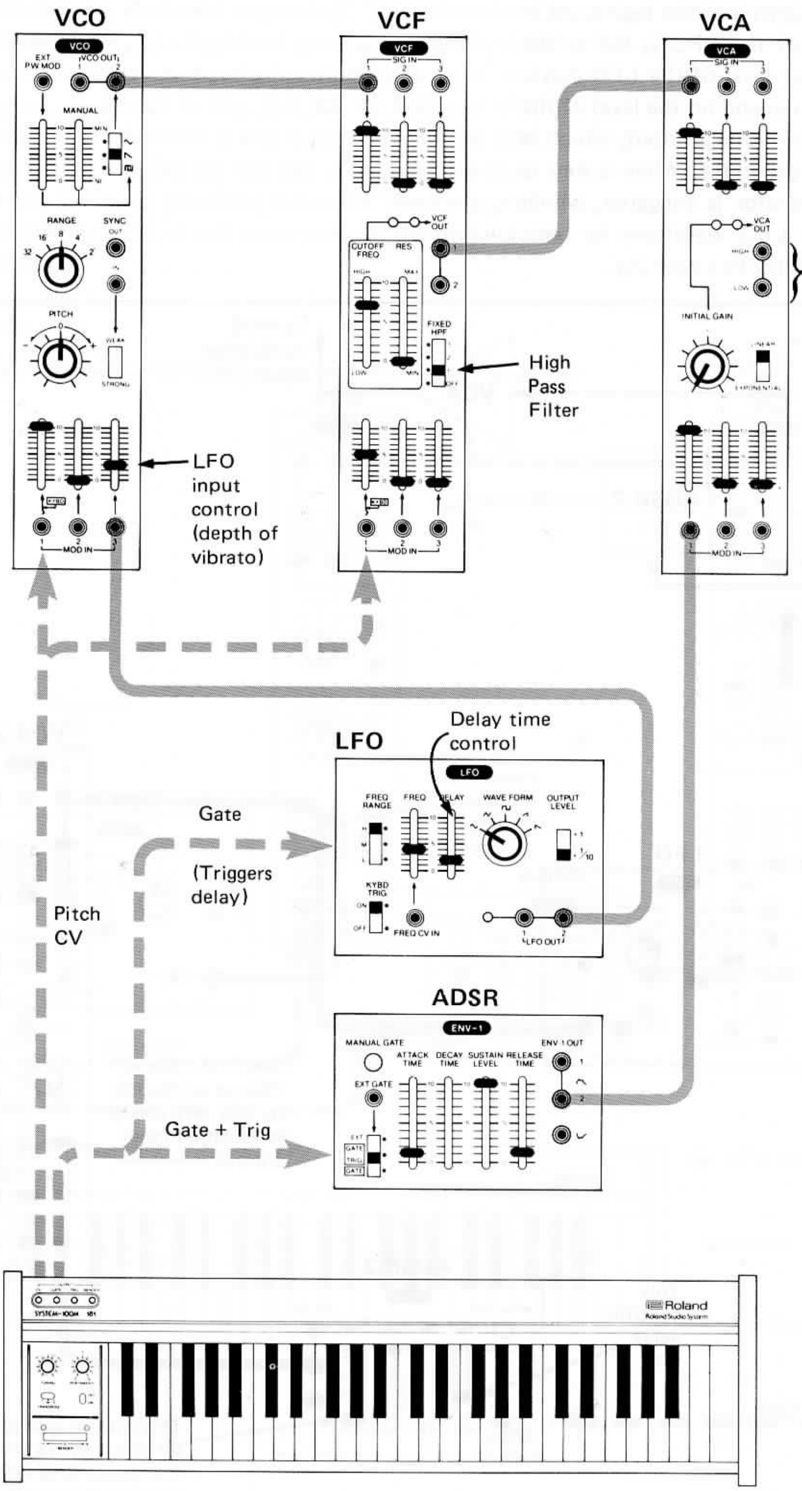
The delay effect is triggered by the keyboard gate pulse rather than the keyboard trigger pulse, so the delay effect occurs only on the first note of any passage played legato.

Fig. 2-2 VIOLIN with Delayed Vibrato (also **VIOLA, CELLO, STRING BASS**)

ⓐ Block Diagram

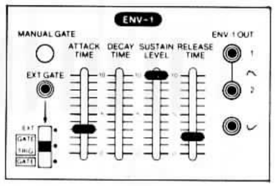


ⓑ Patch Diagram

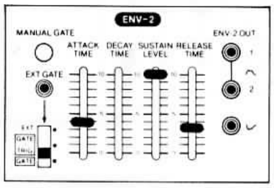


VIOLA:
Set HPF switch at OFF and raise VCF RESONANCE control to "1".

'CELLO:
Same as viola, but set VCO RANGE at 16' and envelope:



STRING BASS:
Same as viola, but set VCO RANGE at 32' and envelope:



(Fig. 4-4 shows typical equalizer settings for improving these sounds.)

LFO's in some synthesizers do not have a built-in delay function. Fig. 2-3 shows how a delay function can be patched. Even with the built-in LFO delay, this patch is worth studying because it gives insights into the logic used in synthesizer patching. The output of the LFO is fed to the signal input of a VCA. The output level of the LFO waveform passed through the VCA will depend on the level of the control voltage at the VCA control voltage input, which is derived from a second envelope generator. When a key is pressed, the second envelope generator is triggered, which "opens" the VCA to allow the LFO waveform to pass. Raising the ATTACK control will delay this opening.

Fig. 2-3 Delayed Vibrato

ⓐ Block Diagram

ⓑ Patch Diagram

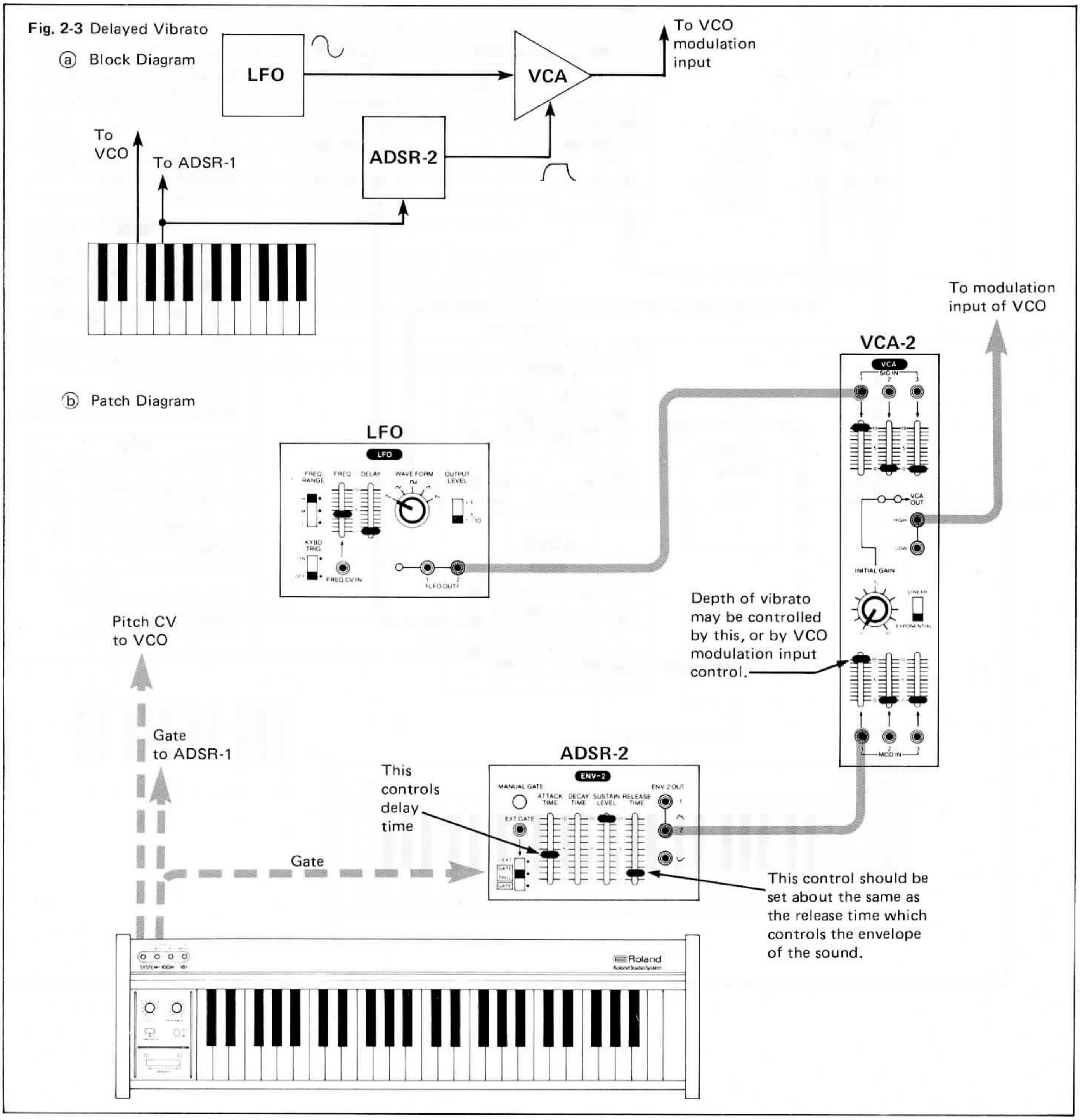
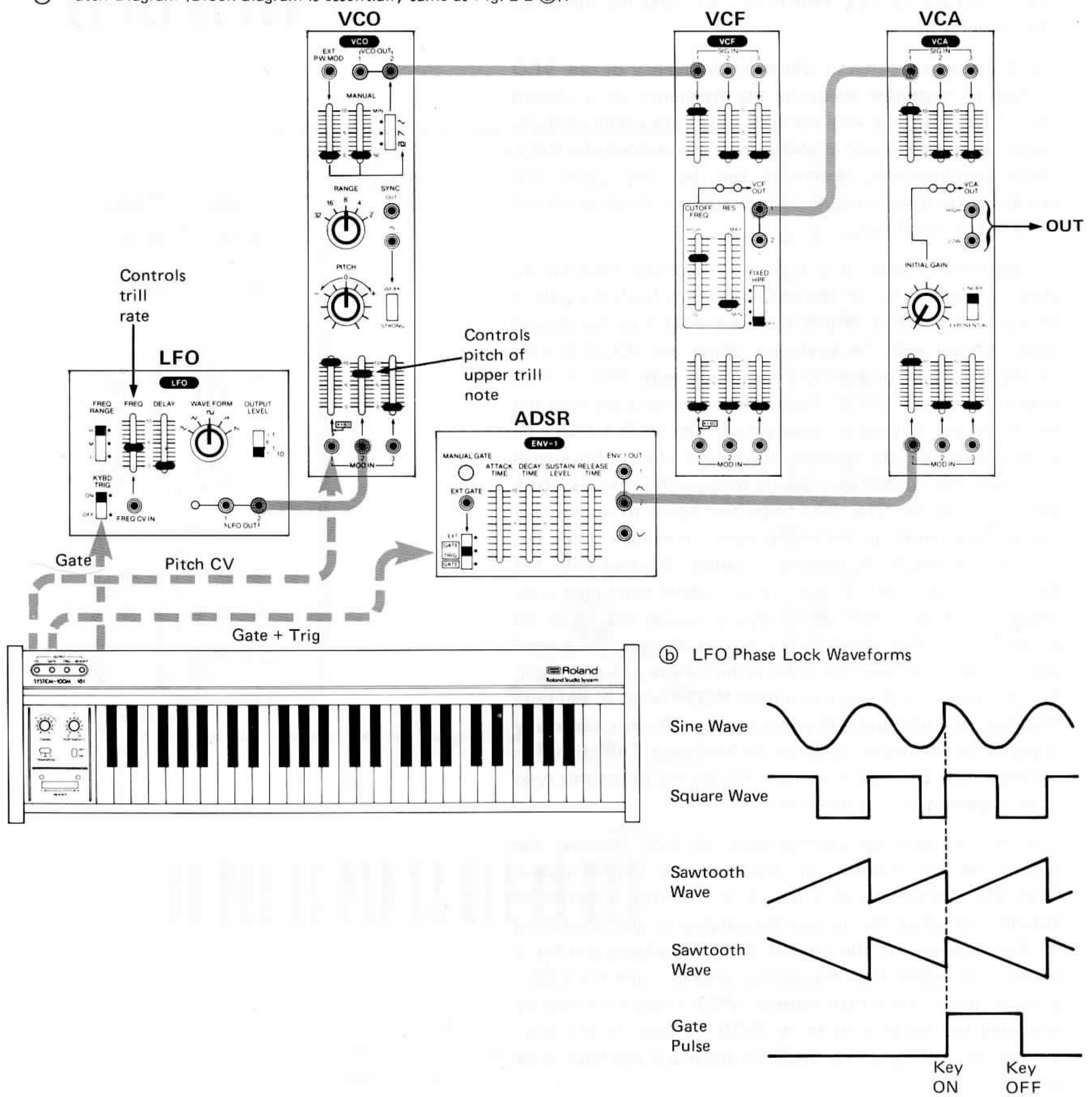


Fig. 2-4 (a) shows how trills can be produced by using the square wave output of the LFO to control the pitch of the VCO. The KYBD TRIG switch is ON so that the square wave is phase locked to the keyboard gate pulse[2]. This means that whenever the beginning of a keyboard gate pulse appears, the LFO is forced to start its wave-generating cycle from the beginning point, as shown in Fig. 2-4 (b). In terms of the trill, this means that whenever a key is pressed, the trill will always start with the upper of the two notes and the first note will be just as long as the other trill notes. Since the gate pulse triggers the function, it will occur only on the first note in a legato passage.

Fig. 2-4 Trill

ⓐ Patch Diagram (Block diagram is essentially same as Fig. 2-2 ⓐ).



2. KYBD TRIG = keyboard trigger. The name comes from the fact that this phase-lock function is *triggered* by the gate pulse.

For tuning the trill, set the LFO at its lowest frequency (FREQ RANGE at "L", FREQ slider at "0"). With the LFO MOD IN to the VCO at "0", tune the VCO in the normal way. To tune the trill to a major third, for example, strike a key and note the pitch. Next, strike the key which is a major third below the first key and raise the LFO MOD IN control of the VCO until the pitch matches the pitch noted before. Striking a key forces the LFO square wave to start over again at its highest point and with the FREQ RANGE switch at "L", this high part of the wave should remain long enough to set the MOD IN level. (If not, strike the key again). Remove the MOD IN patch cord and check the pitch by striking the original key again.

As an experiment, try a trill at a major third, then change the LFO OUTPUT LEVEL switch to "x1" and try the other LFO waveforms.

Fig. 2-5 shows a patch in which the frequency of one VCO is used to modulate (control) the frequency of a second VCO. The result is a very complex waveform which contains overtones normally not available from the output of a VCO. These non-harmonic overtones can be very useful for producing metallic clanging sounds such as those produced by bells and metal bars.

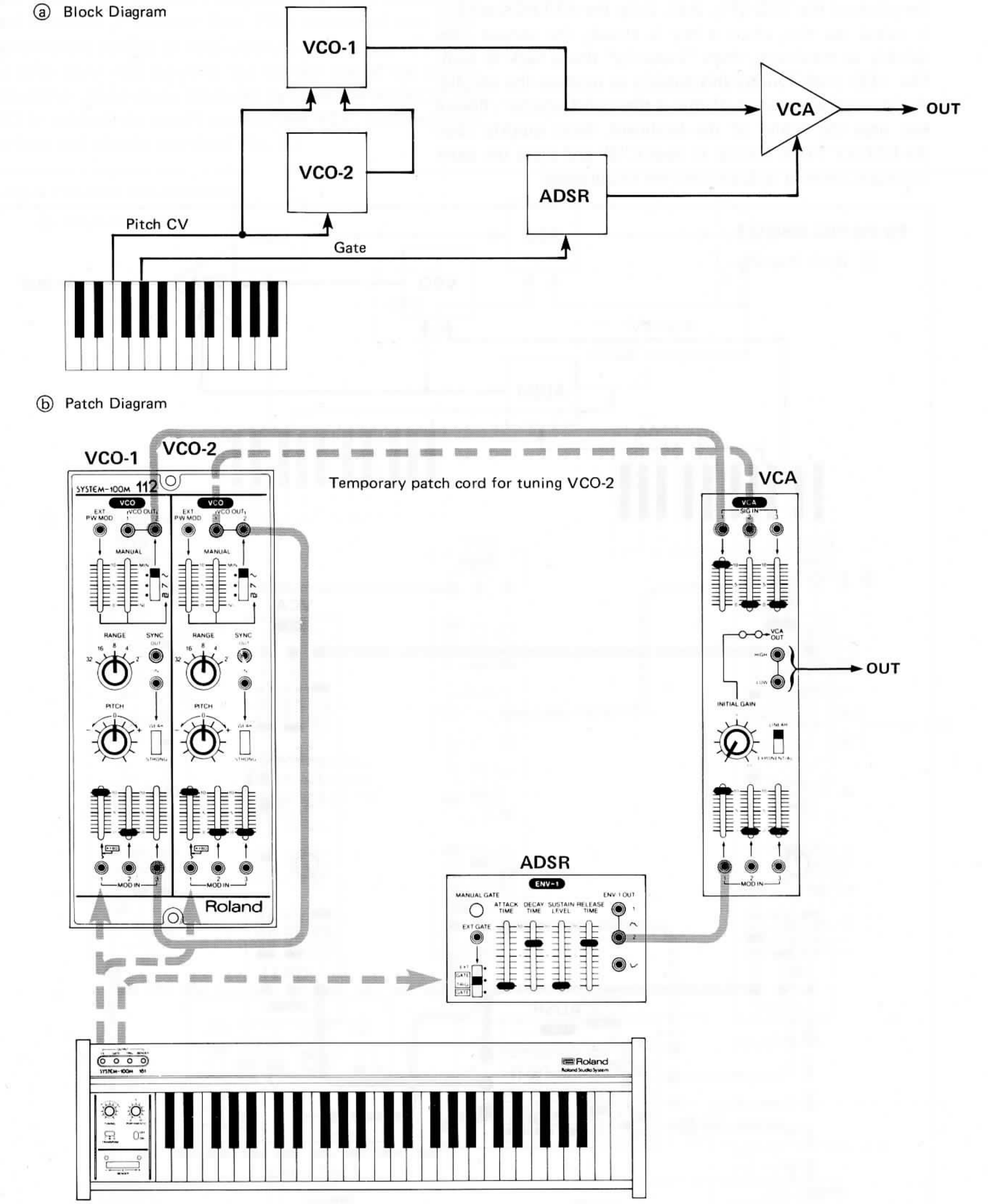
To experiment with this type of frequency modulation, start by lowering to "0" the MOD IN which feeds the output of VCO-2 to VCO-1 (Fig. 2-5). Tune VCO-1 to the desired pitch relation with the keyboard. Raise the VCO-2 SIG IN to the VCA and tune VCO-2 to unison with VCO-1, then return the control to "0". Next, while tapping a key near the center of the keyboard, slowly raise the VCO-1 MOD IN control and note the changes which take place. Above level "7", raise the control very slowly because the changes occur extremely rapidly. The most important point to note is that the apparent pitch of the sound does not remain stable but rises as the MOD IN control is raised. Starting with the MOD IN control at "0" again, play a short five-finger scale passage over and over while slowly raising the MOD IN control. Note that although the tuning changes, the overall pitch relations between the notes remain more or less correct. Try this experiment again with both VCO's set at 2' and note that the pitch relations will sometimes be different, especially if played in the higher range of the keyboard. This change in tuning should be kept in mind if it is desired to use this type of arrangement to play melodies.

Tuning may best be accomplished by first ignoring the tuning and concentrating on producing the desired sound. Once the desired sound is found, it is simply a matter of determining what the tuning discrepancy is and correcting it. For example, if the desired sound produces pitches a minor third higher than the correct pitches, tune the VCO's a minor third lower than normal. VCO-1 may be tuned by removing the patch cord to its MOD IN jack. In this way, the delicate setting of the MOD IN slider will not have to be altered.

Fig. 2-5 Frequency Modulation (by VCO)

ⓐ Block Diagram

ⓑ Patch Diagram

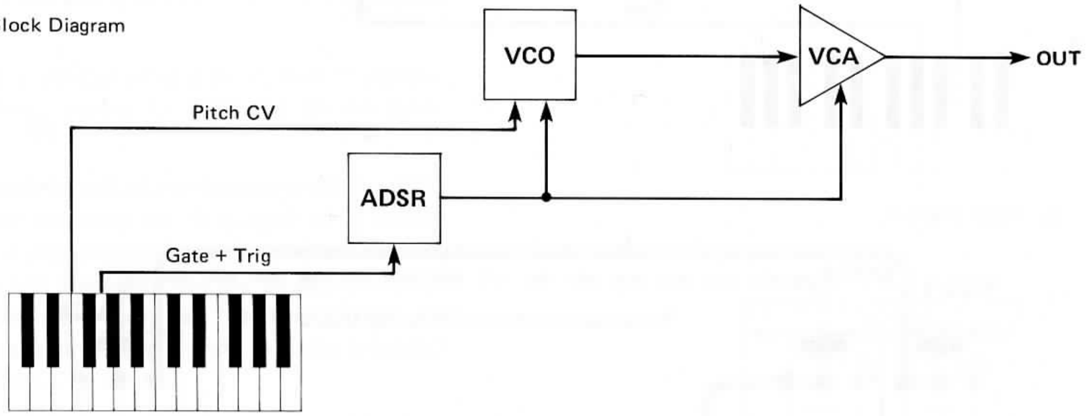


The most obvious variations on this particular sound would be to try different VCO waveforms and tuning the VCO's to intervals other than unison. Another variation would be to leave the VCO-2 SIG IN to the VCA raised. A VCF could also be used to alter the tone color.

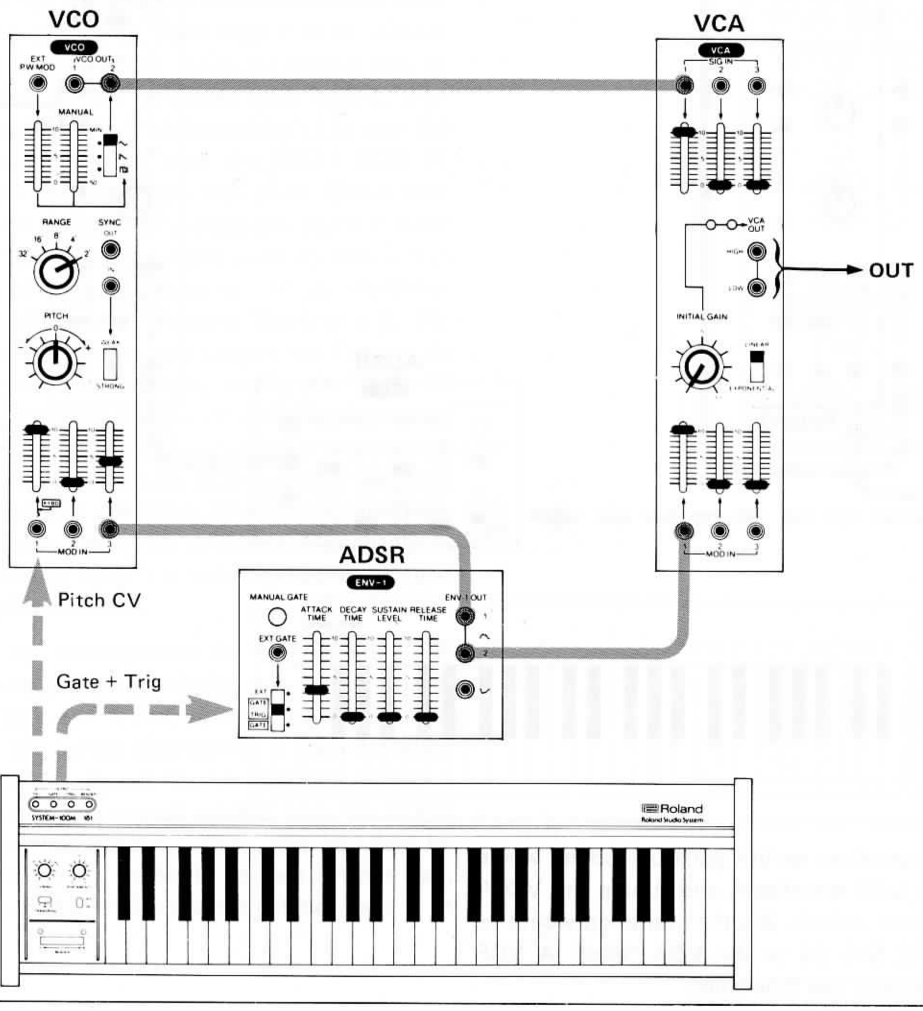
One "practical" application of the synthesizer is to use it to whistle for the dog, using the envelope generator to control the pitch of the VCO (Fig. 2-6). Only the ATTACK control is raised so that when a key is struck, the output rises quickly to maximum, then "instantly" drops back to zero. The VCO pitch follows this pattern to produce the whistle. An interesting variation of this is the "wolf whistle". Press a key near the center of the keyboard. Next quickly raise the DECAY TIME control to about "2" and press the same key again, holding it down until the sound stops.

Fig. 2-6 DOG WHISTLE

ⓐ Block Diagram



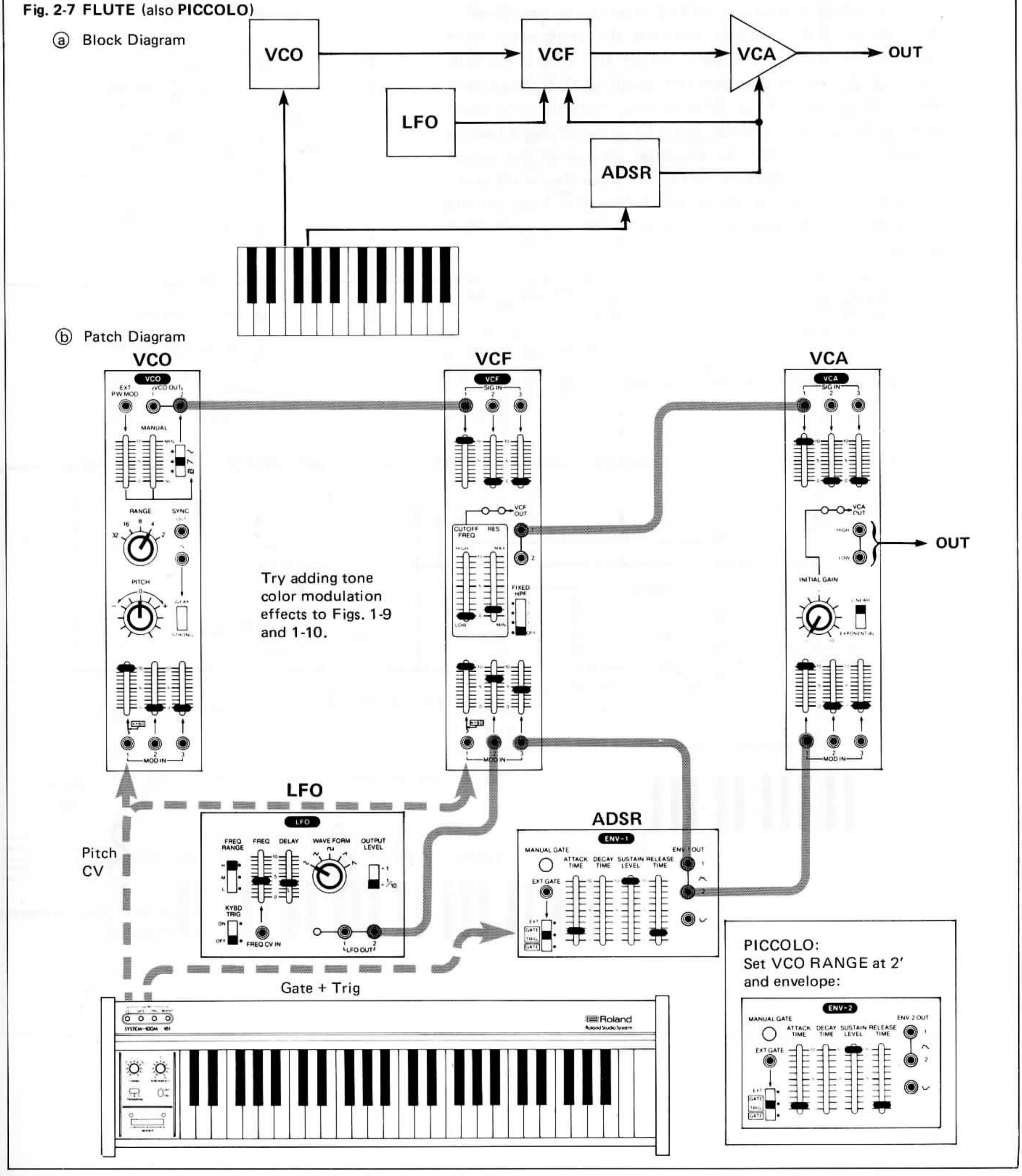
ⓑ Patch Diagram



2-3 Tone Color Modulation

Two of the most common forms of tone color modulation were discussed in Chapter One: Pitch control of tone color and envelope control of tone color. In many types of sound, the tone color will waver at the vibrato rate of the sound. This effect can be easily produced by using the output of the LFO to control the cutoff point of the VCF, as shown with the flute and piccolo sounds of Fig. 2-7.

Fig. 2-7 FLUTE (also **PICCOLO**)

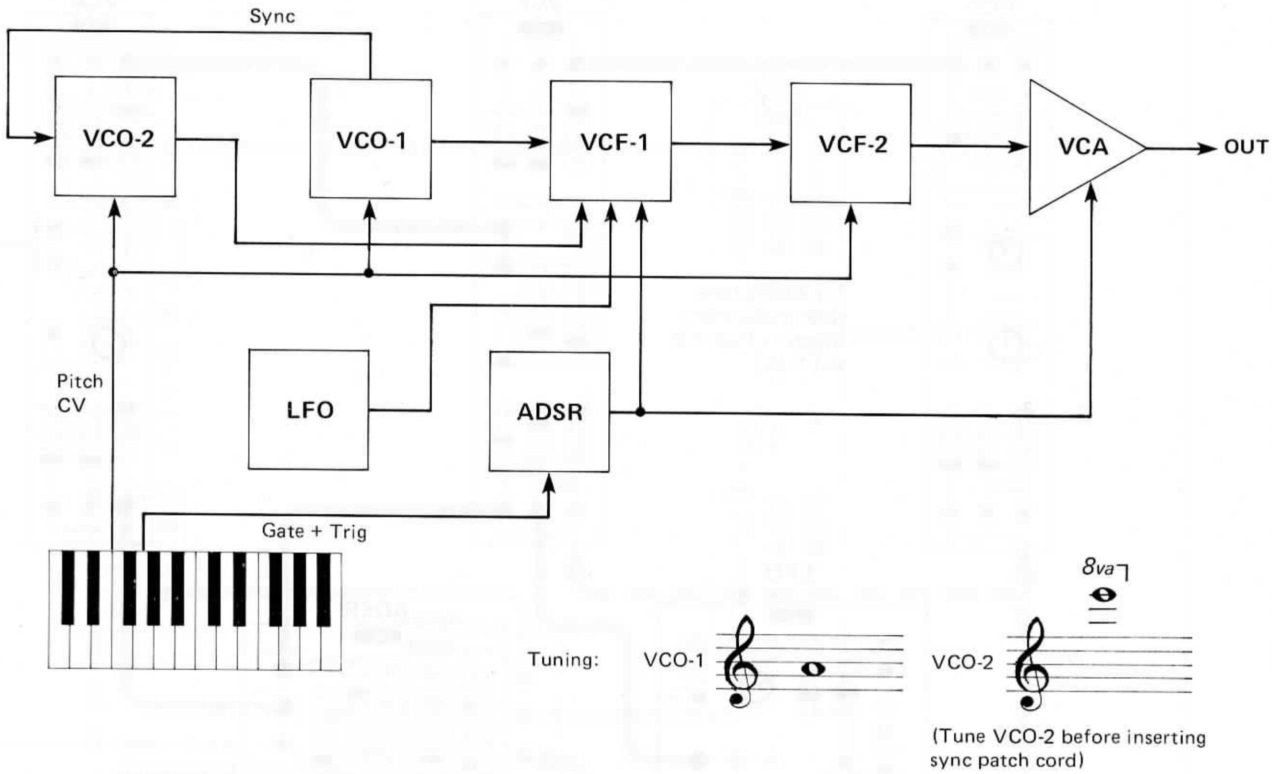


LFO modulation of the VCF cutoff point is called **growl**, although this term is more often associated with the effect produced by exaggerated LFO control of tone color. This effect can be heard with the flute/piccolo patch of Fig. 2-7 by raising the LFO FREQ slider to "10" and changing the LFO OUTPUT LEVEL switch to "x1".

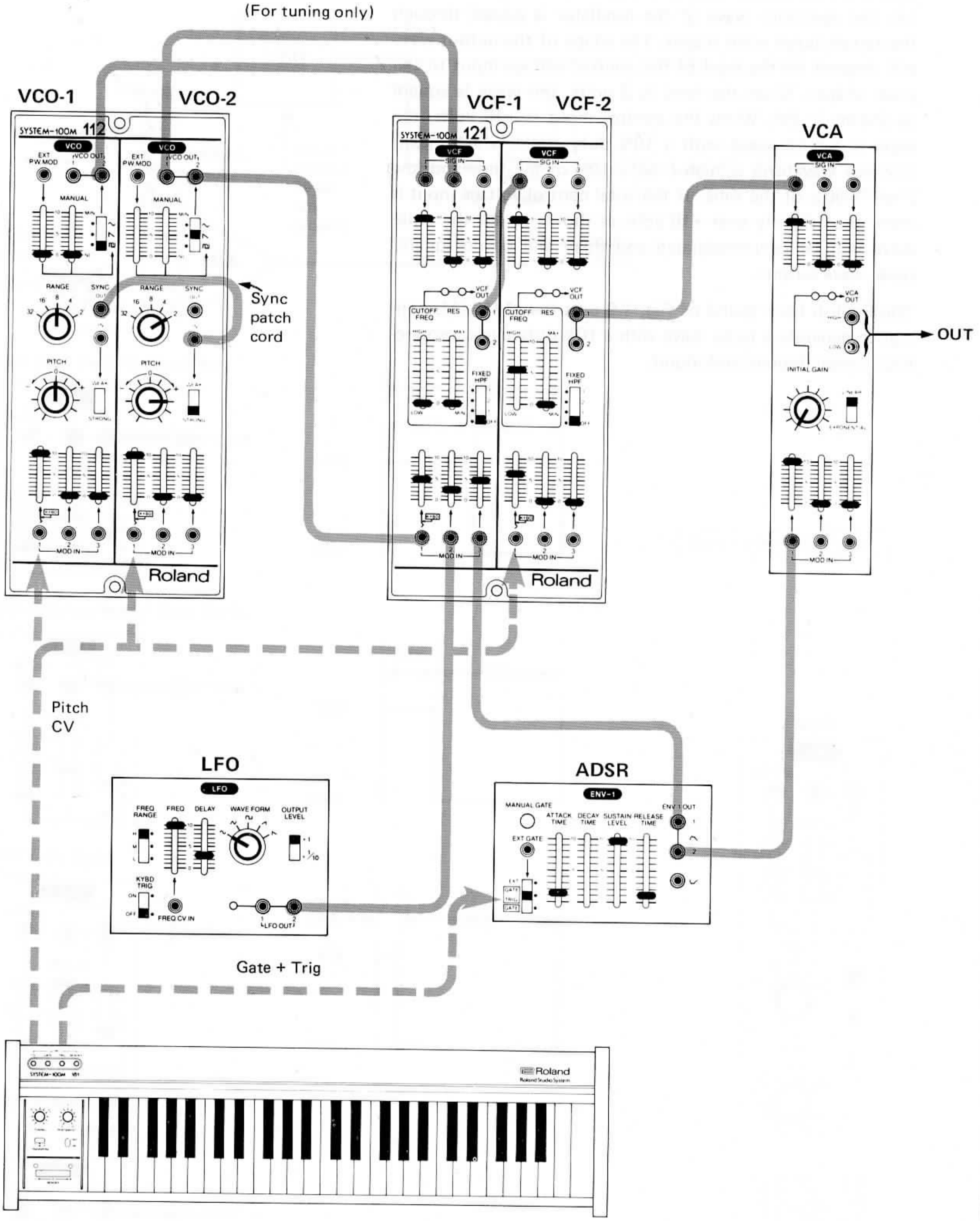
Fig. 2-8 shows a clarinet sound which uses a second VCO to modulate the cutoff point of a VCF. Tune VCO-1 to the desired pitch relation with the keyboard. Raise the VCO-2 SIG IN to VCF-1 and tune VCO-2 two octaves and a perfect fifth above VCO-1 before inserting the sync patch cord between the VCO's. It is easier to set the VCO-2 RANGE switch to 8', tune to a perfect fifth above VCO-1, then return the RANGE switch to 8'. After tuning, insert the sync patch cord. When a key is pressed, the VCF-1 cutoff point rises to a level determined by the envelope control of the cutoff point. The VCO-2 MOD IN to VCF-1 causes the cutoff point to waver very rapidly above and below this level, adding harmonics which were not present in the original VCO-1 square wave.

Fig. 2-8 CLARINET (VCF Modulation by VCO)

ⓐ Block Diagram



ⓑ Patch Diagram

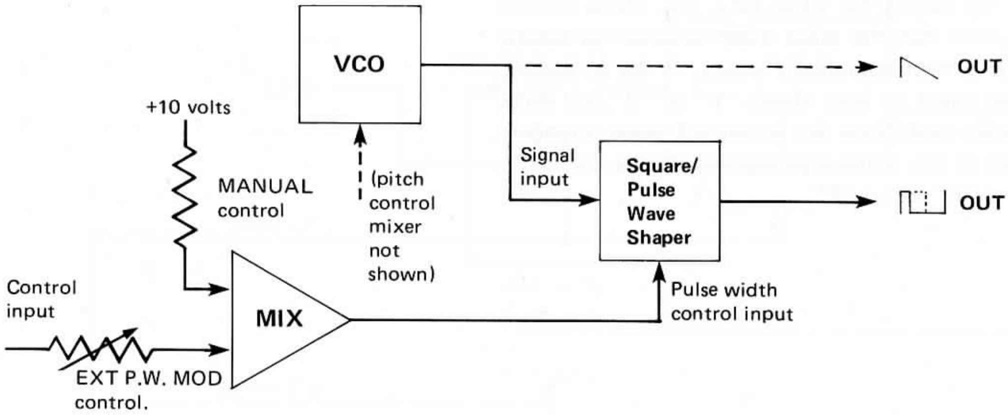


Another form of tone color modulation can be produced with the VCO pulse wave (⊓⊔) output. Fig. 2-9 shows how the VCO square wave and pulse wave are generated. In (a), the sawtooth wave of the oscillator is passed through the square/pulse wave shaper. The shape of the output wave will depend on the level of the control voltage input to the wave shaper. When the level is 0 volts, the wave is square as shown in (b). When the control input is +10 volts, the wave is a pulse wave with a 10% duty cycle, which means that the waveform is high ("on") 10% of the time and low ("off") 90% of the time. If the total control voltage input is more than slightly over +10 volts as shown in (e), the pulse wave will be over-modulated and there will be no square/pulse wave output.

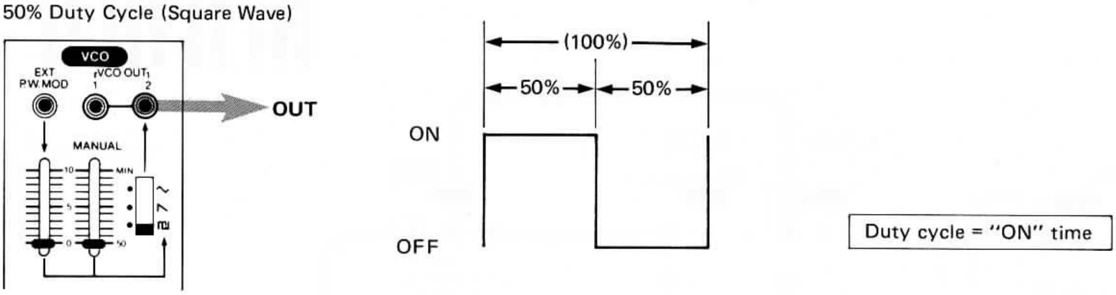
The English horn sound of Fig. 1-9 uses the MANUAL control to generate a pulse wave with a 10% duty cycle, producing its nasal double reed sound.

Fig. 2-9 Square and Pulse Waves

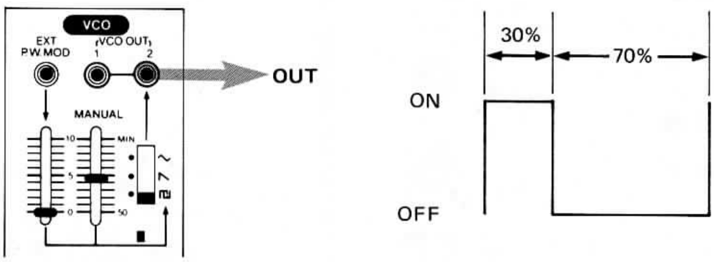
(a) Square/Pulse Wave Block Diagram



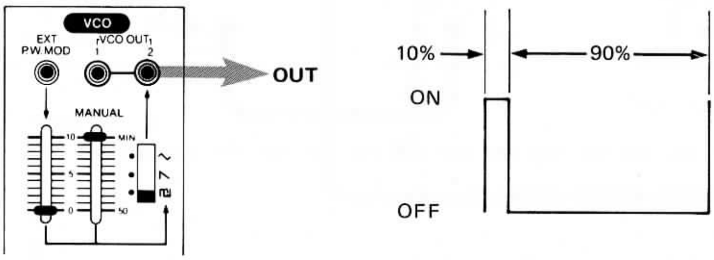
(b) 50% Duty Cycle (Square Wave)



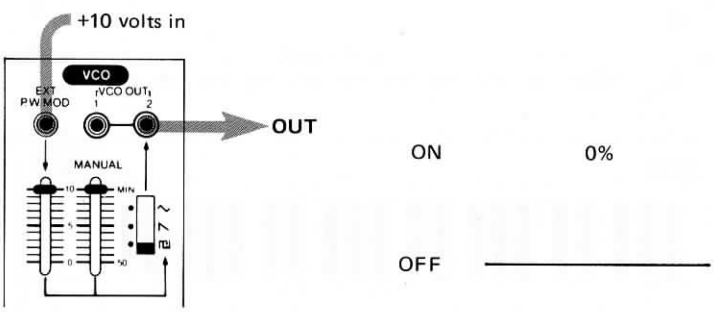
(c) 30% (approximately) Duty Cycle



(d) 10% Duty Cycle



(e) Over-modulation



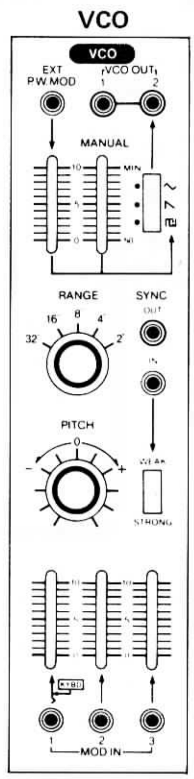


Fig. 2-10 shows a bowed string bass sound which uses the envelope generator to modulate (change) the pulse wave during the progression of each note played. While holding down a key, try raising the VCO EXT P.W. MOD control to "10" and note that the pulse width becomes so narrow that the sound almost completely ceases. If the MANUAL control is now raised to only about "1" or "2", the pulse wave will be over-modulated and sound will cease. Compare the differences in the string bass sound with the EXT P.W. MOD control at "0" and at "4".

Fig. 2-10 BOWED STRING BASS

ⓐ Block Diagram

ⓑ Patch Diagram

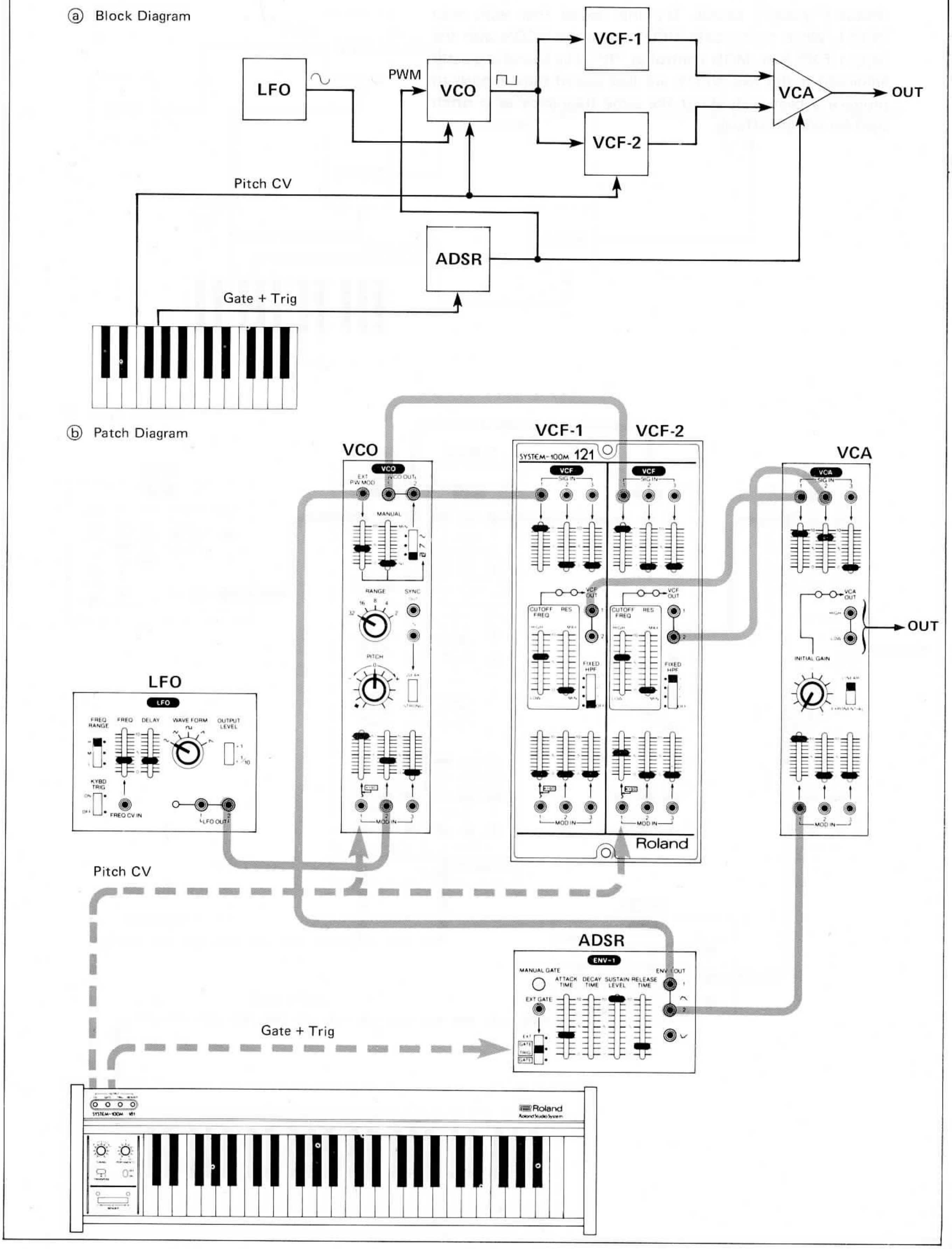
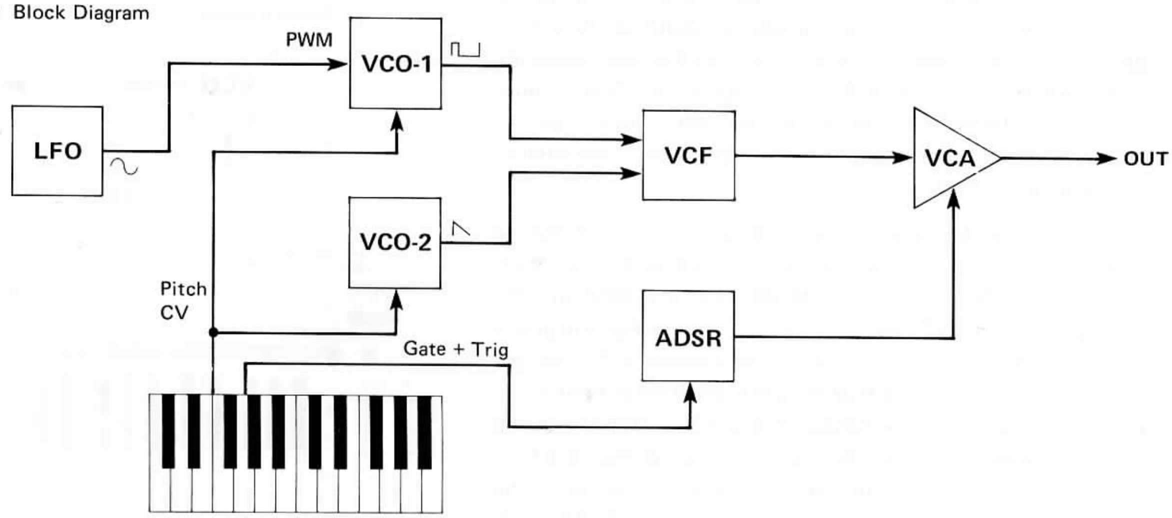


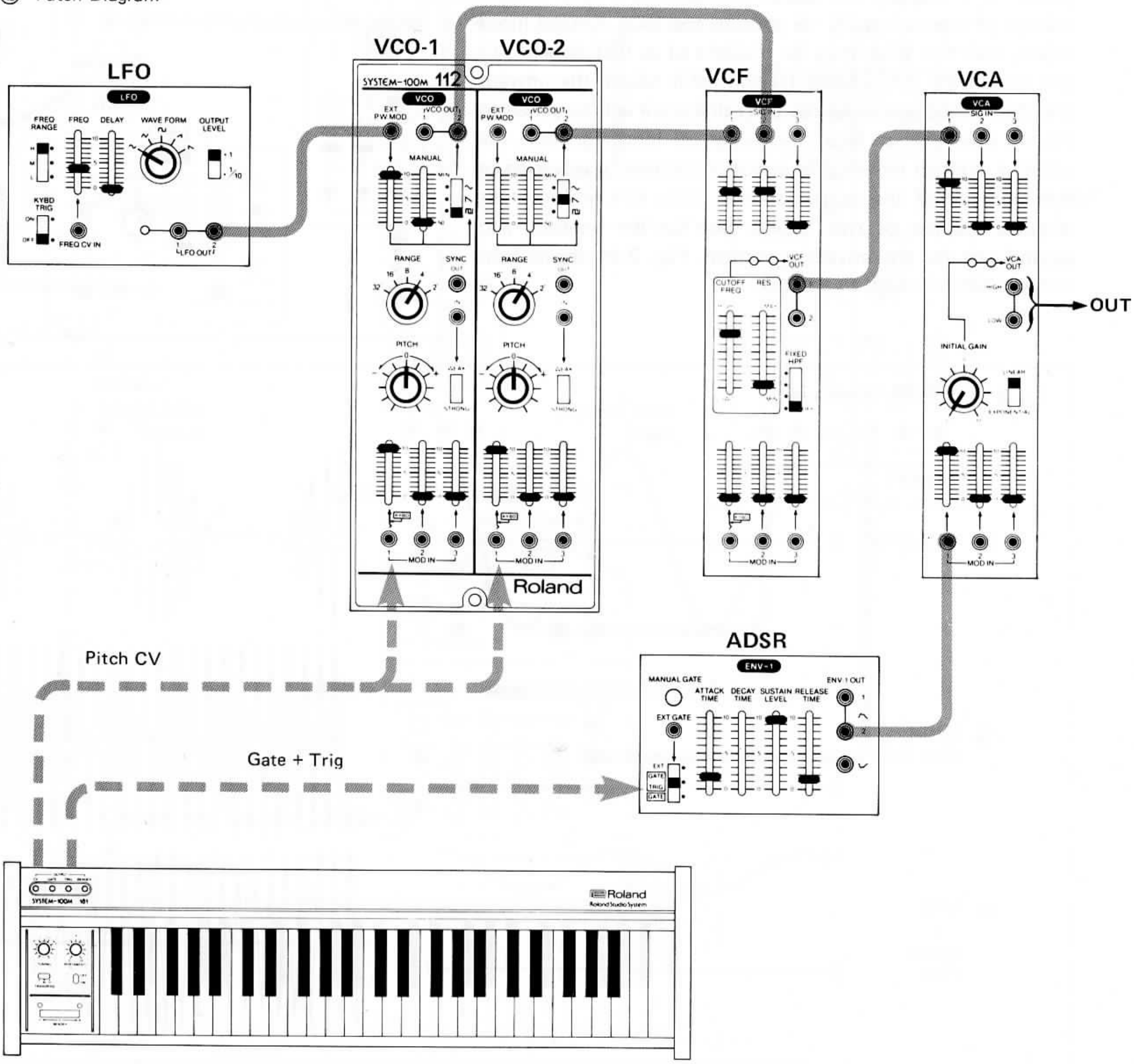
Fig. 2-11 shows a string ensemble sound which uses LFO modulation of pulse width to produce an ensemble or chorus ("group") sound. Try this sound first with only VCO-1. When trying both VCO's, tune the VCO's with the VCO-1 EXT P.W. MOD control at "0". The sound is greatly enhanced if the two VCO's are just out of tune enough to produce a beat with about the same frequency as is often used for vibrato effects.

Fig. 2-11 STRING ENSEMBLE

ⓐ Block Diagram



ⓑ Patch Diagram



2-4 Amplitude Modulation

The control of a VCA by an envelope generator is by far the most common form of **amplitude modulation** found in electronic music. Less common is the use of a low frequency sine wave to control a VCA to produce a wavering of loudness called **tremolo**. Tremolo, growl, and vibrato are all closely related; indeed, tremolo is sometimes considered a special form of vibrato.

Fig. 2-12 shows block and patch diagrams for demonstrating how tremolo effects are produced. Start with the VCA INITIAL GAIN and the LFO MOD IN to the VCA at "0". Slowly raise the LFO MOD IN control and notice the effect on the sound. Technically, VCA's are known as 2-quadrant multipliers which means that they are sensitive to control inputs above zero, but not below. Fig. 2-13 (a) shows that the LFO sine wave output varies above and below the zero line. Since the VCA is sensitive only to levels above zero, the VCA will "open" only half of the time. Lower the MOD IN to the VCA and raise the VCA INITIAL GAIN control to about "5". This has the effect of inputting a fixed control voltage of approximately +5 volts to the VCA control input mixer, and the VCA may be thought of as half "open". In this condition, if the MOD IN control is raised, the upward and downward swings of the LFO sine wave will be added to the +5 volt level so that the swings no longer go into the negative control region. The result is tremolo, a wavering in the loudness of the output sound. Using this arrangement requires the use of two VCA's: one for the tremolo, and second one for the envelope control. Fig. 2-14 shows how tremolo can be added to a patch.

Fig. 2-12 Tremolo experiment

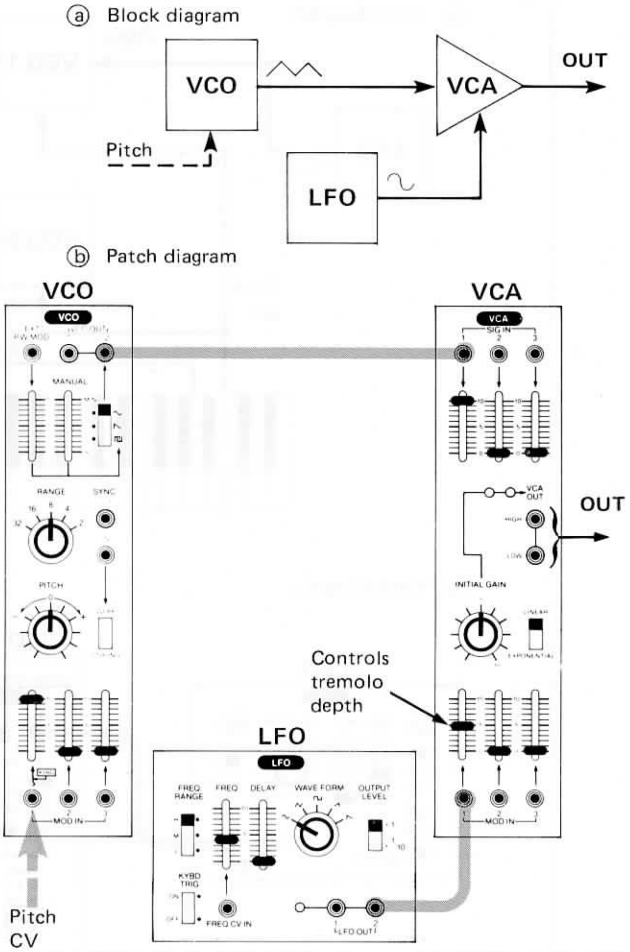


Fig. 2-13 Tremolo Waveforms

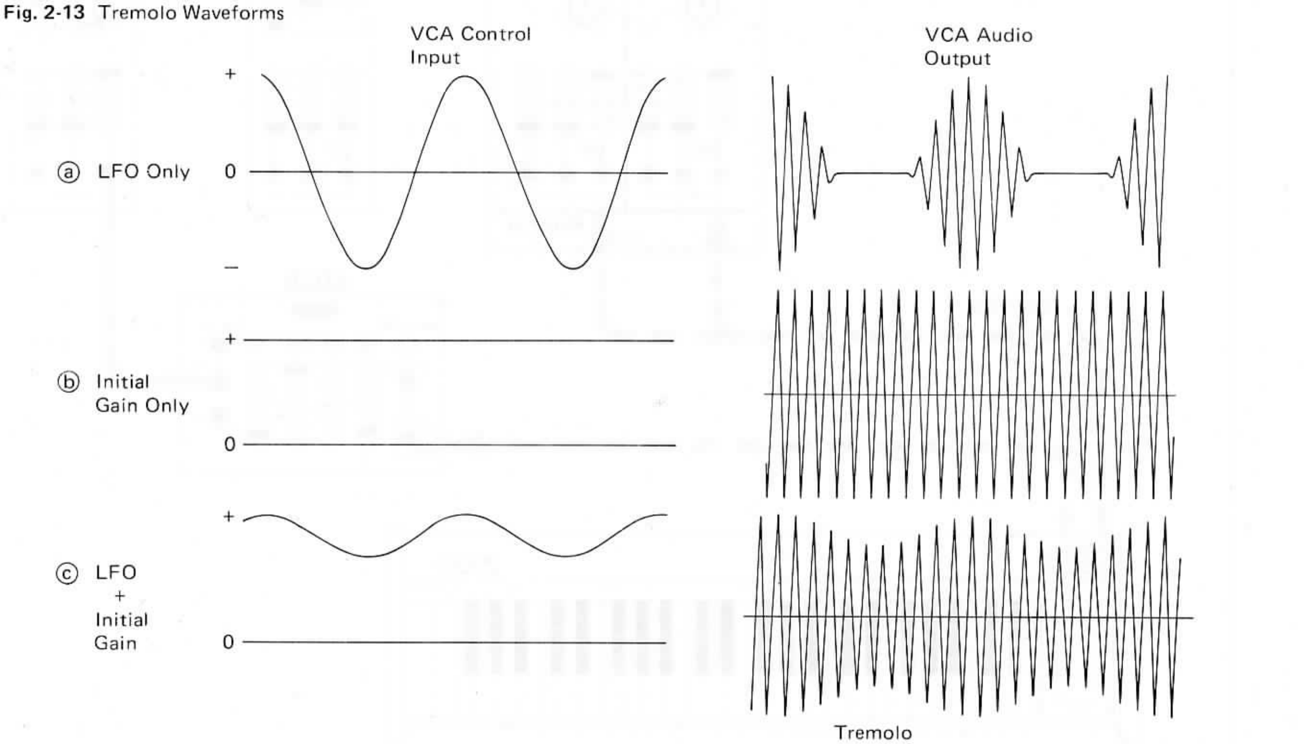
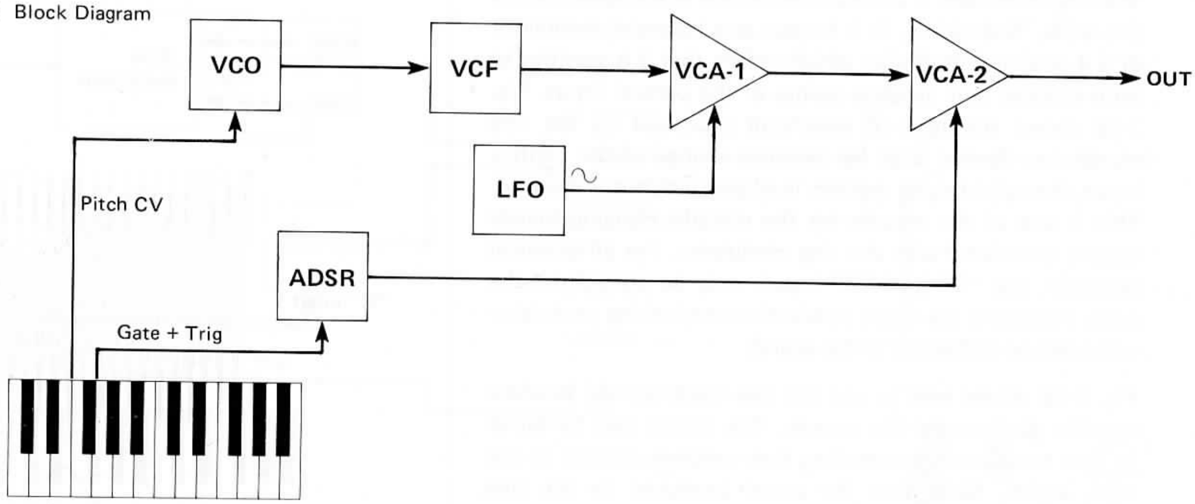
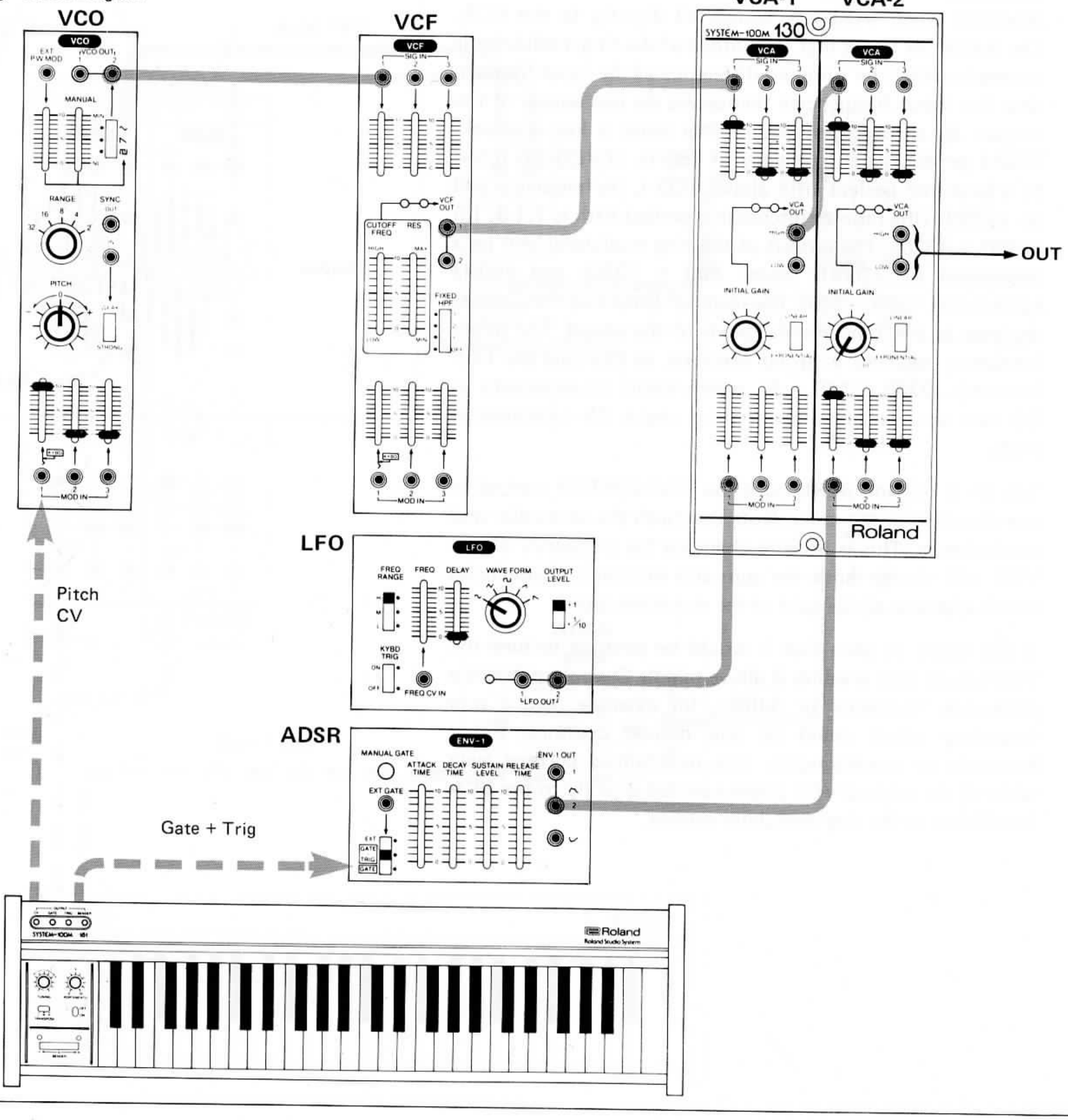


Fig. 2-14 Adding Tremolo to a Patch

ⓐ Block Diagram



ⓑ Patch Diagram



2-5 The Ring Modulator

The **ring modulator** is a circuit which falls in the same class as the VCA. Technically, it is known as a balanced modulator or a 4-quadrant multiplier which means that it is sensitive to both positive and negative swings of the control input. Fig. 2-15 shows the type of waveform produced by the ring modulator. Notice that for negative swings of the control input, the output signal reverses in phase so that it is inverted. This is one of the reasons for the metallic clanging sounds usually associated with the ring modulator. For all practical purposes, the "X" and "Y" inputs may be considered the same. Reversing the input connections to the ring modulator will make no difference in the sound.

Fig. 2-16 shows how to use the ring modulator to produce metallic glockenspiel-like sounds. The VCO's may be tuned by first temporarily connecting their outputs directly to the VCA inputs. Note that the sound produced by the ring modulator in this patch is one octave below the sound produced when VCO-1 is connected directly to the VCA. The reason for this is that the output of the ring modulator is a compound of the sum and difference of the input frequencies; the input frequencies themselves do not appear at the output. As an example, assume that when a key is struck, VCO-1 produces the frequency of 880Hz. If VCO-2 is tuned to a beat-free perfect fifth above VCO-1, its frequency will be 1320Hz (the ratio of pitches in a perfect fifth is 1:1.5; 1.5 x 880 = 1320). The output of the ring modulator will be a compound of 2200Hz (sum: 880 + 1320) and 440Hz (difference: 1320 – 880). The lower of these two frequencies we hear as the fundamental pitch of the sound. The upper frequency becomes a strong overtone, in this case the fifth harmonic (2200 ÷ 440 = 5), which would be equivalent to C# two octaves and a major third above the fundamental pitch.

Tap on a key and slowly turn the VCO-2 PITCH control in one direction or the other. Note that both the tone color and pitch change. This is because changing the frequency of one VCO will change both the sum and difference frequencies which appear at the output of the ring modulator.

It can easily be seen that it would be possible to tune the VCO's such that pressing A above middle C would produce a difference frequency of 440Hz, for example, and a sum frequency which could be any desired overtone, either harmonic or non-harmonic. The possibilities become even wider if the original VCO pitches are fed directly to the VCA in addition to the ring modulator output.

Fig. 2-15 Ring Modulator Waveforms

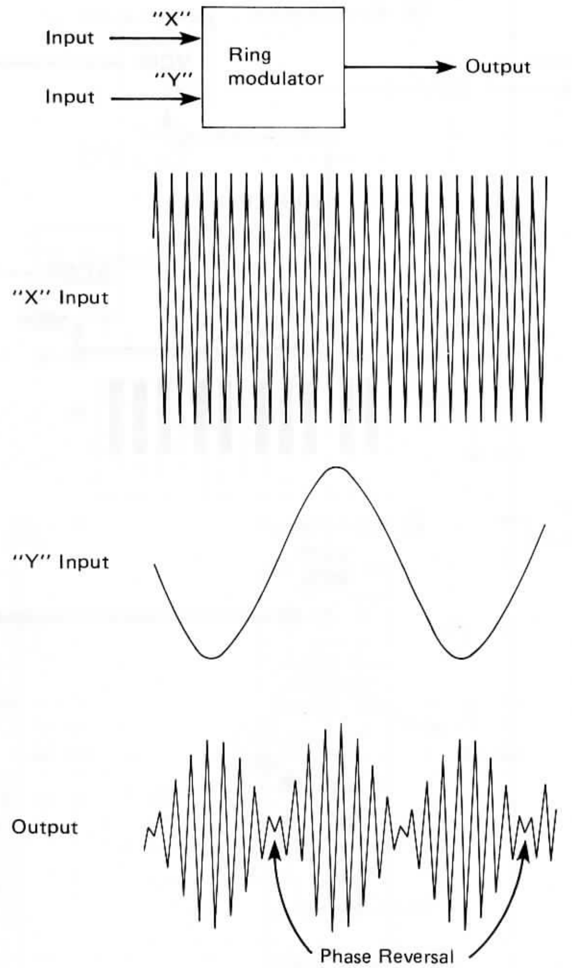
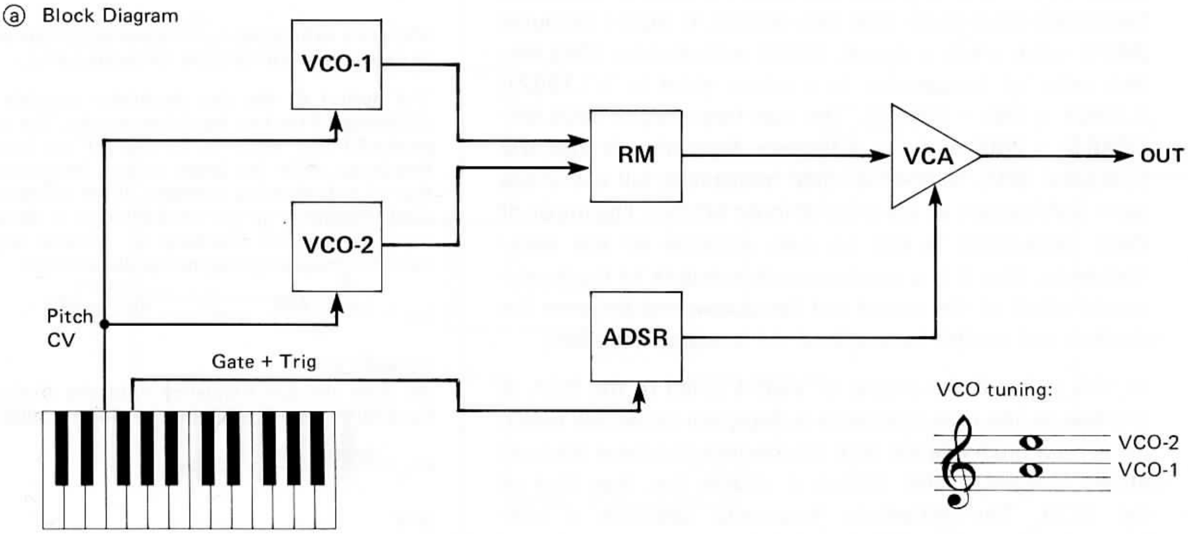
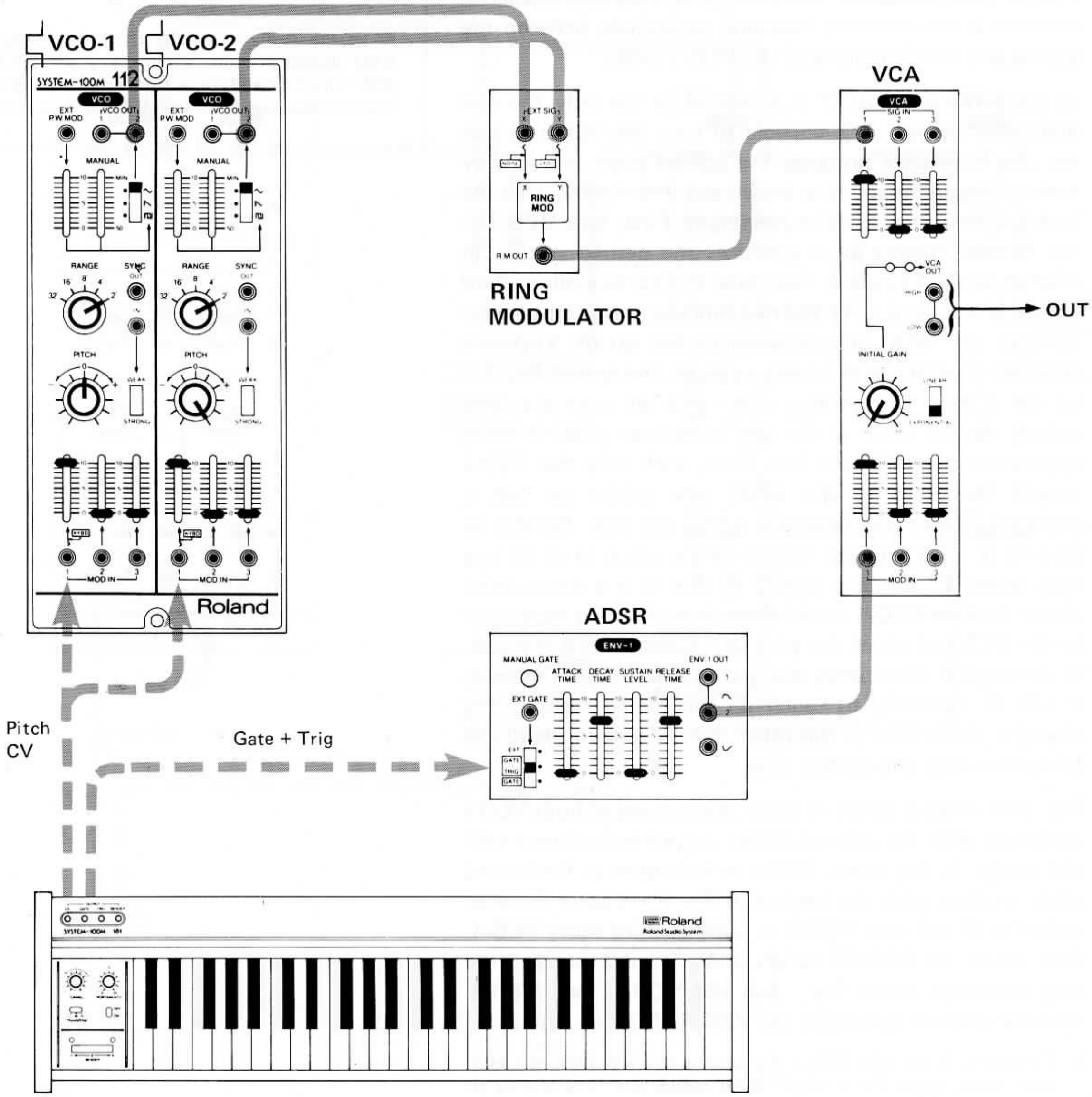


Fig. 2-16 GLOCKENSPIEL

ⓐ Block Diagram



ⓑ Patch Diagram



Try the patch of Fig. 2-16 with VCO-2 tuned a minor third above VCO-1[3]. In this case, the apparent pitch of the sound has moved down more than two octaves. If VCO-1 produces 880Hz when a key is struck, VCO-2 will produce 1046.5Hz (the ratio of frequencies in a minor third is 1:1.18921; 1.18921 x 880 = 1046.5). The sum frequency is 1926.5Hz (1046.5 + 880) and the difference frequency is 166.5Hz (1046.5 – 880). Neither of these frequencies fall within the same scale system as the original input pitches. The upper of these frequencies is not an even multiple of the lower frequency, thus it is a non-harmonic overtone of the fundamental pitch of the sound and lies somewhere between the eleventh and twelfth harmonic of the fundamental pitch.

At this point, if the output of VCO-1 is fed to the VCA in addition to the ring modulator output, the sound will regain the correct pitch relation with the keyboard because the level of the ring modulator output is slightly less than that of the VCO. The difference frequency becomes a non-harmonic frequency which lies slightly over two octaves and a major third below the fundamental, and the sum frequency becomes a non-harmonic overtone somewhere between the second and third harmonic of the VCO-1 pitch.

In the above example, if it is desired to use only the ring modulator output, it is possible to tune the VCO's so that the ring modulator produces the correct pitch relations by first setting up the sound as shown and then finding what the tuning discrepancy is and correcting it. First, tune VCO-1 in the normal manner so it produces the desired pitches in relation to the keyboard; then tune VCO-2 to a minor third above. Next, using only the ring modulator output connections to the VCA, determine which key on the keyboard produces the pitch of *A*. In this example, the nearest key will be the *D* key. Disregarding pitch shifts of more than one octave, we can think of the ring modulator pitch as being approximatly a fifth too low. Next, with only the VCO-1 output connected to the VCA, tune VCO-1 so that it produces pitches approximately a fifth too high. Do this by striking an *E* and tuning VCO-1 to the pitch of *A* (*A* is a fifth above *E*). Re-tune VCO-2 so that it is a minor third above the new VCO-1 pitch. Re-connect the ring modulator to the VCA and check the pitch produced when A is struck. If the original discrepancy was not an exact musical interval, it will be necessary to touch up the tuning slightly. Try changing either one or the other PITCH control slightly to bring the sound into perfect tune.

Fig. 2-17 shows a patch in which the outputs of both VCO's are mixed with the ring modulator output to produce a bell-like sound. In this patch, VCO-1 is first tuned to the desired pitch relation with the keyboard. Set the VCO-2 RANGE switch at 8' and tune VCO-2 to a minor third above VCO-1, then return the RANGE switch to 4'. Try the sound of the ring modulator alone first, then add VCO-1, then VCO-2. Also try different settings of the VCO RANGE switches.

3. First press A and tune VCO-1 to unison with a test pitch or tuning fork. Next, press F# a minor third below and tune VCO-2 to unison with the test pitch. Now when A is pressed, VCO-1 will be at unison with the test pitch and VCO-2 will be a minor third above.

CALCULATING RING MODULATOR INPUT FREQUENCIES

Musicians with access to a frequency counter and an inclination to experiment will find the following useful.

The output of the ring modulator consists of the sum and difference of the two input frequencies. The pitch of the sound produced will seem to be that of the lower or difference frequency while the upper or sum frequency will seem to be that of a dominating overtone. If we assume the difference or pitch frequency to be the 440Hz of A above middle C, the input frequencies necessary to produce any given sum or overtone frequency may be calculated with:

$$In_1 = \frac{\text{sum} + 440}{2} \text{ ; and } In_2 = \frac{\text{sum} - 440}{2}$$

EXAMPLE:
We want the sum frequency to be the ninth harmonic which, for 440Hz, would be 3960Hz (9 x 440 = 3960); therefore:

$$In_1 = \frac{3960 + 440}{2} = 2200\text{Hz};$$

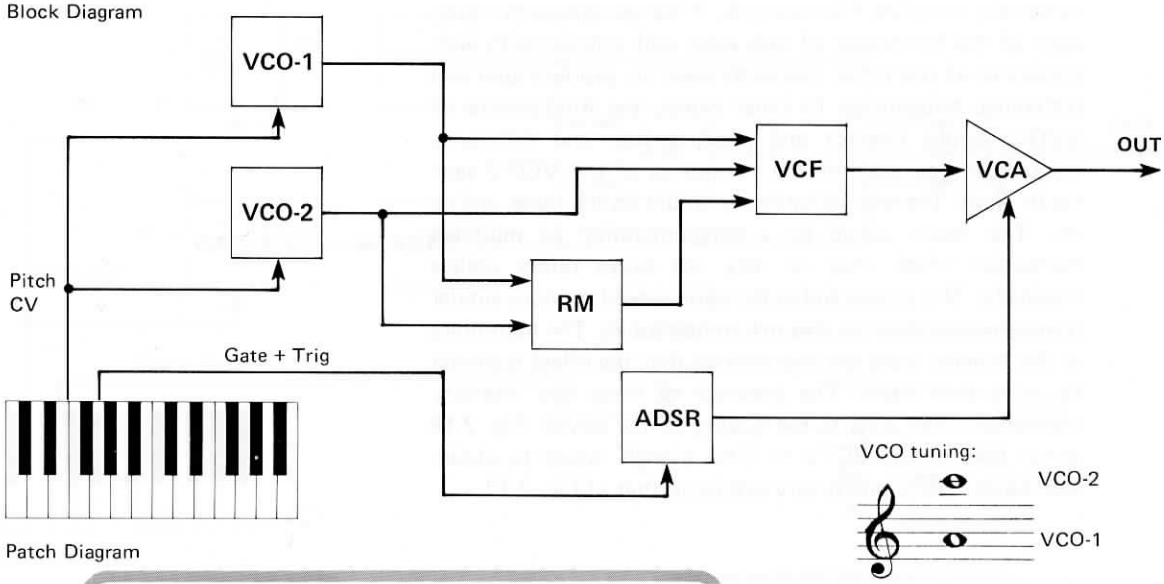
and:

$$In_2 = \frac{3960 - 440}{2} = 1760\text{Hz}.$$

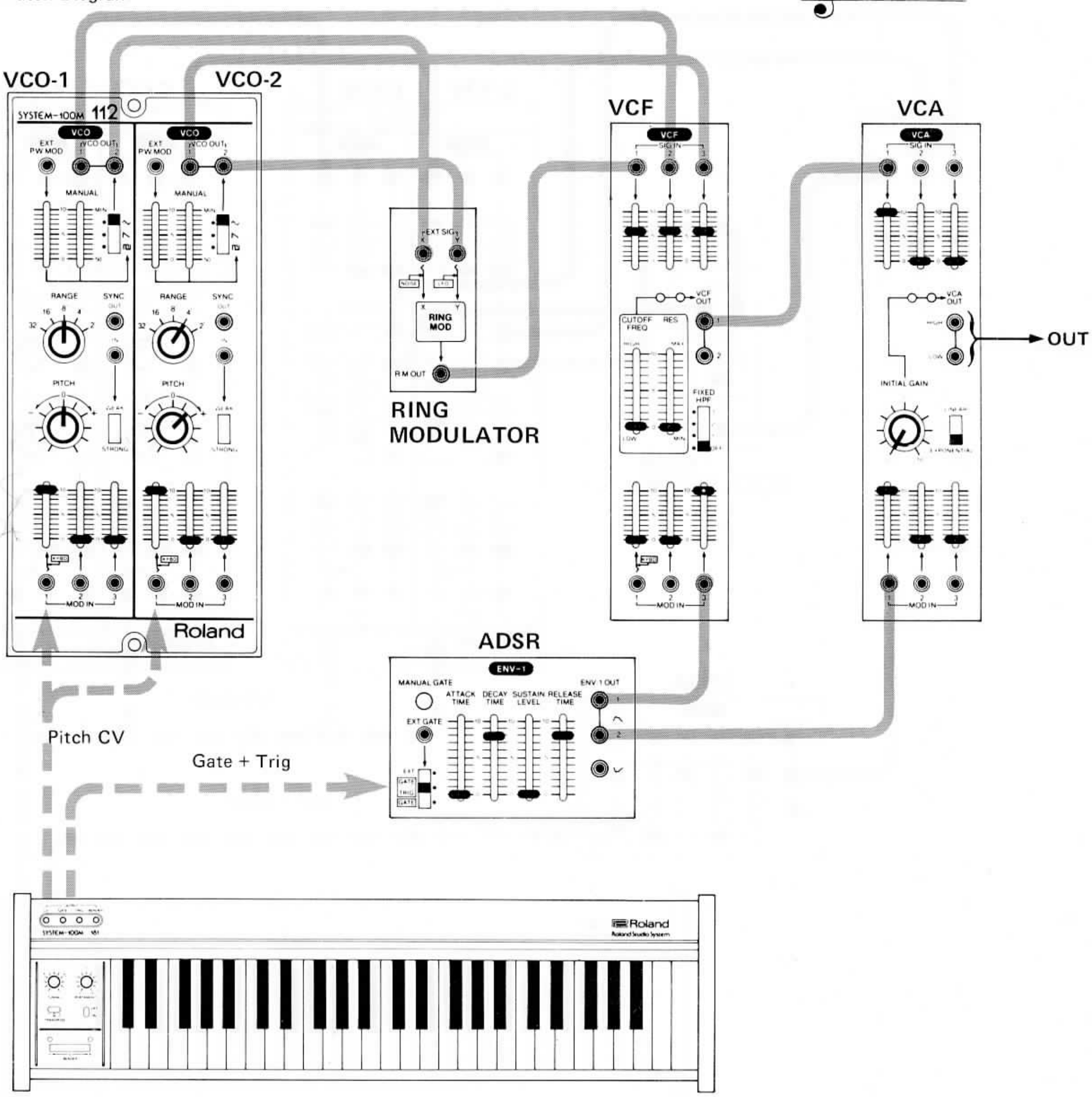
When the A above middle C on the keyboard is struck, one VCO should produce a frequency of 2200Hz and the second VCO should produce a frequency of 1760Hz. The apparent pitch of the sound will be 440Hz with a harmonic of 3960Hz.

Fig. 2-17 BELL

ⓐ Block Diagram



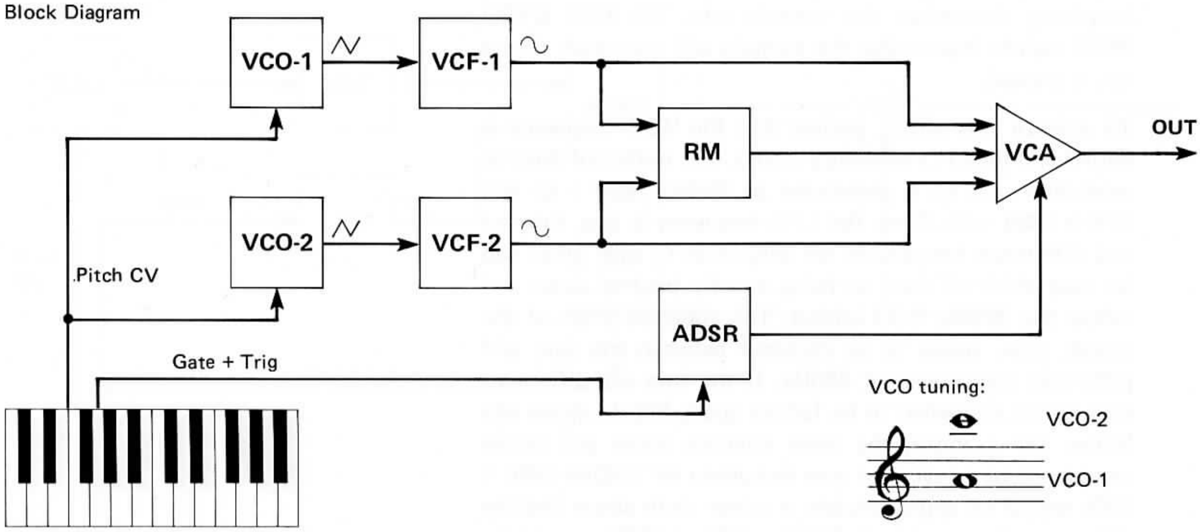
ⓑ Patch Diagram



If waveforms other than sine waves or triangle waves are used, the spectrum of the ring modulator output becomes extremely complex. For example, if we use sawtooth waves, each of the harmonics of one wave will interact with each harmonic of the other sawtooth wave to produce sum and difference frequencies. In other words, the fundamental of VCO-1 would interact and produce sum and difference frequencies for each of the harmonics of the VCO-2 sawtooth wave. The second harmonic would do the same, and so on. The result would be a conglomeration of muddled harmonics which may or may not prove rather useless musically. The square and pulse waves would produce similar results because they are also rich in harmonics. The harmonics in the triangle wave are low enough that the effect is similar to using sine waves. The presence of these low intensity harmonics often adds to the quality of the sound. Fig. 2-18 shows how to use VCF's to filter triangle waves to obtain sine waves. The sound is very similar to that of Fig. 2-17.

Fig. 2-18 BELL Using Sine Waves

ⓐ Block Diagram



ⓑ Patch Diagram

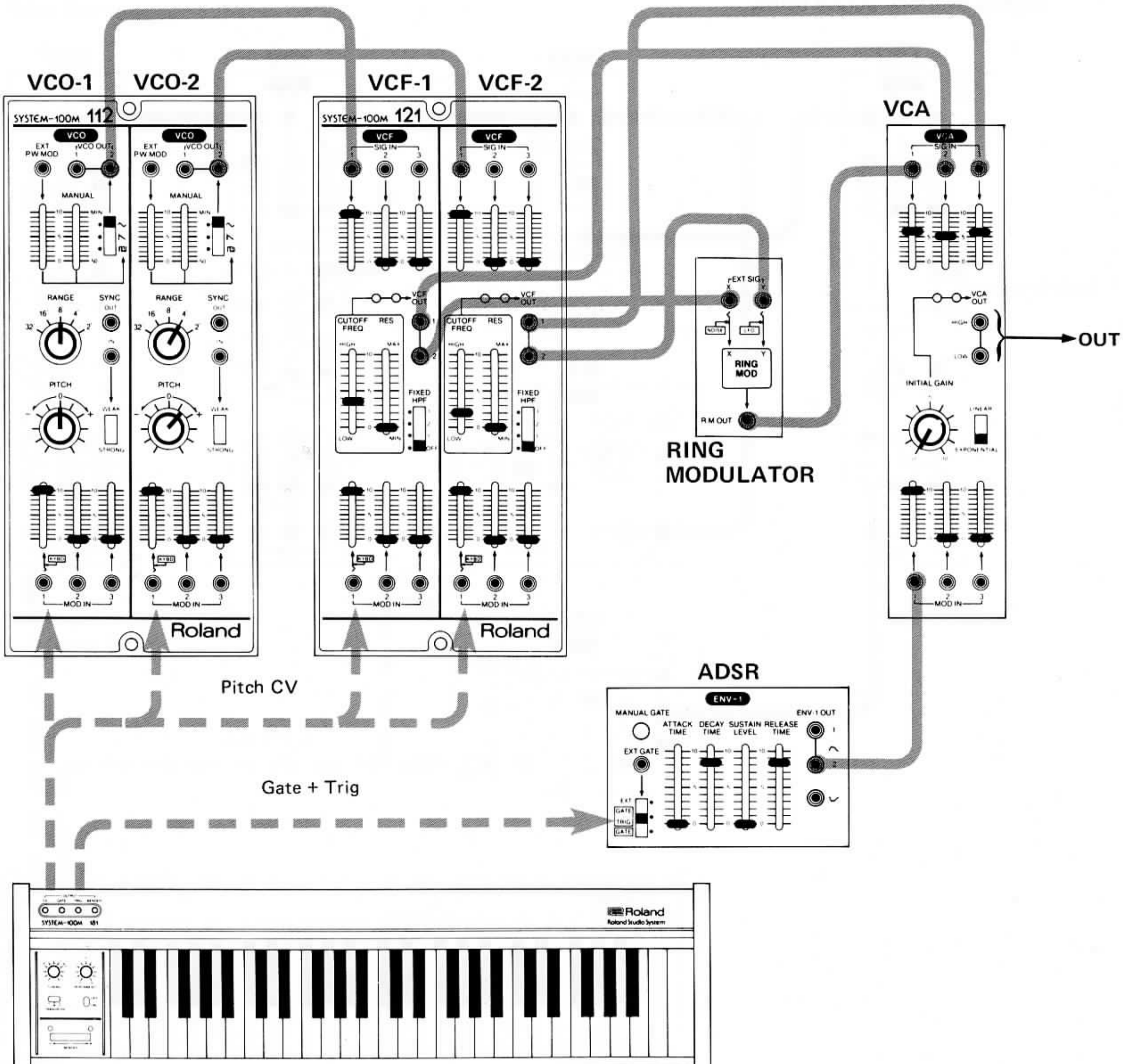
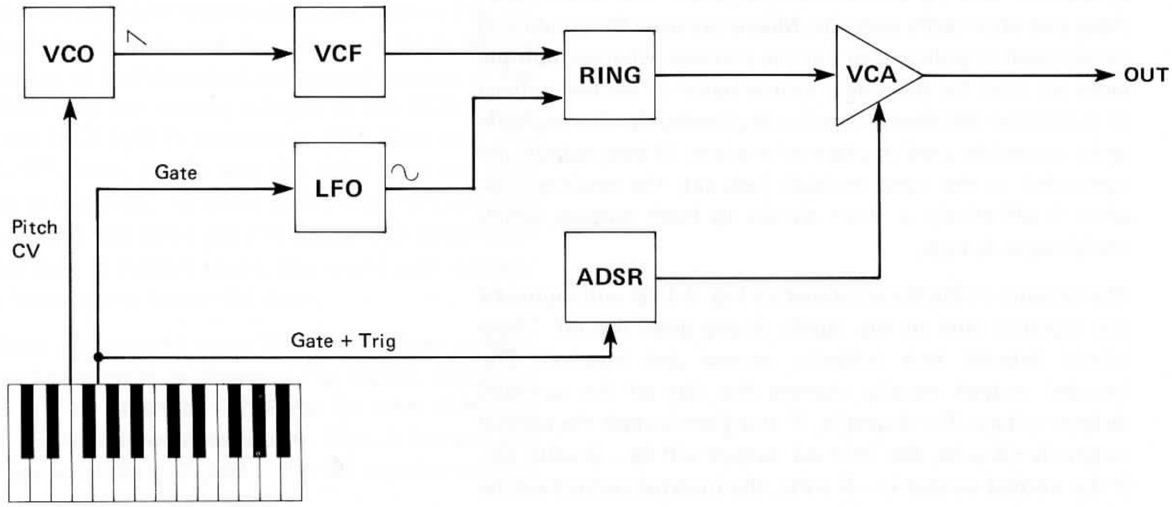


Fig. 2-19 shows how to use the ring modulator to produce a very effective bowed tremolo string sound. The LFO frequency determines the tremolo rate. The LFO KYBD TRIG switch insures that the tremolo will start each time a key is pressed.

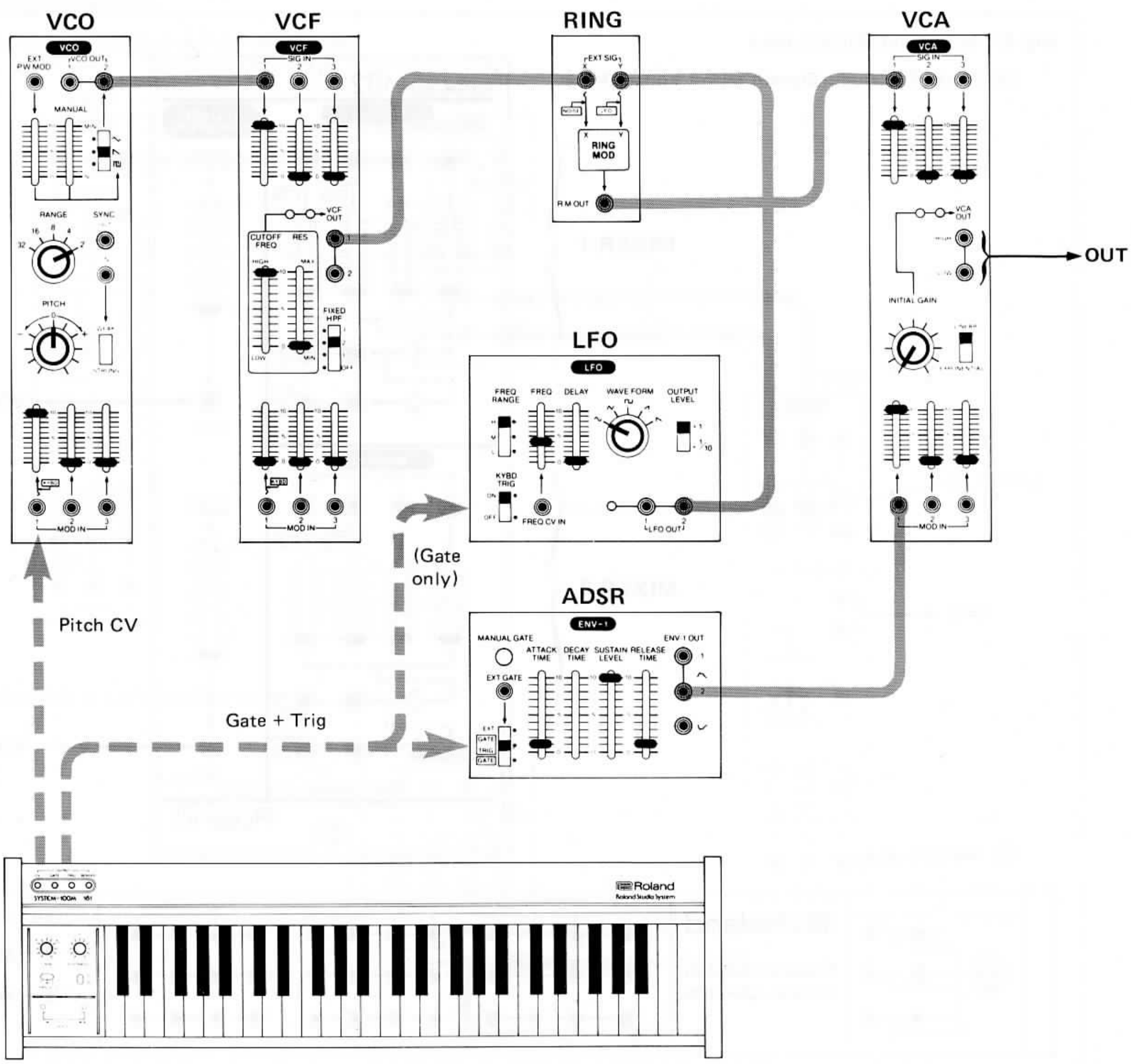
To analyze this sound, assume that the VCO frequency is 880Hz and the LFO frequency is 6Hz. The output of the ring modulator will be a compound of 886Hz (880 + 6) and 874Hz (880 − 6). Since the LFO frequency is low, the sum and difference frequencies are very close to each other and we may think of them as being equally pitched above and below the 880Hz VCO output. The apparent pitch of the sound, then, seems to be centered between the sum and difference frequencies at 880Hz. If the sum and difference frequencies happened to be farther apart, 500Hz above and below, for example, the pitch intervals above and below would not be equal. The sum frequency of 1380Hz (880 + 500) would be approximately a minor sixth above and the difference frequency of 380Hz (880 − 500) would be slightly over one octave below.

Fig. 2-19 **VIOLIN**, Bowed Tremolo

ⓐ Block Diagram



ⓑ Patch Diagram



Chapter Three:
Advanced Synthesis

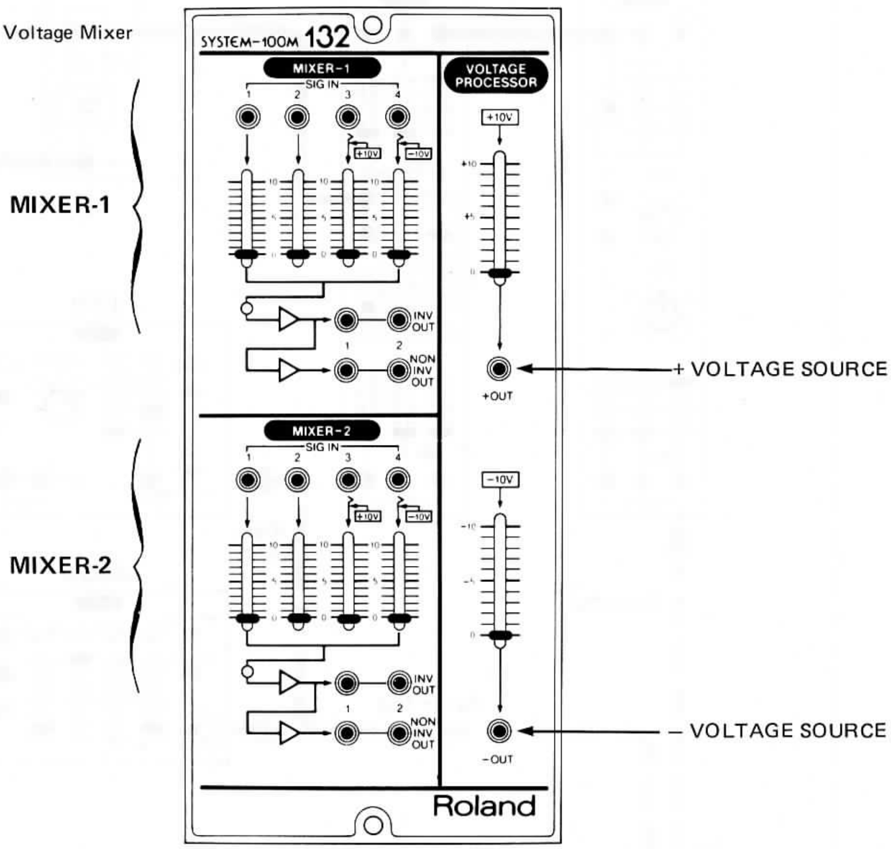
3-1 Mixers and Fixed Voltage Sources

In Fig. 3-1, part (a) shows an example of a mixer as used in a synthesizer and (b) shows multiple jacks. The functions of these two are exactly opposite. **Mixers** are used for combining various audio signals and/or control voltages, whereas **multiple jacks** are used for dividing a source signal or control voltage so that it can feed several inputs simultaneously. *The multiple jacks cannot be used in place of a mixer*. If two outputs are connected to the same multiple jack set, the result will be what is effectively a short circuit to both outputs which could cause damage.

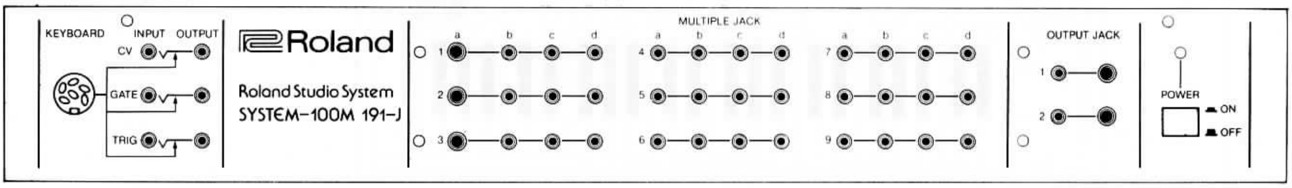
The outputs of the mixers shown in Fig. 3-1 (a) will represent the algebraic sum of the inputs at any given instant. These mixers provide two outputs: normal and inverted. The inverted output merely changes the sign of the summed output voltage. For example, if at a given instant the normal output is +3 volts, the inverted output will be –3 volts. Or, if the normal output is –5 volts, the inverted output will be +5 volts. In many situations, this inverting function can be quite convenient and important.

Fig. 3-1 Mixers and Multiple Jacks

ⓐ Model 132 Audio Signal/Control Voltage Mixer



ⓑ Multiple Jacks

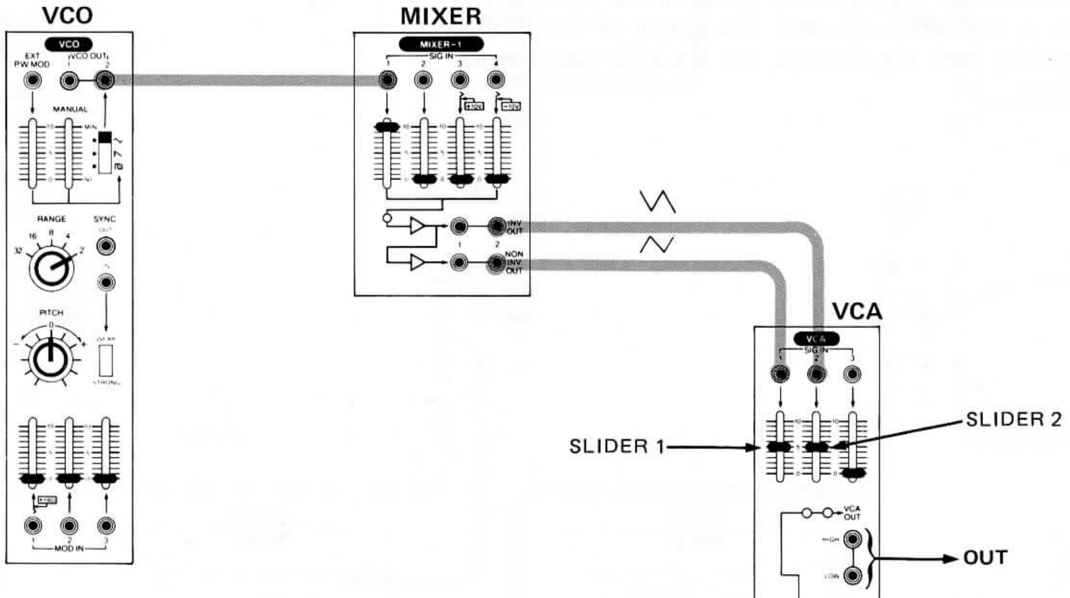


Often, when mixing audio signals, it will not matter whether the normal or inverted output is used, but in some cases it may be important since signals which are similar but of opposite phase (inverted) will tend to cancel each other. Fig. 3-2 demonstrates this. In this case the mixer in (a) is merely acting as a source of both inverted and normal versions of the same waveform and the mixing is done at the VCA input. Start with the VCA SIG IN controls at "0". First raise one of them to "5". Next, slowly raise the other level and note the decrease in loudness. At some point near "5", the two input levels will be the same and the signals will cancel each other. If the level is further raised, the sound will reappear because the levels are no longer the same.

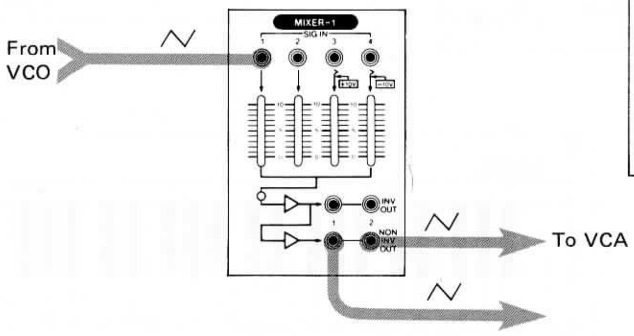
Try the above experiment using the same mixer output, either inverted or normal, as shown in Fig. 3-2 (b). Raise one of the VCA SIG IN sliders to "5". Raise the other slider and notice how the sound becomes louder. This is because the two signals are in phase with each other so that the voltages at any given instant are added together.

Fig. 3-2 Adding Waveforms

ⓐ Cancellation of Out of Phase Waveforms



ⓑ Addition of In Phase Waveforms



The mixers shown in Fig. 3-1 incorporate built-in voltage sources which are useful when it is necessary to bias some signal or control voltage. **Bias** refers to the addition of a fixed value to a signal or control voltage. The "function control" of Fig. 2-1 can be thought of as merely biasing the module function at some level after which control inputs are used to alter that function level. In the tremolo example of Fig. 2-12, for example, the VCA INITIAL GAIN control biases the VCA so that when added to the LFO input, the result is tremolo.

Fig. 3-3 shows how to use a voltage source for biasing the output of an LFO so that tremolo can be added to a sound without the need for a second VCA. With the "−10v" mixer level control at "0", set the VCA MOD IN level for the LFO to the approximate depth of tremolo desired while holding a key on the keyboard down. Next, raise the "−10v" mixer level until the tremolo effect disappears when a key is not being held down. This should happen near "5". With the level at "5", a voltage of about −5 volts is added to the LFO waveform. This pushes the zero line of the LFO sine wave down to −5 volts, a level low enough to keep the LFO sine wave from "opening" the VCA when a key is not depressed. When a key is depressed, the output of the envelope generator is added to the VCA's control inputs so that the LFO sine wave is now above the zero. The output of the VCA is shaped by both the envelope and the LFO wave simultaneously.

Fig. 3-3 Tremolo (One VCA)

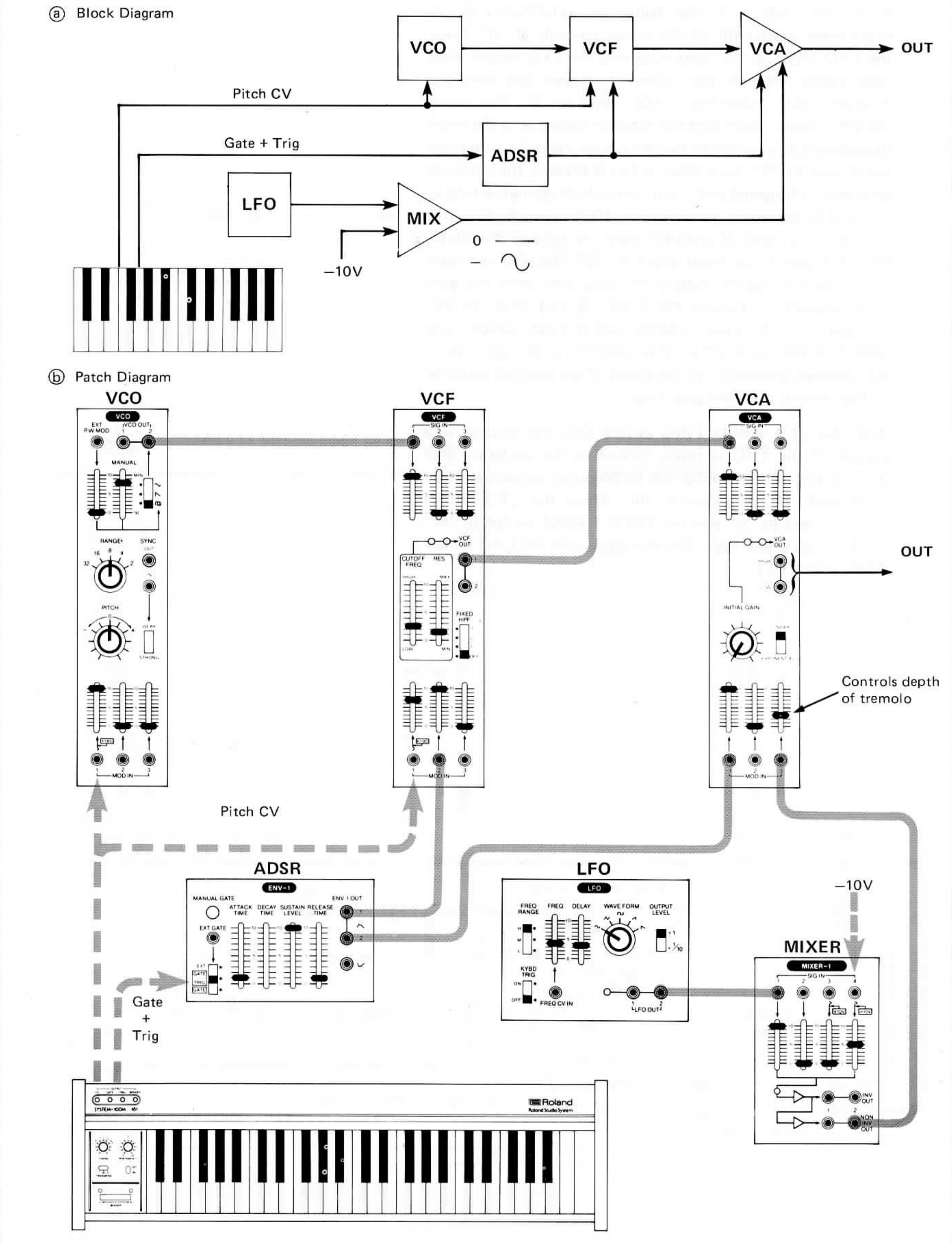
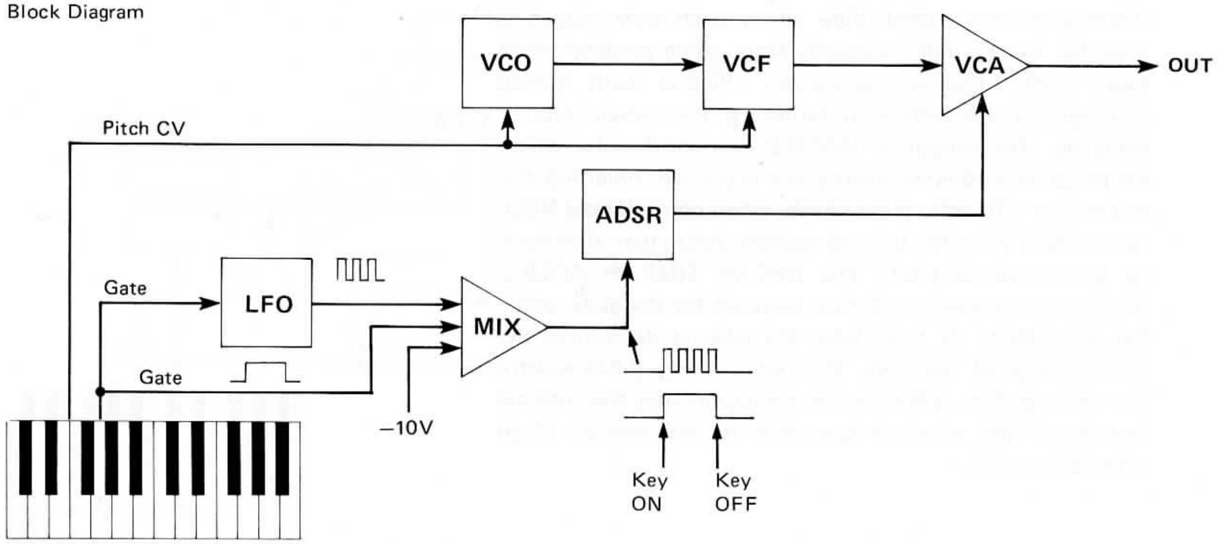


Fig. 3-4 shows how to use this principle of baising to provide a rapidly repeating trigger to imitate a playing technique sometimes used with the banjo or xylophone. As an experiment, start with all the mixer controls at "0". Raise the LFO input to "5" and note that the LFO square wave now passes through the mixer to trigger the envelope generator. Next, raise the "−10v" level to "5". This biases the LFO square wave into the negative region, thus the envelope generator is no longer triggered. Last, raise the gate pulse input level to "5". Now when a key is pressed, the envelope generator is triggered and continues to be triggered as long as the key is depressed. With the "−10v" slider at "5", this represents an input of approximately −5 volts to the mixer. With the gate pulse input slider at "5", the +10 volt gate pulse becomes approximately +5 volts, and when the gate pulse appears, it cancels the fixed −5 volt level. In this configuration, the mixer is being used as a gate which turns some function on or off (in this case, the LFO triggering of the envelope generator) as the result of an external pulse or voltage (in this case, the gate pulse).

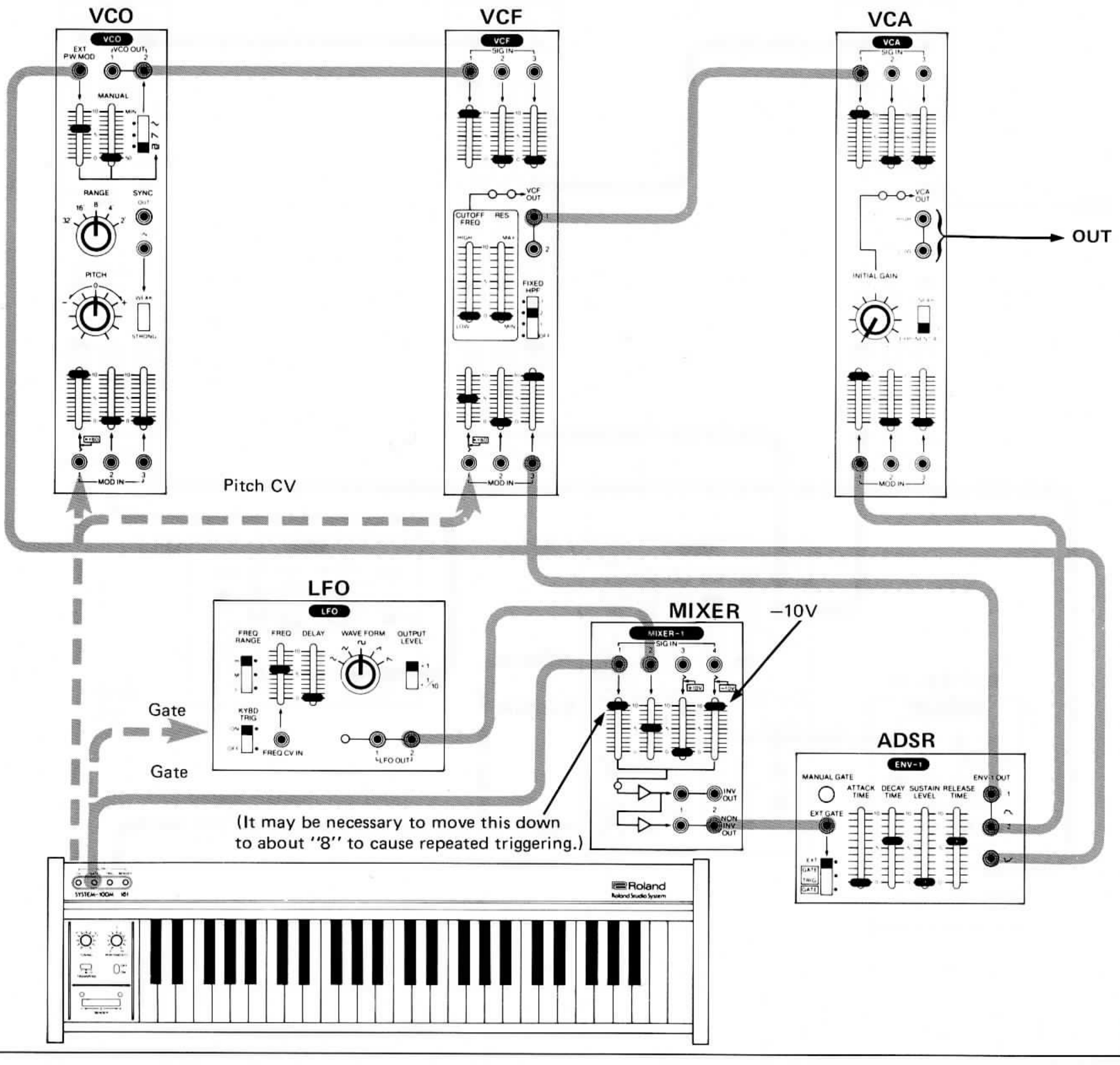
With the LFO KYBD TRIG switch ON, the square wave output of the LFO is phase locked to the keyboard gate pulse to ensure that sound will be produced as soon as a key is pressed. To demonstrate this, Move the LFO FREQ control down to "0" and the FREQ RANGE switch to "M" or "L". Tap a few keys. Try this again with the LFO KYBD TRIG switch OFF.

Fig. 3-4 BANJO (Repeating Trigger)

ⓐ Block Diagram



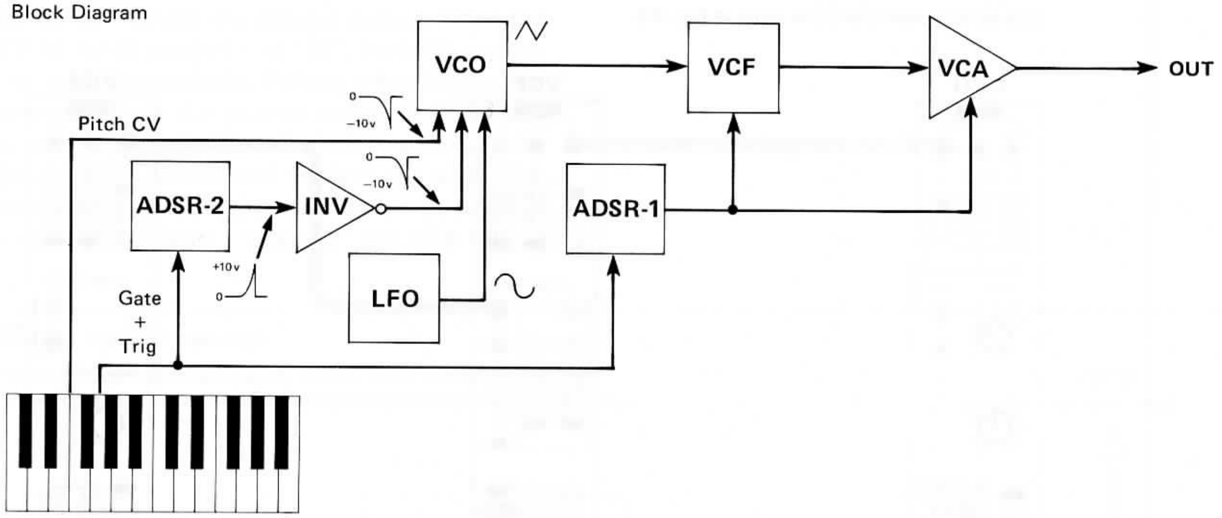
ⓑ Patch Diagram



The remainder of this section on mixers will be concerned with the invert function. Fig. 3-5 shows how to produce an effect sometimes called **glide** where each note played is preceded by a "slide" upwards from some predetermined lower pitch. This is particularly effective with human generated sounds such as imitations of the human voice or whistling. The output of ADSR-2 is a short pulse which jumps up to +10 volts when a key is pressed. Inverting this makes it a −10 volts pulse which, when controlling a VCO, causes the VCO pitch to jump quickly down, then slide back up to its normal pitch. The DECAY TIME of ADSR-2 determines the amount of time required for the glide, while the VCO MOD IN level from the inverter determines the starting pitch of the glide. The patch of Fig. 3-6 is exactly the same as Fig. 3-5 with the exception that the internal inverter of the envelope generator is used instead of an external inverter.

Fig. 3-5 WHISTLING (Glide)

ⓐ Block Diagram



ⓑ Patch Diagram

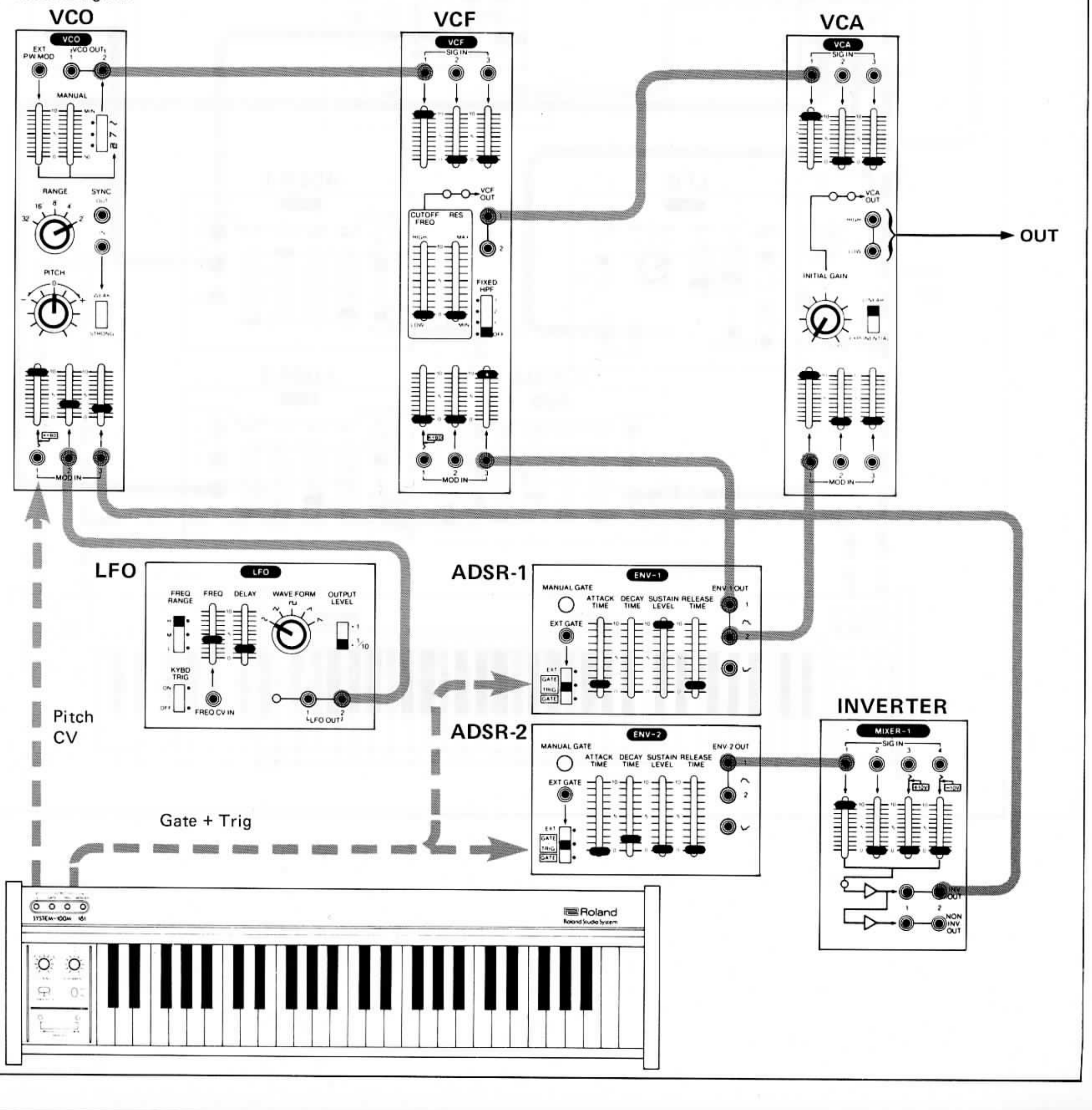


Fig. 3-6 WHISTLING

This patch is essentially the same as Fig. 3-5.

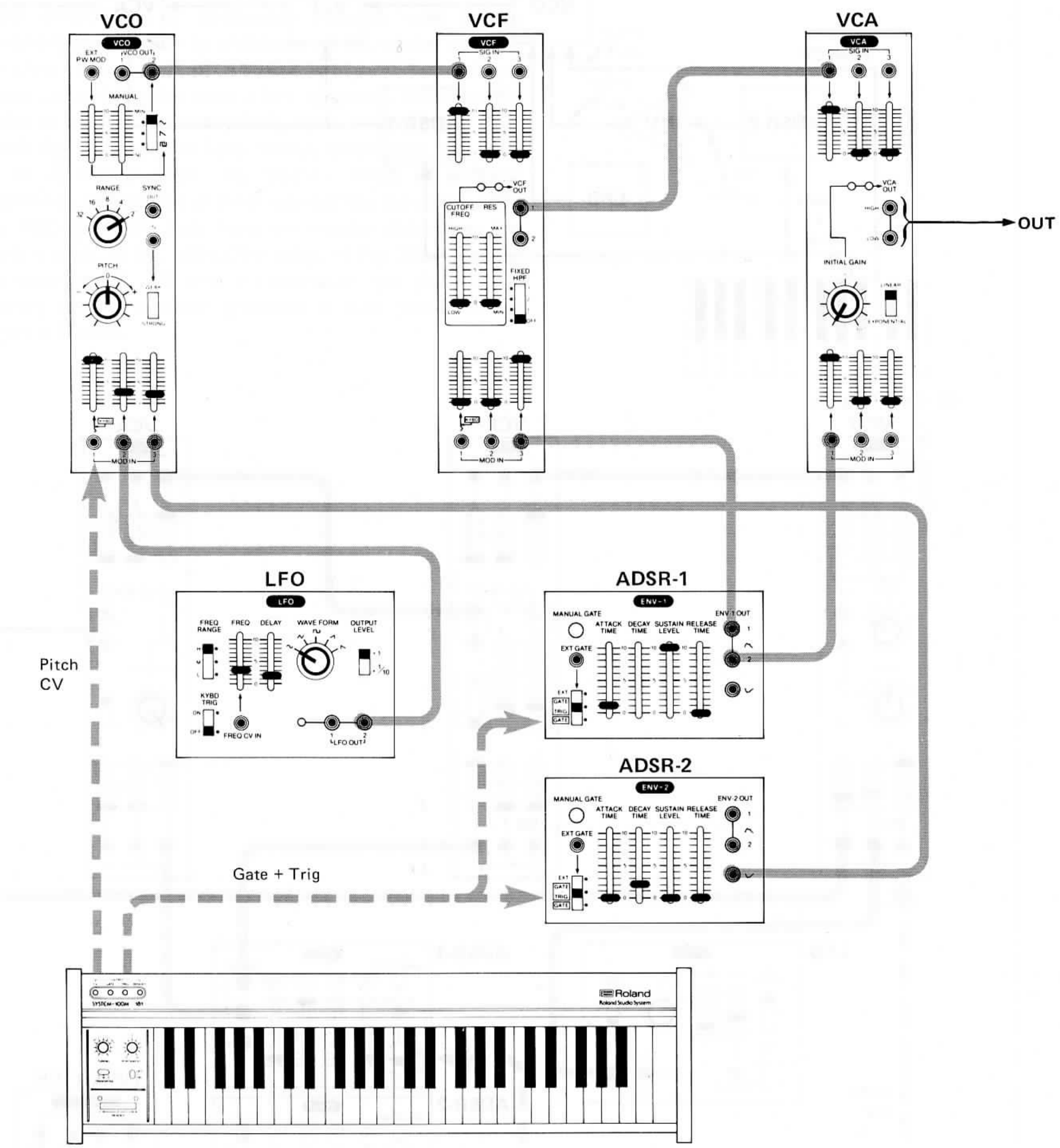
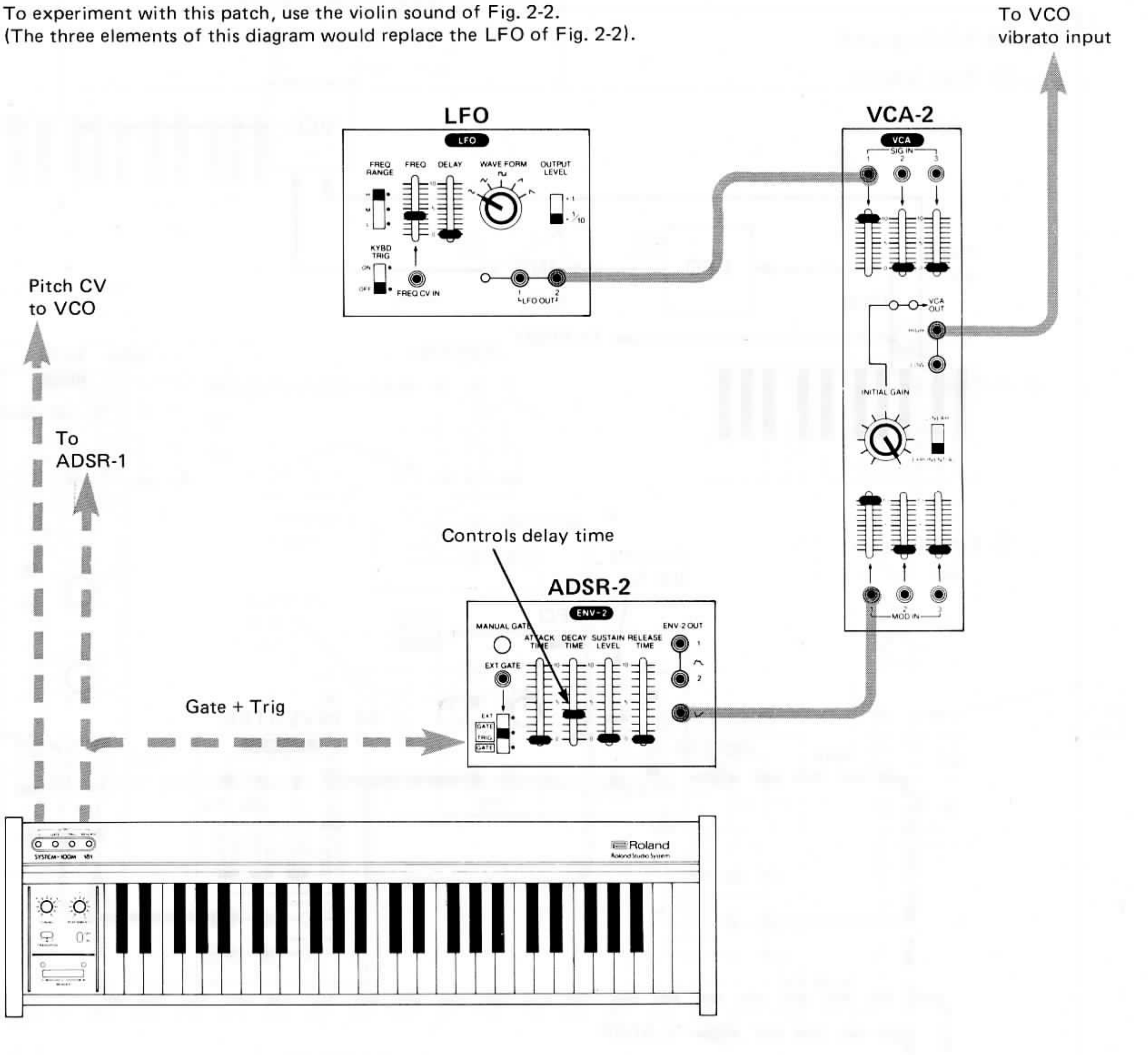


Fig. 3-7 shows how to use the above principle as a better way to produce delayed vibrato effects in those synthesizers with **LFOs which do not include the delayed output. Since the** VCA-2 INITIAL GAIN control is at "10", the VCO receives the LFO sine wave continuously. Pressing a key triggers the envelope generator. Since the inverted output of ADSR-2 is being used, a negative pulse occurs which counteracts the VCA INITIAL GAIN control and "closes" the VCA. The DECAY control determines the time required for the ADSR-2 voltage to rise to zero again, "opening" the VCA after a delay.

Fig. 3-7 Delayed Vibrato (Improved)

To experiment with this patch, use the violin sound of Fig. 2-2.
(The three elements of this diagram would replace the LFO of Fig. 2-2).



In the patch of Fig. 2-4 where the square wave output of the LFO is used to generate trills, the use of the LFO KYBD TRIG function causes the trills to start on the upper of the two pitches. If the square wave is inverted as shown in Fig. 3-8, the result will be a trill which starts on the lower pitch. Fig. 3-9 shows how to make the trill sound a little more natural by using the inverted envelope generator output to control the frequency of the LFO. When the inverted envelope jumps down to its negative value, the frequency of the LFO falls. As the inverted envelope rises, the frequency of the trills will increase. This imitates the style of playing where the first two or three notes of a long trill are slower than the remaining notes.

Fig. 3-8 Trill (Improved)

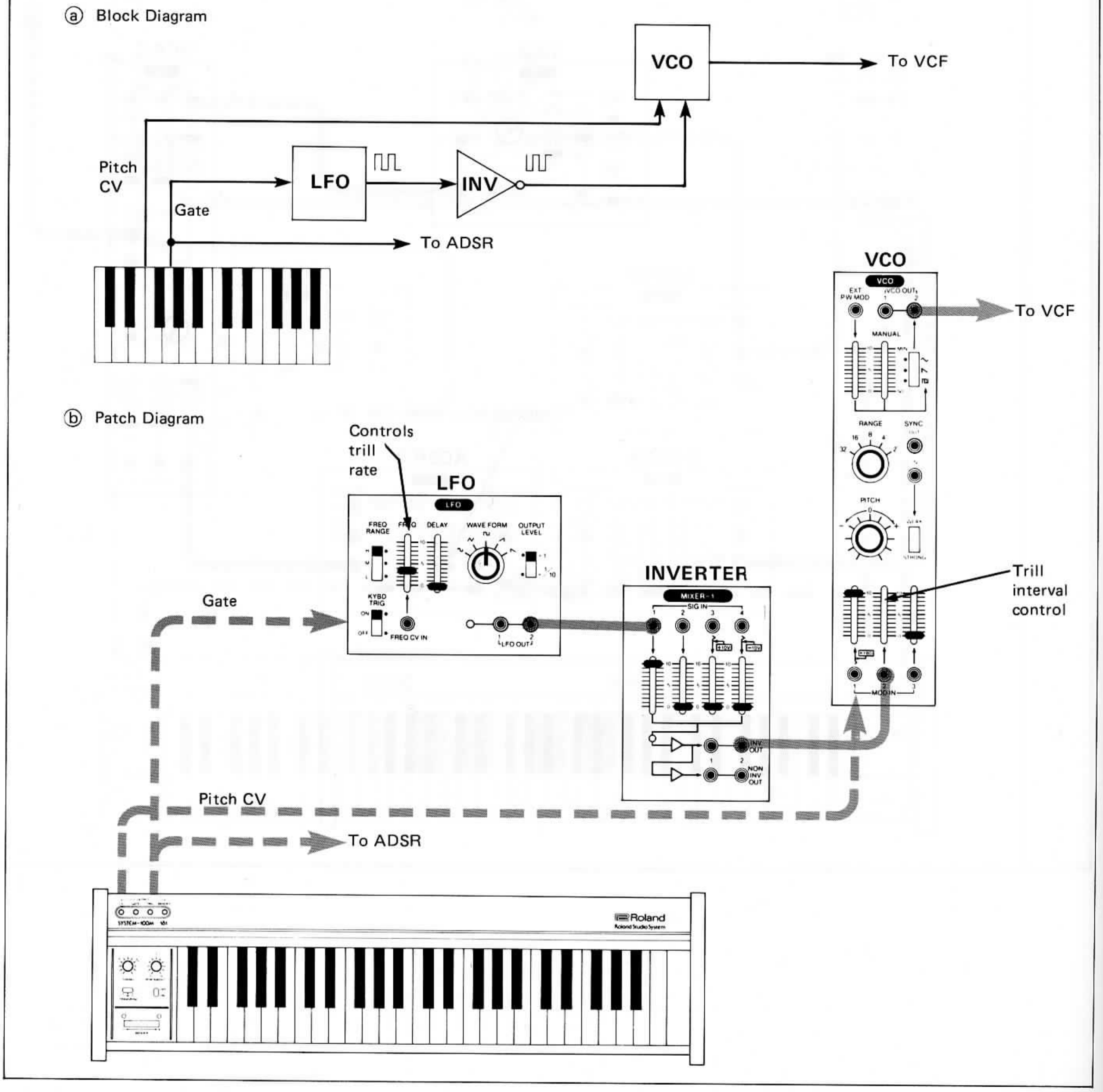
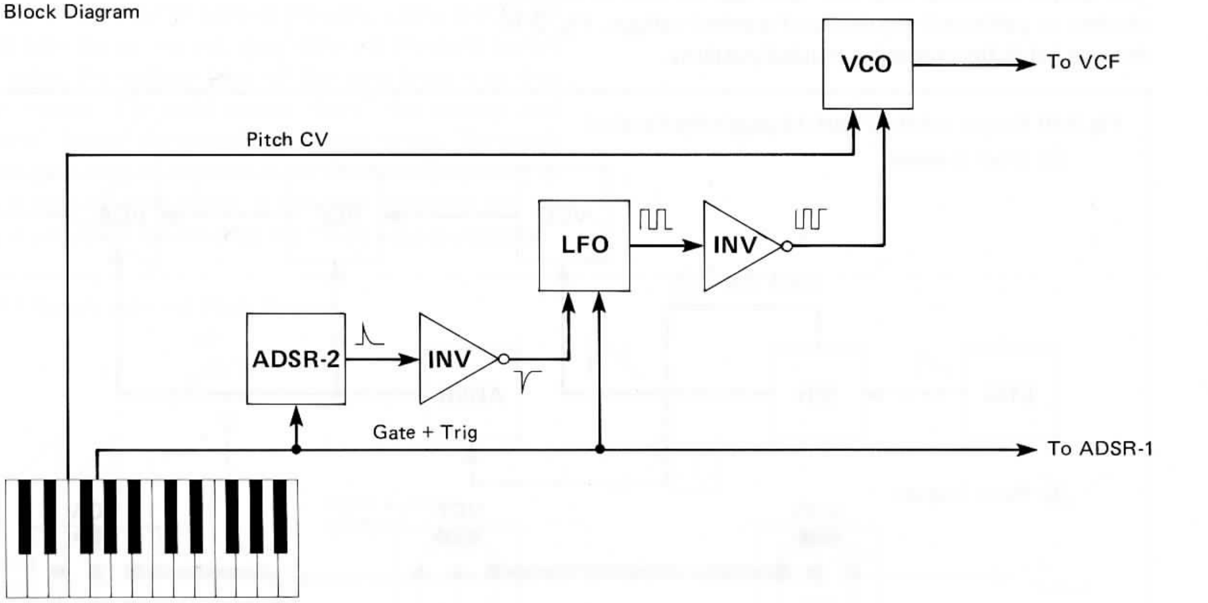
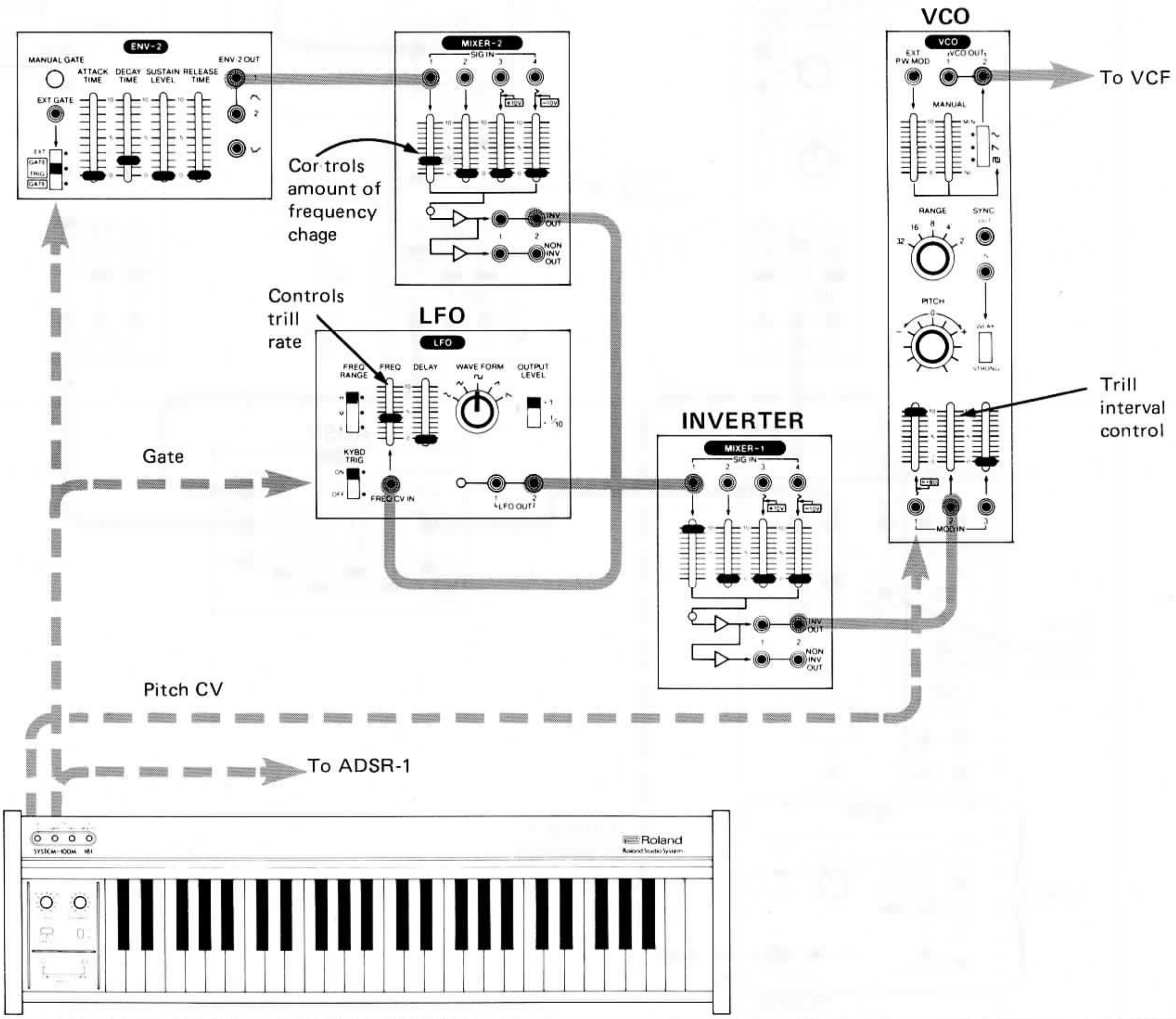


Fig. 3-9 Natural Trill

ⓐ Block Diagram



ⓑ Patch Diagram

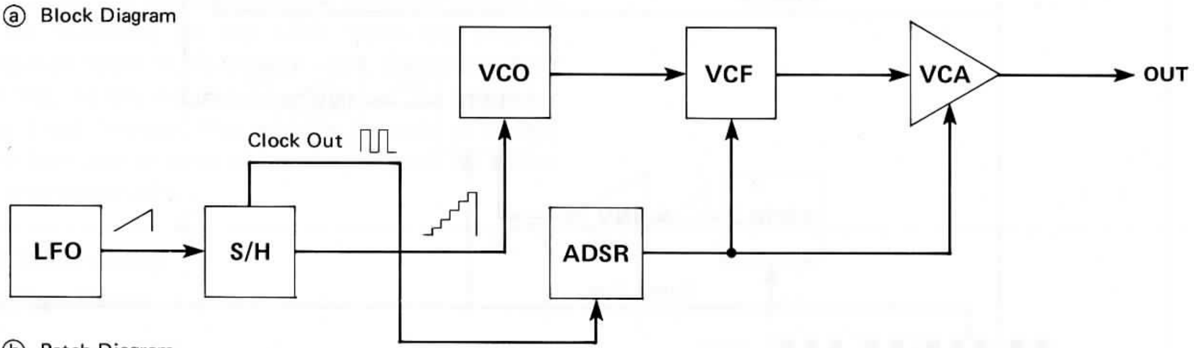


3-2 The Sample and Hold

The **sample and hold** (S&H, or S/H) can be used to produce random or patterned sequences of control voltage. Fig. 3-10 shows a patch for producing musical patterns.

Fig. 3-10 Sample and Hold Patch (Arpeggio-like Patterns)

ⓐ Block Diagram



ⓑ Patch Diagram

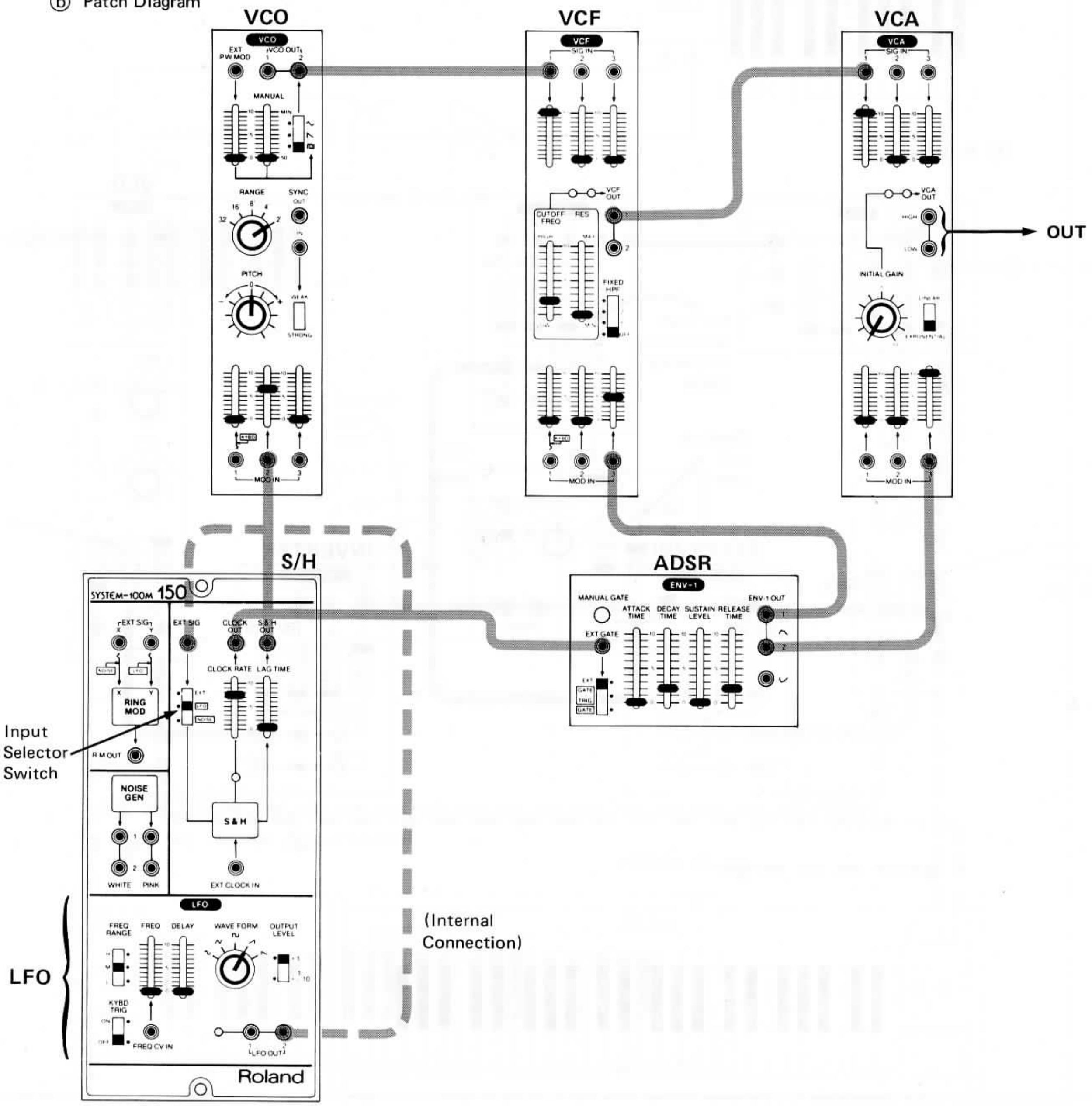


Fig. 3-11 shows how the sample and hold works. At the left, the clock circuit generates short, sharp pulses at regular intervals which are used to control the gate. Each clock pulse opens the gate for an instant, thus allowing the hold circuit to "see" what the voltage level of the waveform is at that particular instant. The hold circuit "sees" this voltage and "remembers" it until the next time the gate opens. The clock circuit also generates a square wave of the same frequency as the gate pulses so that other synthesizer elements can be triggered in synchronization with the timed voltage samples.

Fig. 3-11 Sample and Hold Block Diagram

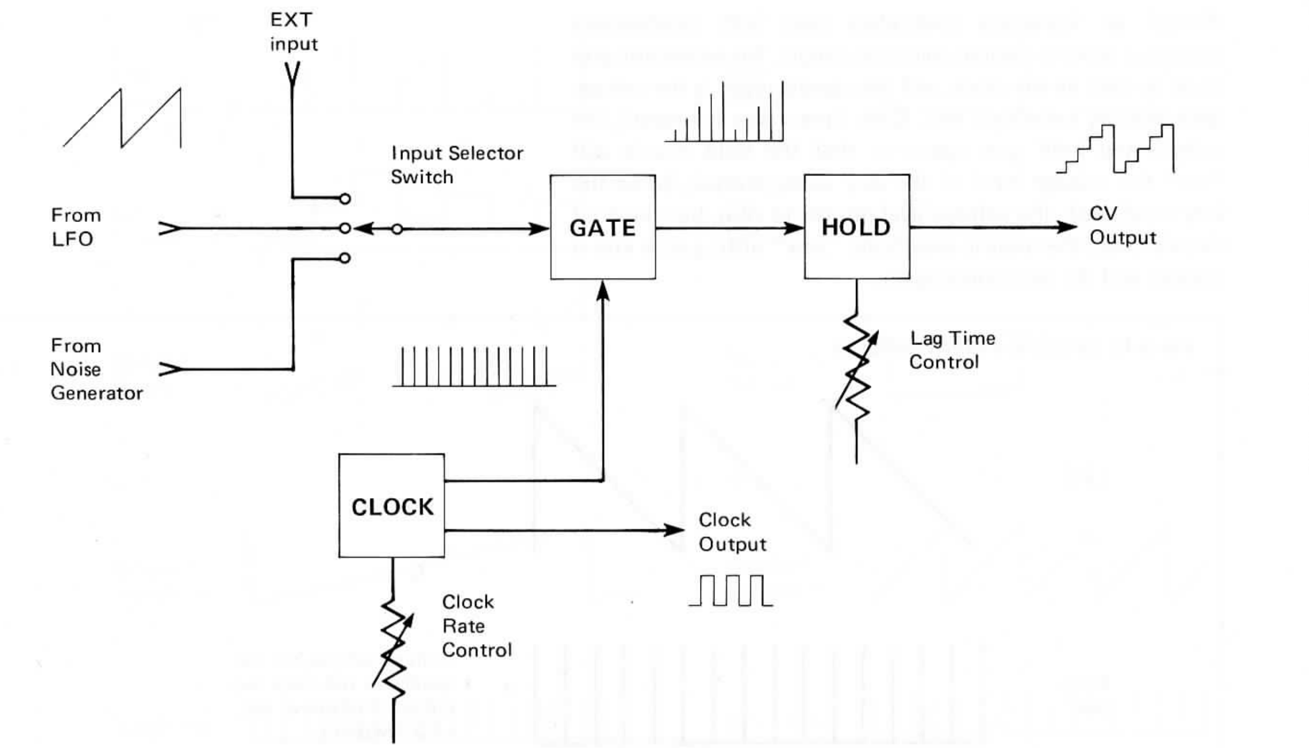


Fig. 3-12 shows more clearly the relations between the different waveforms in the sample and hold circuit. Also shown is the musical representation of the pattern produced if the sample and hold output is used to control the pitch of a VCO. The actual frequencies of these pitches will depend on the voltage levels at the time each sample is taken and, more often than not, these will not fall within conventional musical scale systems. This is not a disadvantage, however, as the resulting sound pattern can add beautiful tonal effects and coloring to music built on a conventional scale. By changing the frequency of the input waveform and/or the rate at which the samples are taken (CLOCK RATE control), the patterns may be varied. Fig. 3-13 shows a few possibilities.

Almost all keyboard controllers used with synthesizers contain a built-in sample and hold circuit. The keyboard gate pulse is used as the clock and the sample input is the voltage generated by pressing a key. Each time a key is pressed, the sample and hold gate opens so that the hold circuit will "see" the voltage level of the key being pressed. When the key is released, the voltage level returns to zero, but the hold circuit holds the level it previously "saw" until a new key is pressed and the gate opens again.

Fig. 3-12 Sample and Hold Waveforms

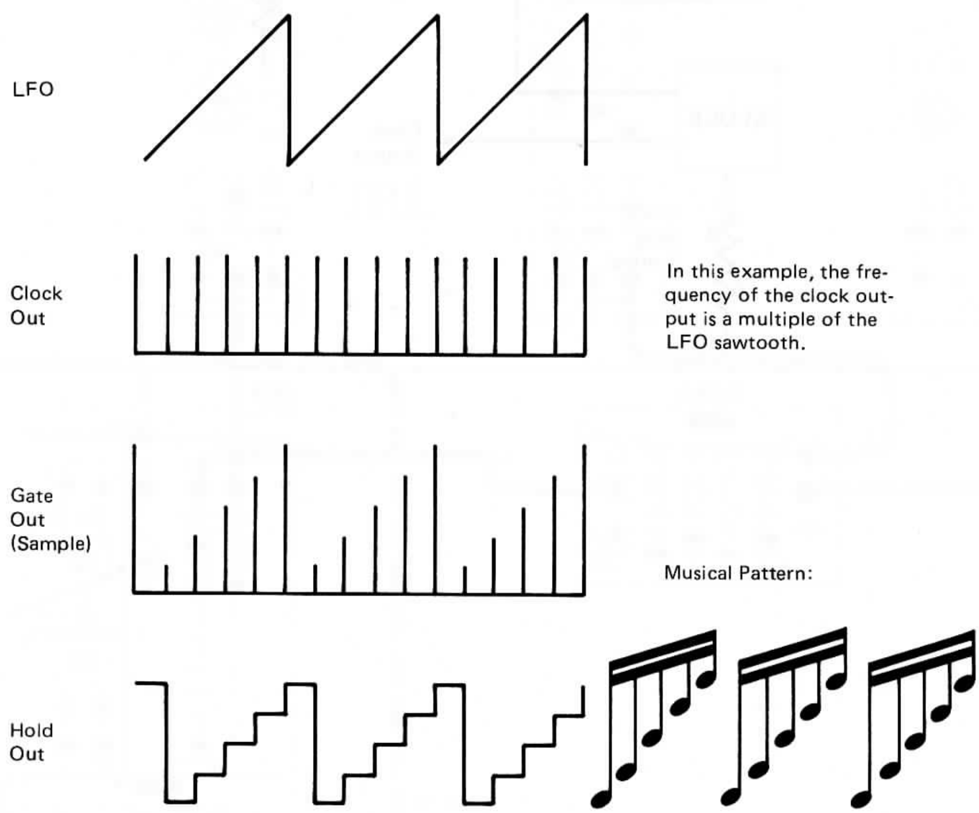
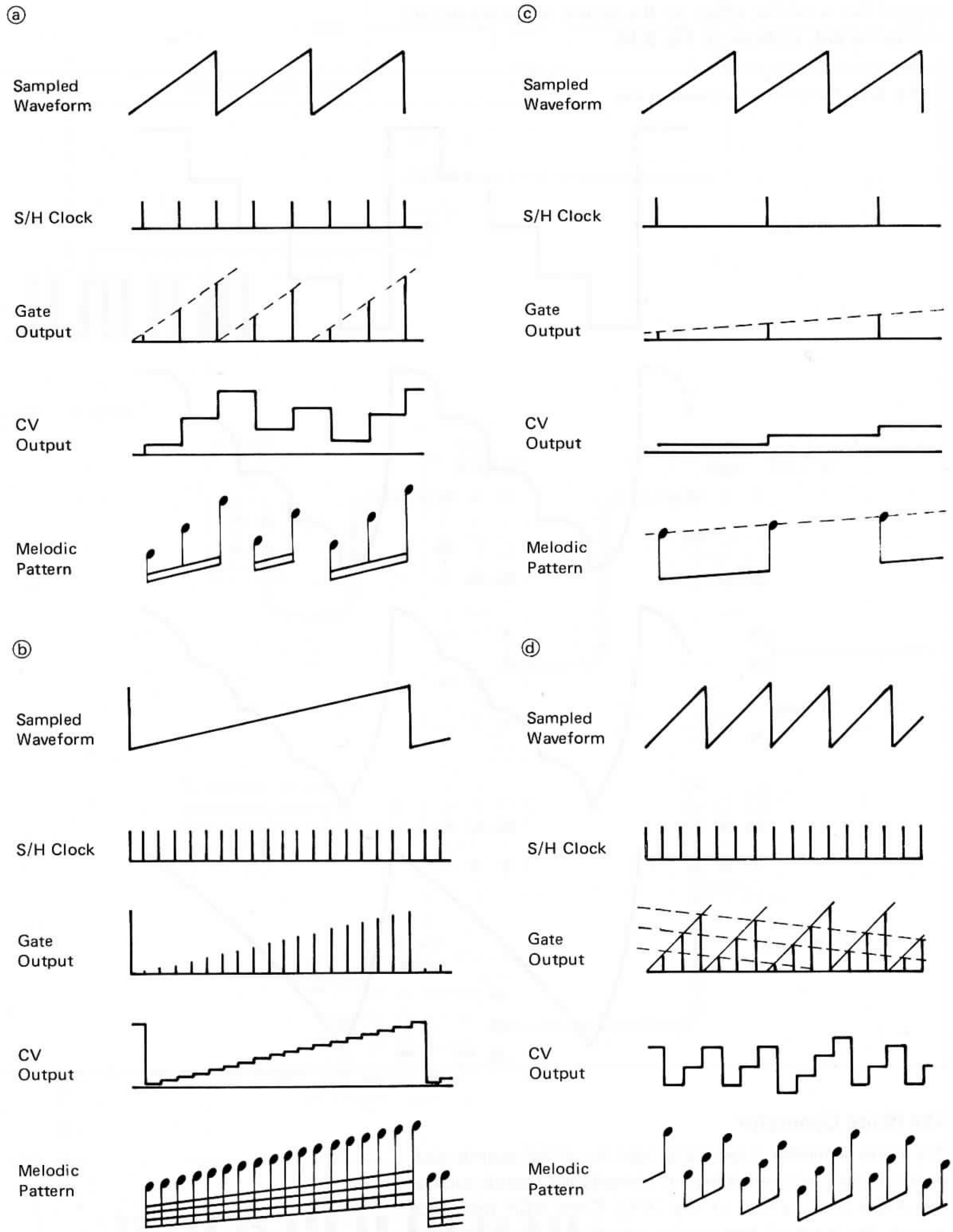


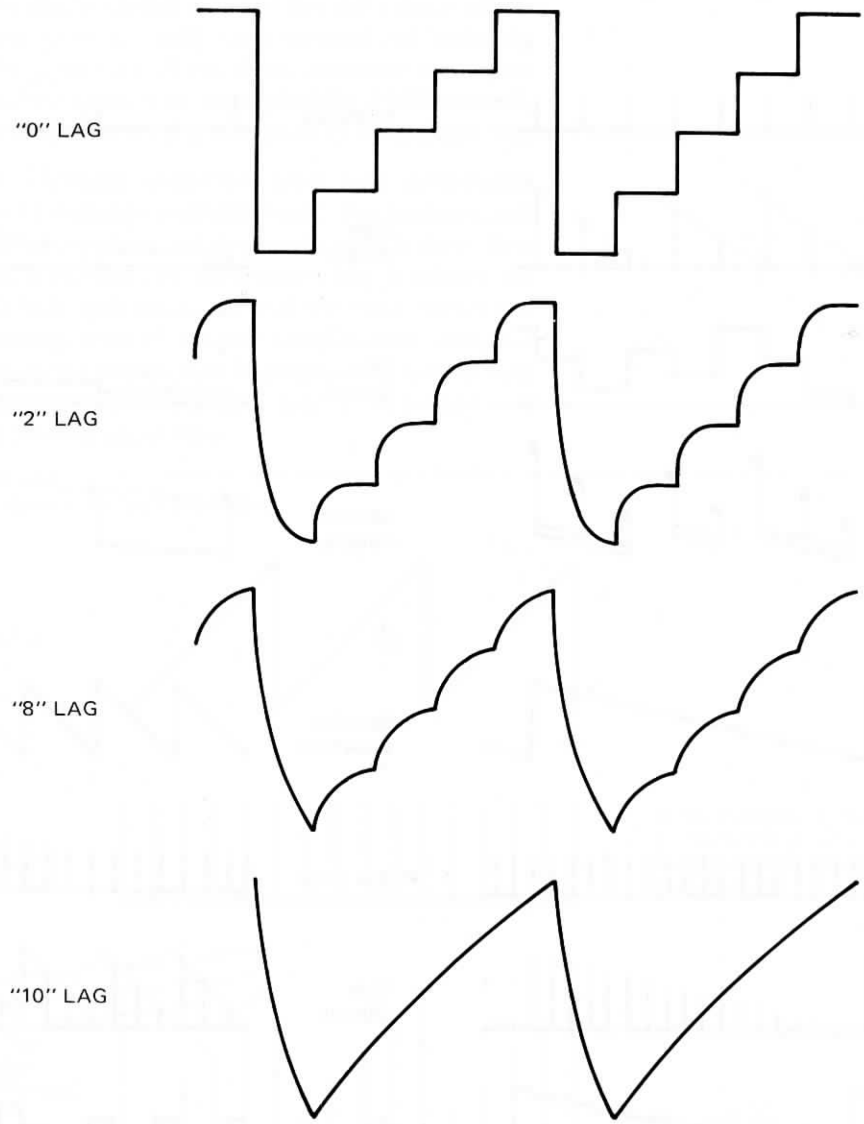
Fig. 3-13 Sample and Hold Patterns

Below are a few of the patterns possible when sampling the LFO sawtooth wave.



Portamento acts like a low pass filter on the keyboard voltages changes. When it is used, the keyboard control voltage output will slide from one level to the next when different pitches are played. The sample and hold LAG TIME control has a similar effect on the sample and hold control voltage output, as shown in Fig. 3-14.

Fig. 3-14 Sample and Hold Output Lag

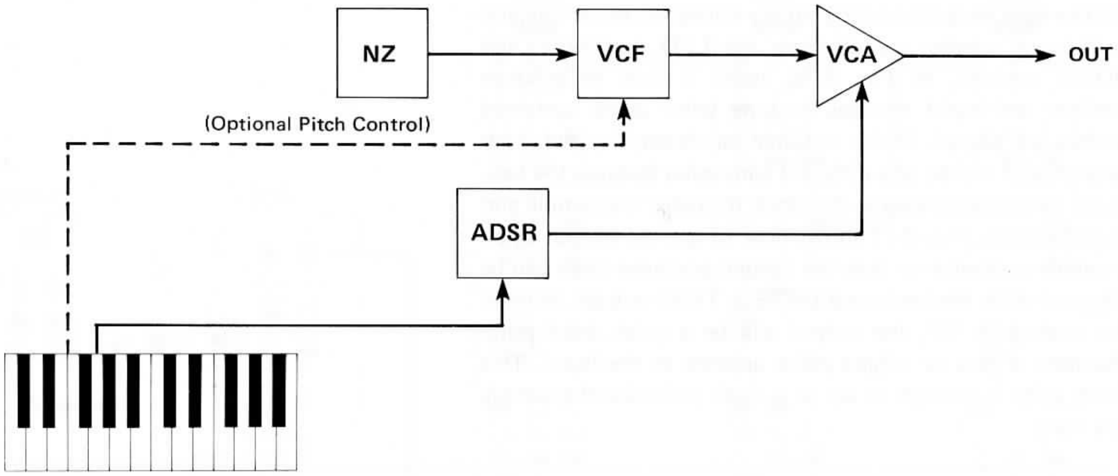


3-3 The Noise Generator

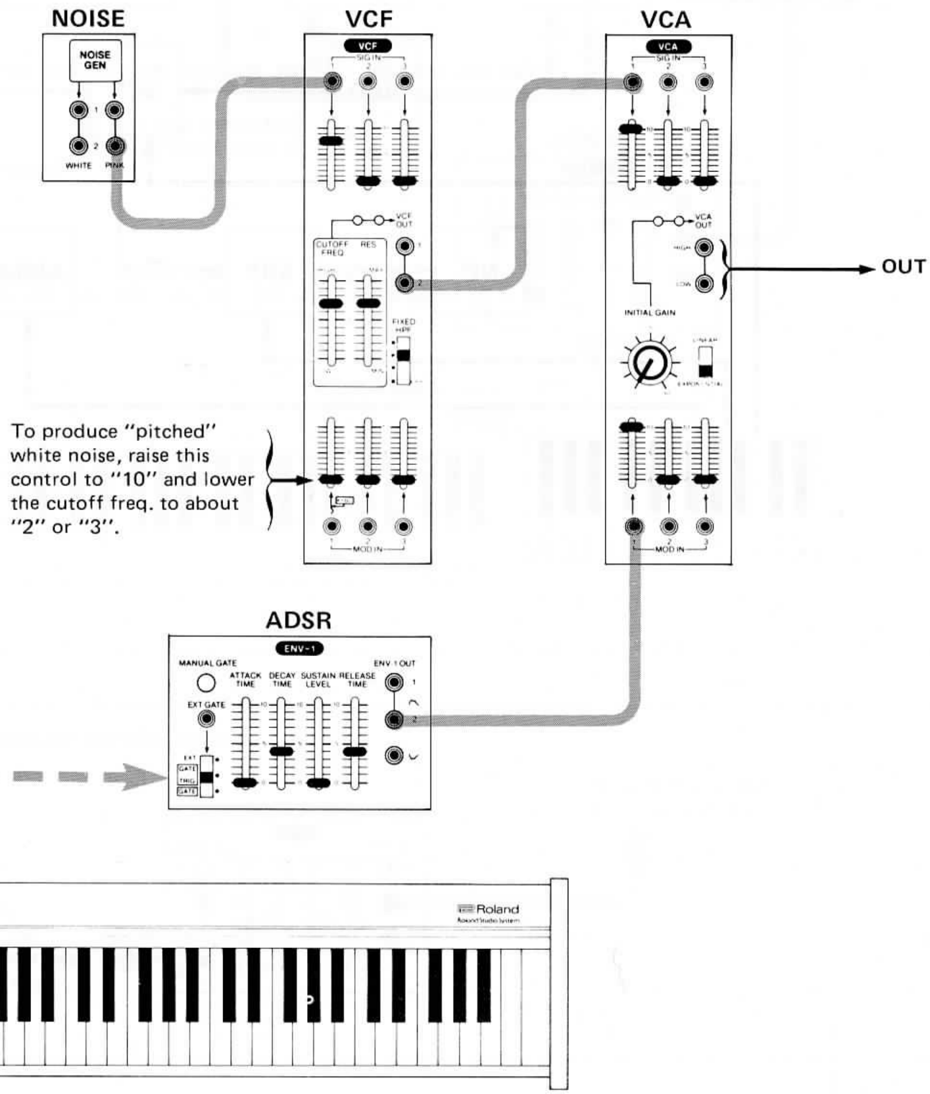
The **noise generator** is generally used for effect sounds such as wind, surf, and whistling, or non-pitched sounds such as the wood block sound of Fig. 3-15. Since VCF resonance accents the band of frequencies around the cutoff point of the filter, it is possible to play melodies by using the keyboard pitch control voltage to control the cutoff point of the filter. This may be done by raising the VCF KYBD MOD IN control to "10" in Fig. 3-15. Adding this voltage to the VCF cutoff point raises the cutoff point too much, so this must be corrected by lowering the CUTOFF FREQ control to "2" or "3". Now melodies can be played.

Fig. 3-15 WOOD BLOCKS (Noise Generator)

ⓐ Block Diagram



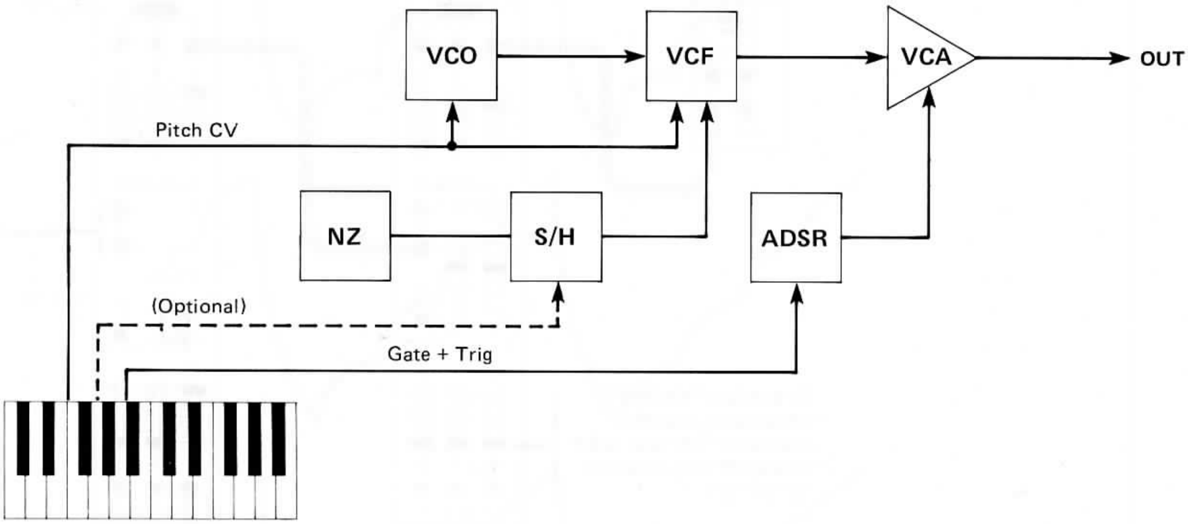
ⓑ Patch Diagram



Since noise is a *random* combination of all frequencies, the sample and hold may be used to sample the noise waveform for producing random voltage patterns. In Fig. 3-10, random pitches may be produced by simply changing the sample and hold input selector switch from the LFO position to the NOISE position. In Fig. 3-16, noise is used to produce random and rapid changes in tone color when sustained pitches are played. If the optional patch cord is used, each note played will be of a different tone color because the keyboard gate pulse is used as the clock to trigger the sample and hold function. Fig. 3-17 shows how to use the output of an envelope generator so that the sample and hold clock can be triggered with the keyboard GATE + TRIG output. With all the control at "0", the output will be a quick sharp pulse whenever a gate or trigger pulse appears at the input. This short pulse is enough to act as a clock pulse for the sample and hold.

Fig. 3-16 Random Tone Color Modulation

ⓐ Block Diagram



(b) Patch Diagram

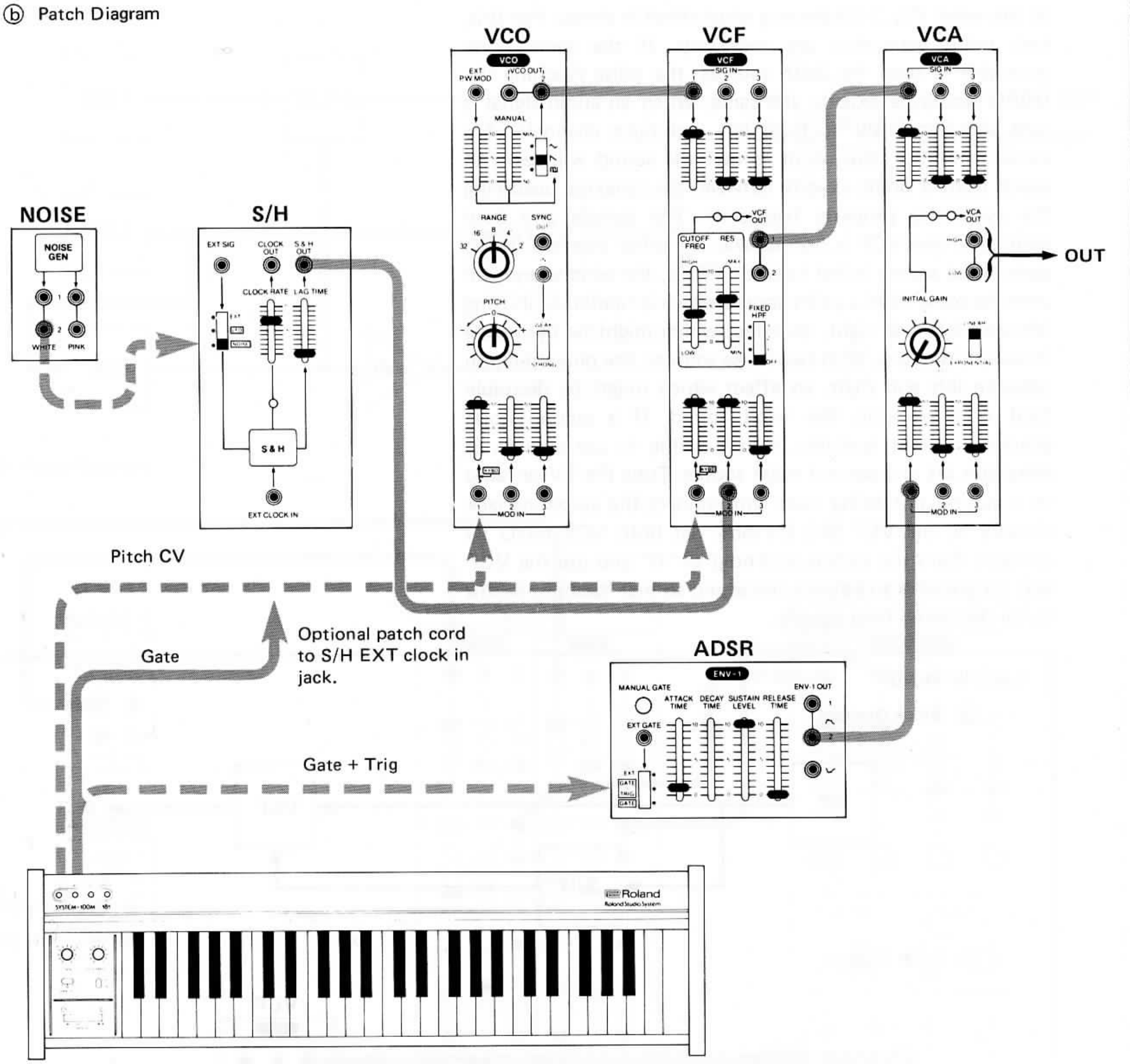
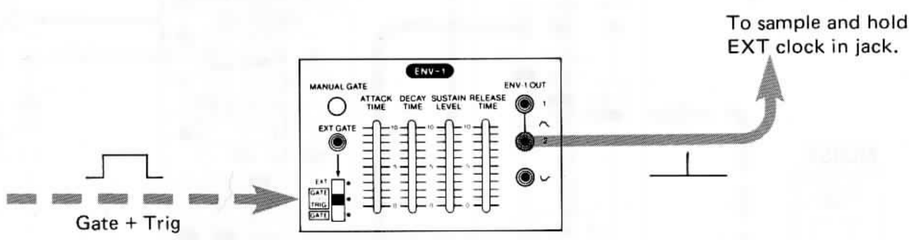


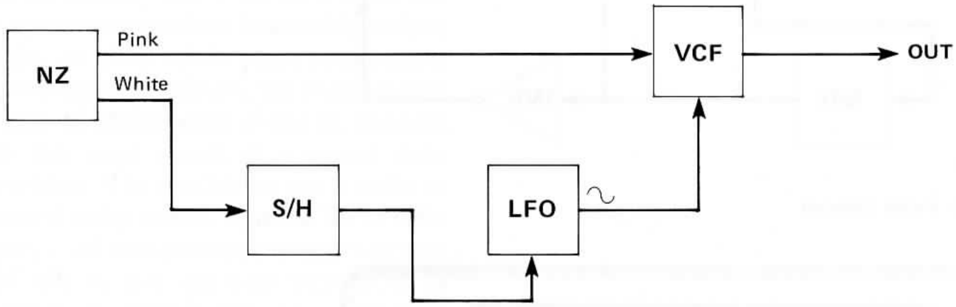
Fig. 3-17 Gate + Trig for Sample and Hold Clock Input.



Surf may be produced by using the rushing sound of **pink noise**, as shown in Fig. 3-20. With surf, the roll of the waves should be a little more regular than the changing patterns of wind; therefore, the LFO sine wave output is used as the basic source of control. The frequency of the LFO, however, is controlled by the random output of the sample and hold so that the pattern of waves does not become too regular. Fig. 3-21 shows a stereo surf patch.

Fig. 3-20 SURF

ⓐ Block Diagram



ⓑ Patch Diagram

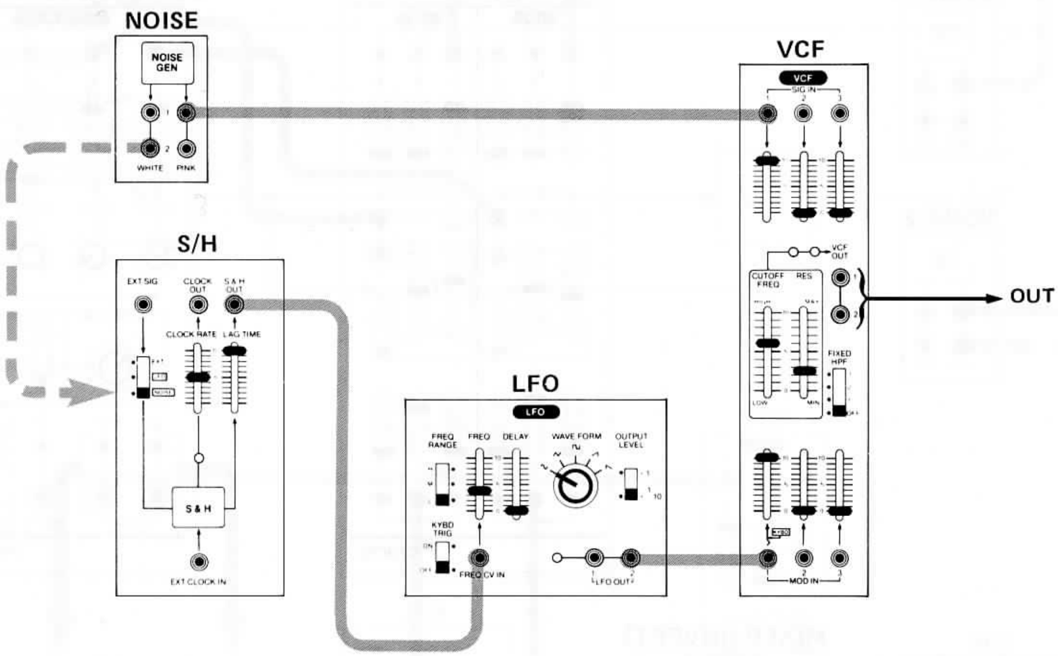
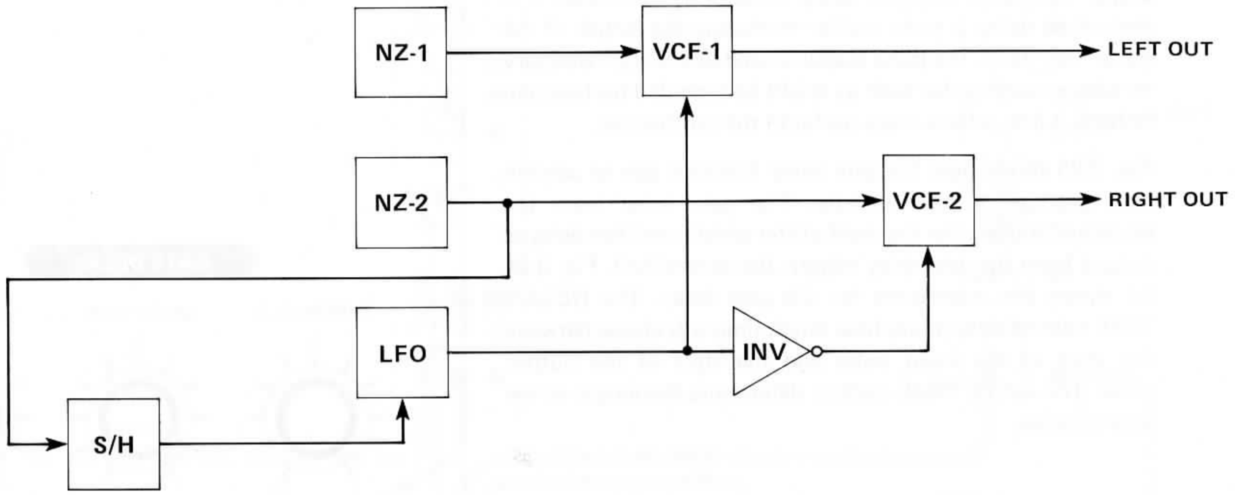
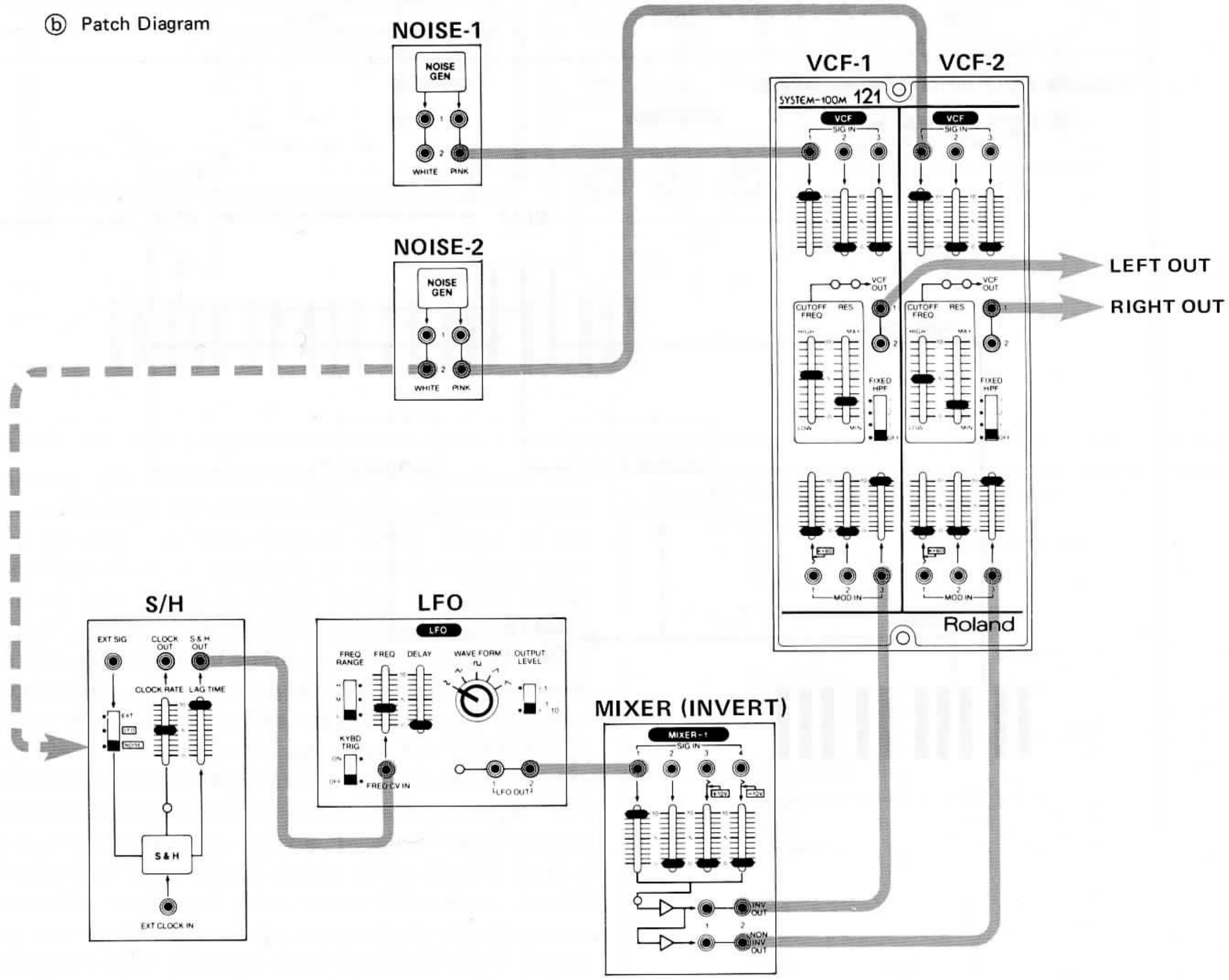


Fig. 3-21 SURF in Stereo

ⓐ Block Diagram



ⓑ Patch Diagram



3-4 The Gate Delay and Pulse Shaper

In the System 100M, the gate delay portion of the Model 172 module (Fig. 3-22) provides both **gate delay** and **pulse shaper** functions. The gate delay function is used when it is desired to delay a pulse and/or to change the length of the pulse "on" time. The pulse shaper is used where it is necessary to take a rough pulse such as might be recorded on tape and reshape it into a form more useful in the synthesizer.

Fig. 3-23 shows how the gate delay function can be used to help produce a wolf whistle. The gate pulse from the keyboard triggers the first half of the whistle and the delayed output from the gate delay triggers the second half. Fig. 3-24 (a) shows the waveforms for the gate delay. The DELAY TIME control determines how much time will elapse between the start of the input pulse and the start of the output pulse. The GATE TIME control determines the length of the output pulse.

Fig. 3-22 Gate Delay Portion of the Model 172 Module (Complete module shown in Fig. 4-9)

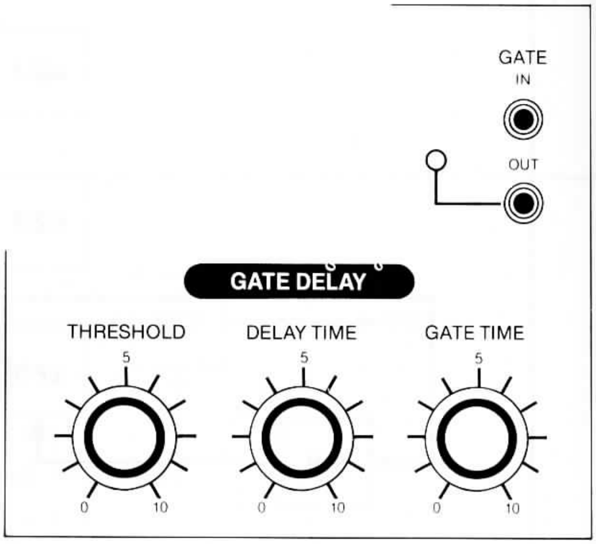
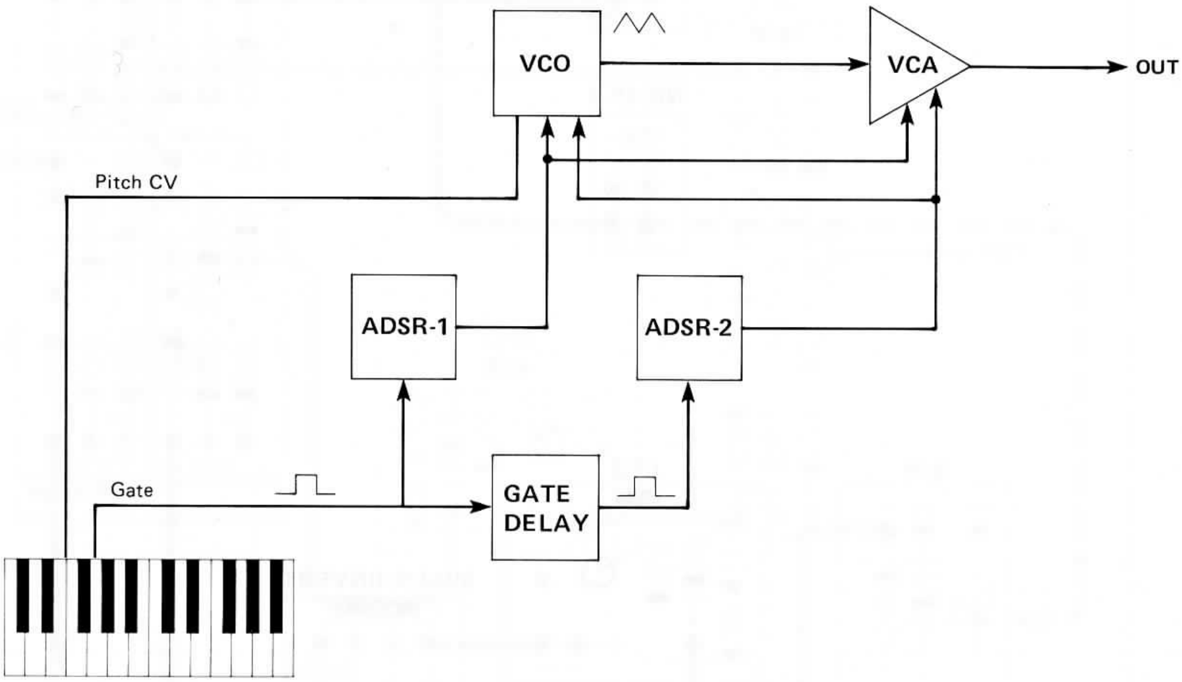
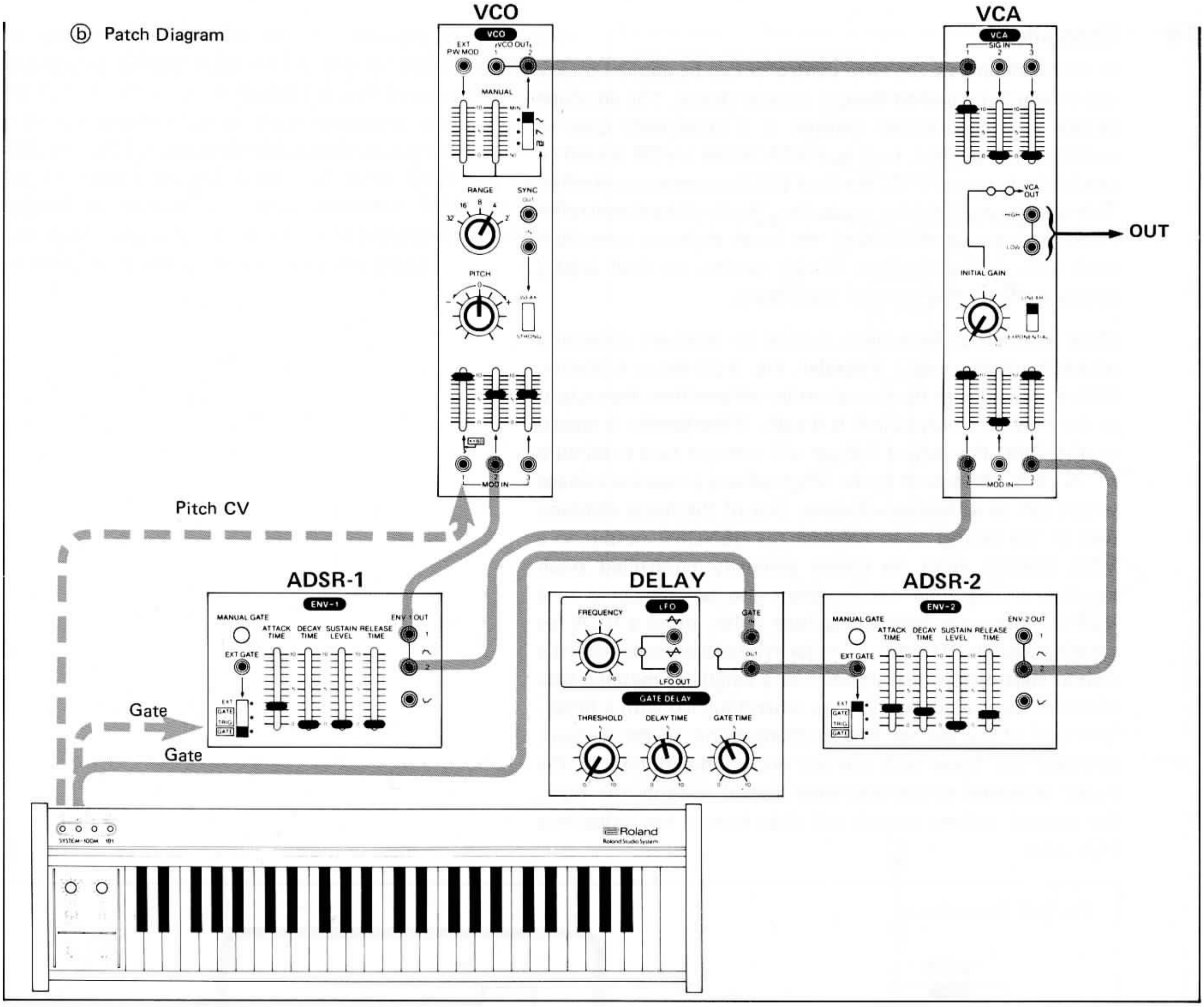


Fig. 3-23 WOLF WHISTLE (Gate Delay)

ⓐ Block Diagram

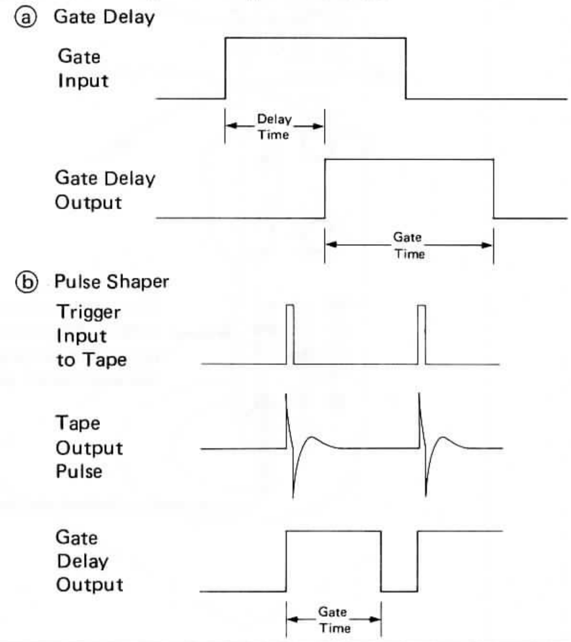


ⓑ Patch Diagram



Sometimes it is desirable to synchronize some device such as an analog sequencer (discussed later) by triggering it from pulses recorded on tape. Because of the nature of the recording medium, it is possible to record only very short clicks on tape to act as trigger pulses. The pulse shaper function lengthens these pulses so that they are more useful. Also, the output level of a tape recorder is usually quite low in comparison with normal synthesizer levels. The input of the gate delay has an amplifier stage which brings this level up to where it will trigger the gate delay. The THRESHOLD control determines the level of input which will trigger the gate delay. Normally, set the THREHOLD control at "10", then turn it slowly down towards "0" while inputting pulses from the tape. There will be a place where the gate delay is triggered for each input pulse. Above this setting, the gate delay will not be triggered and below this the gate delay may be triggered by random tape noise. The GATE TIME control should be set somewhere above "0", since at "0" the gate pulse may be too short to be noticed. The DELAY TIME control should, of course, be set at "0" unless a delay is desired. The pulse shaper waveforms are shown in Fig. 3-24 (b).

Fig. 3-24 Gate Delay/Pulse Shaper Waveforms

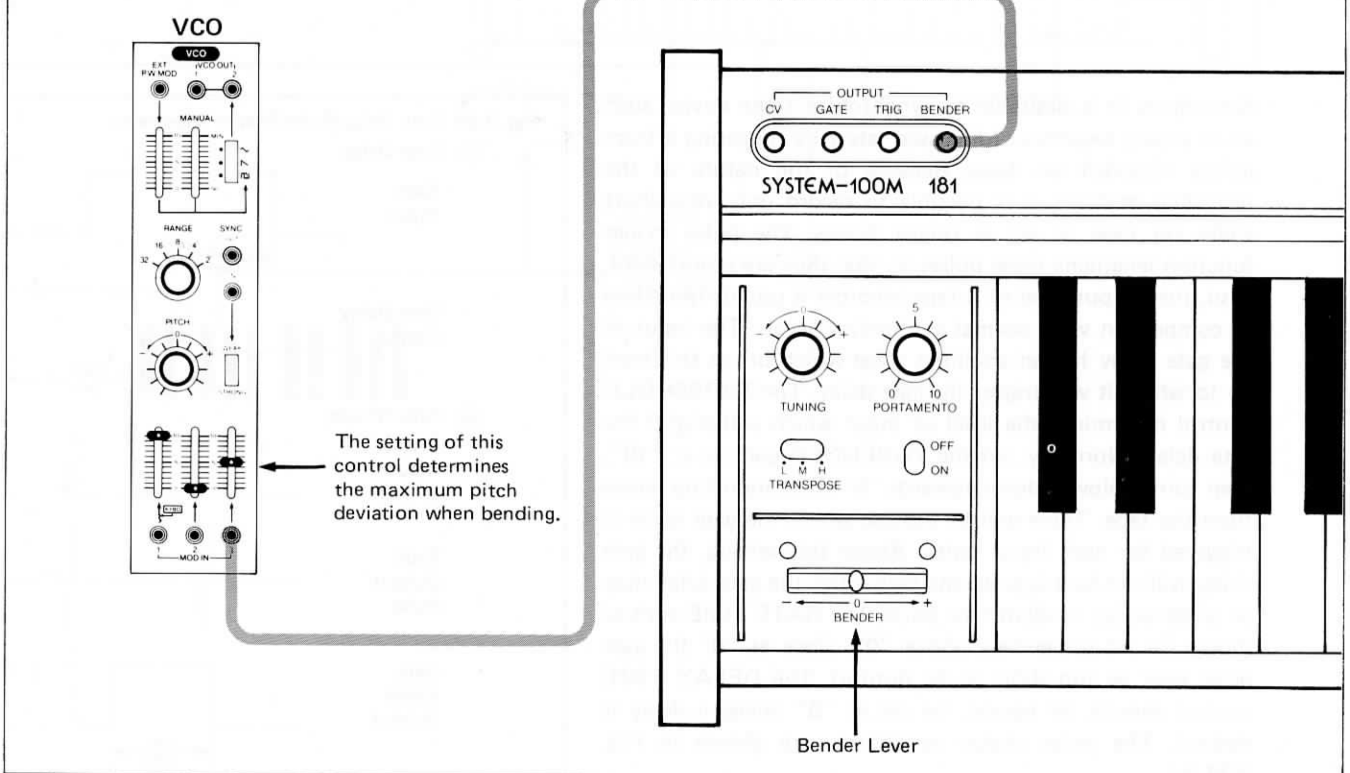


3-5 Controllers

In electronic music the term **controller** can be applied to any device which is used to control another device. The envelope generator is a controller because it is commonly used to control other devices, such as a VCA. When a VCO is used to modulate a second VCO, the first VCO becomes a controller. The sample and hold is a controller. The keyboard controller could perhaps be considered the most common controller used with the synthesizer. In this section we shall briefly consider a few other types of controllers.

Many synthesizer keyboards include an auxiliary controller which is usually called a **bender**. Fig. 3-25 shows a bender. With the bender at its normal center-off position, the output at the BENDER output jack is 0 volts. If the bender is moved to the right, the output voltage will increase to a maximum of +5 volts. Movement to the left produces a negative voltage which can go as low as −5 volts. One of the more common uses of the bender is to connect the BENDER output to a VCO control input to create manually controlled pitch bending effects. The bender could also be connected to a VCF for manually controlling tone color, or to a VCA for creating accents. There is a similar type of controller called a **ribbon controller** which consists of a length of metallic film which outputs a control voltage when touched with a finger. The level of the voltage will be proportional to the distance between the finger and the left end of the ribbon. If the finger is placed at the left, then moved towards the right, the control voltage output will slide from a low value to a high value.

Fig. 3-25 Bender Lever



The **joy stick controller** can be thought of as a two-dimensional bender lever which can be moved to the left and right ("x" axis), or backward and forward ("y" axis), or in any combination of these directions, as shown in Fig. 3-26. Fig. 3-27 shows a three-dimensional joy stick which can also be moved up and down ("z" axis). Each axis can be assigned to control a specific parameter. With the three-dimensional joy stick, it is possible to simultaneously control the pitch, tone color, and loudness of a sound.

Fig. 3-26 Joy Stick Controller

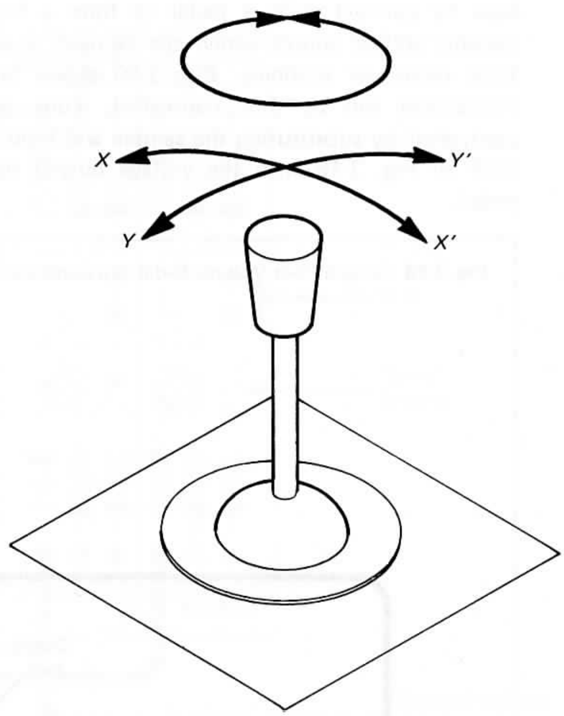
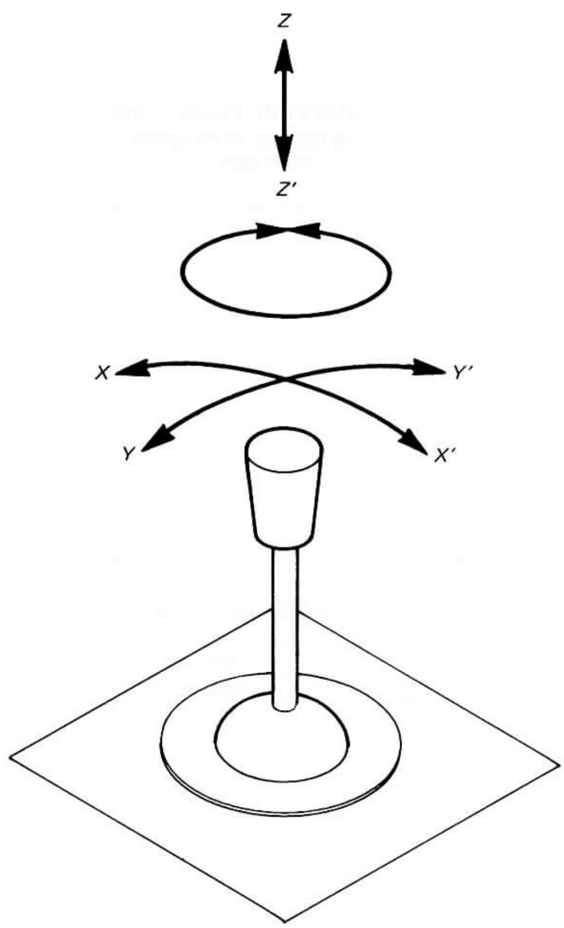


Fig. 3-27 Three-Dimensional Joy Stick Controller



The **foot volume control pedal** (such as the Roland FV-2) can also be used as a synthesizer controller. Fig. 3-28 shows how to connect such a pedal to form a foot-controlled variable voltage source which can be used to control pitch, tone color, or loudness. Fig. 3-29 shows how loudness (dynamics) can be thus controlled. Tone color can be controlled by substituting the sample and hold input to the VCF of Fig. 3-16 with the voltage output from the foot pedal.

Fig. 3-28 Using a Foot Volume Pedal as a Controller

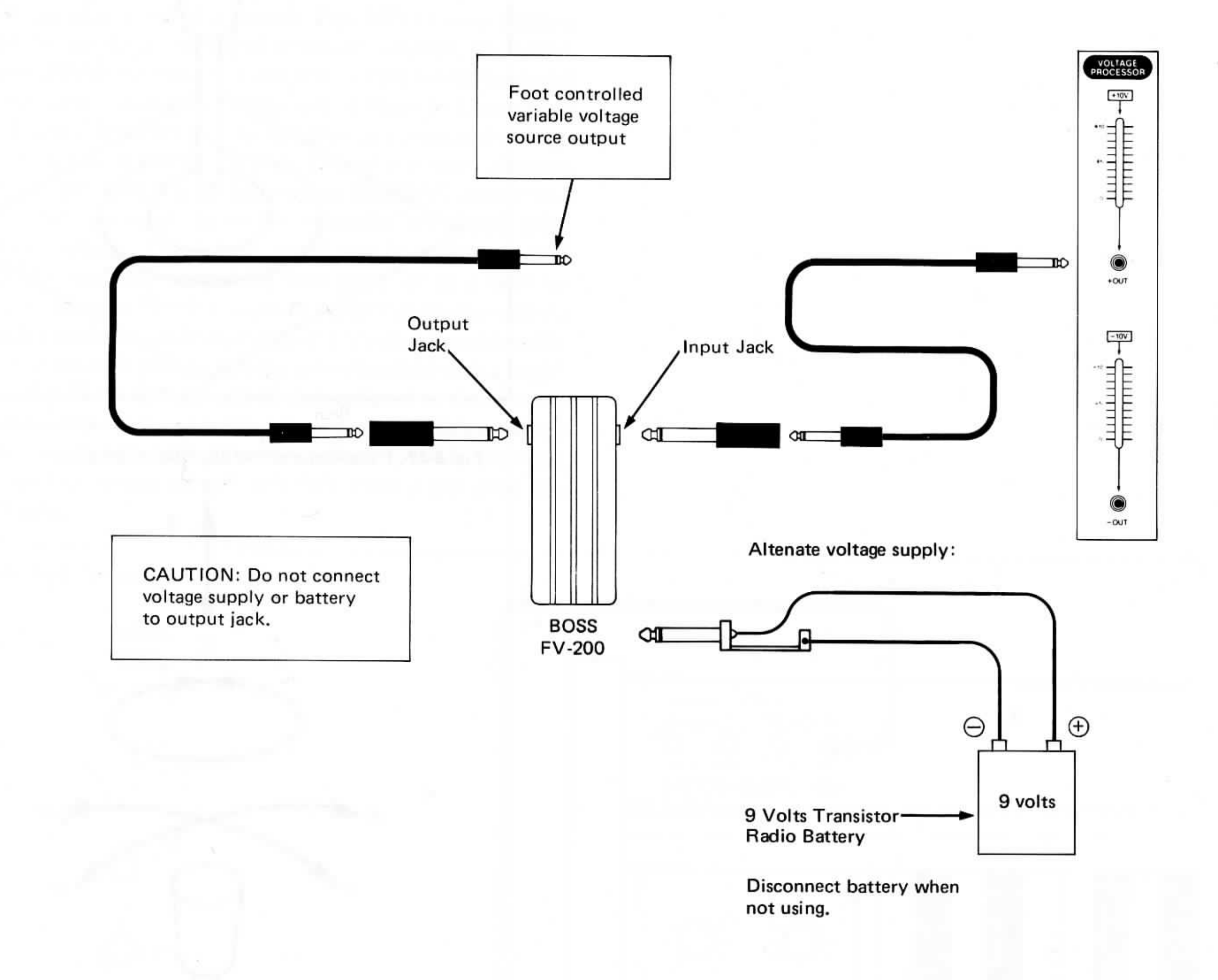
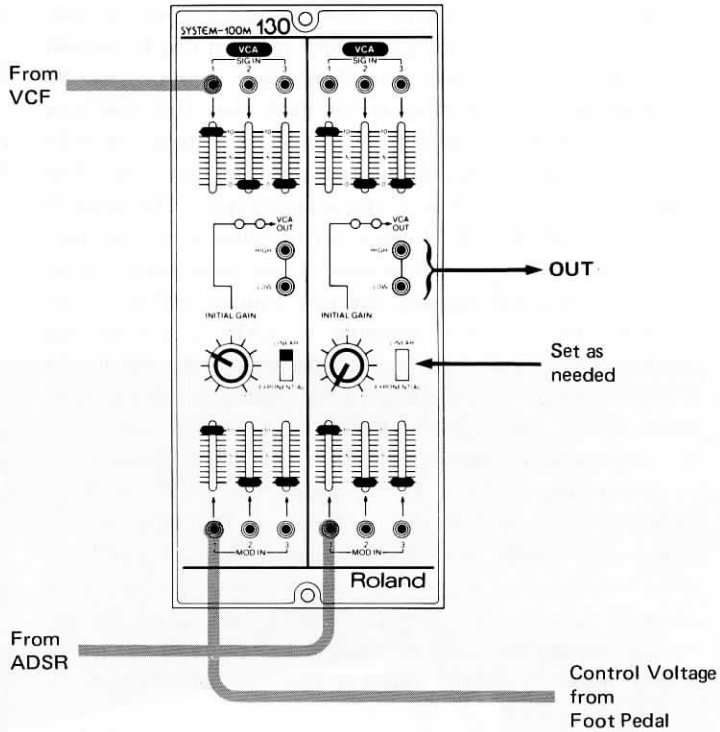
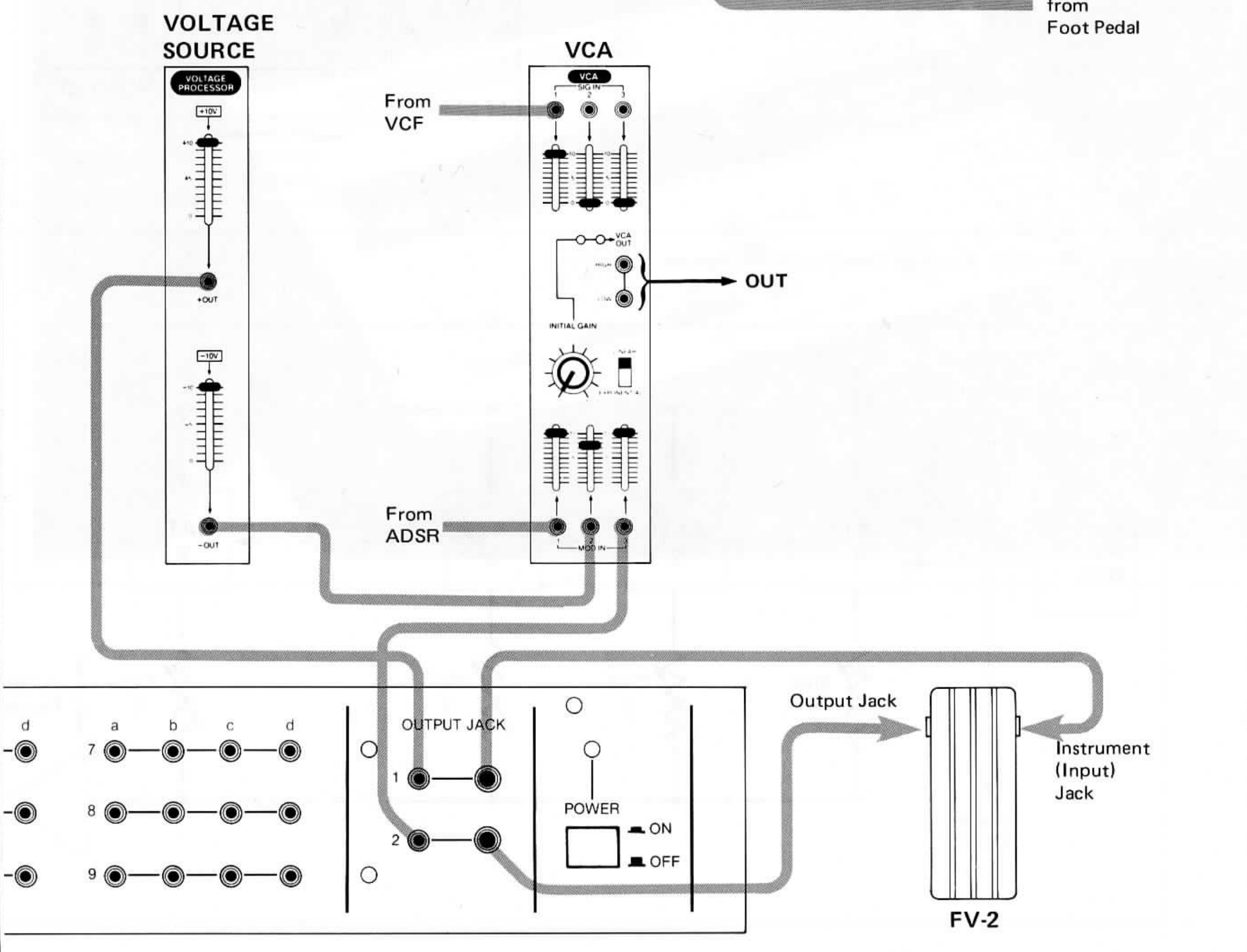


Fig. 3-29 Foot Control of Dynamics

ⓐ Using Two VCA's



ⓑ Using One VCA



The **analog sequencer** is another common type of controller used with the synthesizer. Fig. 3-30 shows a two-channel sequencer. The voltage registers determine the voltage output for each step in the sequence. When the series output is used, it is possible to produce a sequence of from one to sixteen steps. When the parallel output is used, it is possible to produce two voltage outputs for each step (for two-note chords, for example) with from one to eight steps. Fig. 3-31 shows a simplified block diagram for the sequencer. The clock is an LFO which generates a square wave. The series of blocks labeled "GATE" form a circuit called a ring counter. Only one of the gates will be open at any given instant. Each time a clock pulse appears, the ring counter will shift one place to the right. For example, if GATE 2 is open, the voltage source will pass through the gate and through the STEP 2 voltage register, then to the output. When a clock pulse appears, GATE 2 will close and GATE 3 will open, thus the voltage at the output will change so that it corresponds to the setting of the STEP 3 voltage register. If the clock oscillator is allowed to run continuously, the ring counter will run in circles so that the sequence repeats itself. The most obvious use for the sequencer would be to run a melodic pattern such as an arpeggio, in which case the CV output would be used to control a VCO and the clock output would be used to trigger an envelope generator.

Fig. 3-30 Analog Sequencer Panel Diagram

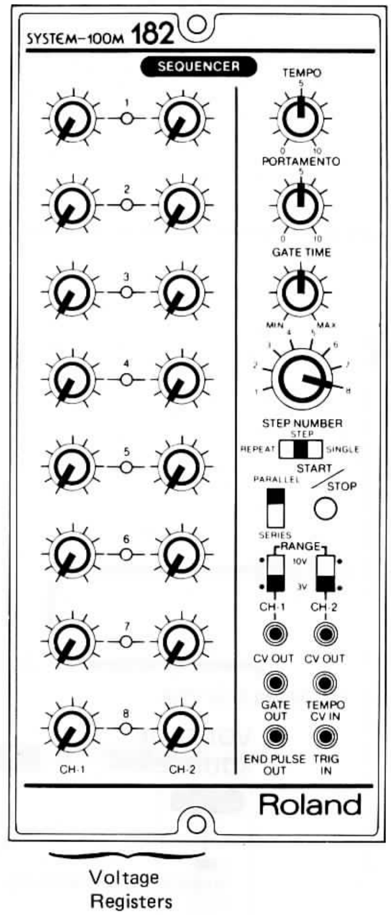
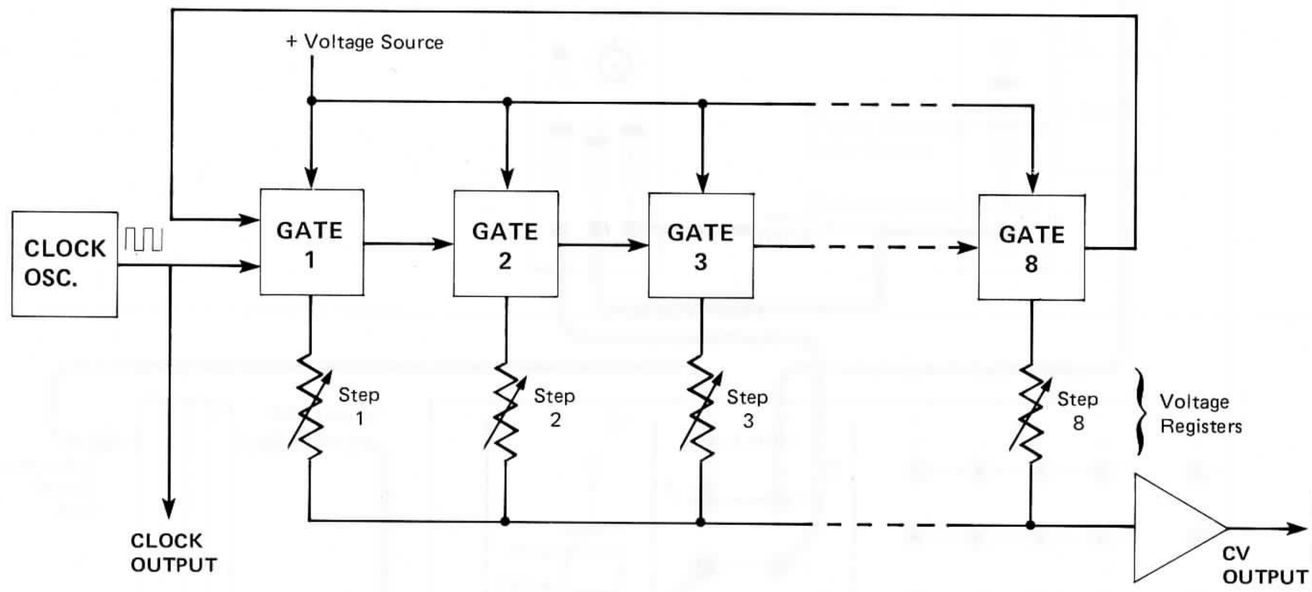


Fig. 3-31 Analog Sequencer Simplified Block Diagram



The control of a synthesizer by means of a computer gives the musician complete and absolute control over the music being produced. There are two basic approaches to **computer control**. The first is to use a computer especially designed for the control of a synthesizer, such as the Roland MC-4 Micro-Composer (Fig. 3-32). The second method is to adapt an ordinary computer to this type of control. This requires intimate knowledge of computer programming, and usually involes the use of custom-made interface devices for converting the output of the computer into control voltages and pulses for controlling the synthesizer.

Fig. 3-32 Roland MC-4 MicroComposer (Computer Controller)

Chapter Four:
Accessories Used in Synthesis

4-1 Introduction

The voltage controlled filter, being a dynamic filter, is most often used for controlling changes which occur in tone color during the production of notes. It must be remembered that most sound sources also contain tone color elements which remain relatively constant, such as those introduced into a sound by the body of a musical instrument. This is why, in trying to reproduce many kinds of sounds, many synthesized sounds seem inadequate unless one is striving for purely electronic sounds. Imitation of acoustic instruments, whether real or imaginary, often demands more than just the synthesizer elements discussed so far.

Most of the devices to be discussed in this chapter are used in conventional music studios as effects devices, or as corrective devices to compensate for some fault in the recording chain. In electronic music, of course, these devices can serve a similar purpose, but in this chapter we shall look at these devices primarily as aids to synthesis.

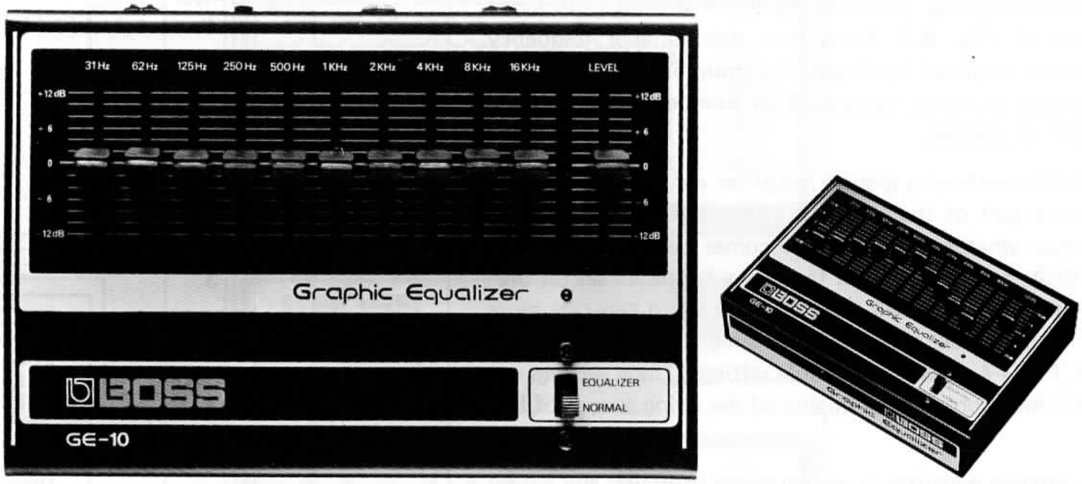
4-2 The Graphic Equalizer

In the conventional music studio the **graphic equalizer** is used, more often than not, as a corrective device. Most literature on the subject seems to take a justifiably conservative attitude on their use, the idea being that if we have to use equalization to make a flute sound like a flute, something must be wrong somewhere in the recording chain. In some cases, equalizers can be very useful. As one example, equalization can help to clarify instrumental voices in a "muddy" passage, but still it is very easy to contract "equalization fever" in which equalization is applied to everything abundantly and with abandon so that the recording no longer resembles what was originally desired.

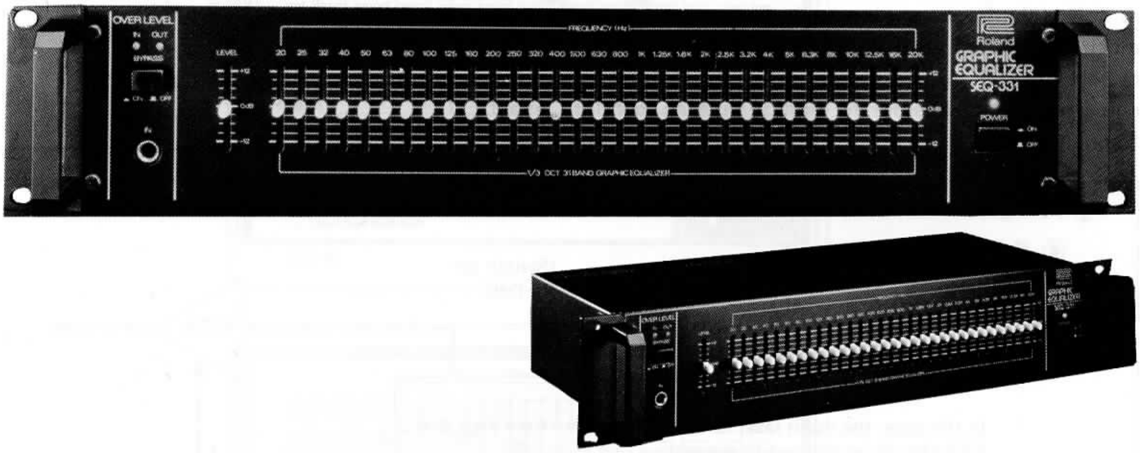
In electronic music we may, more or less, ignore the conservative attitude of conventional music recording studios because the graphic equalizer is a very important tool in the process of synthesis. It is so important, in fact, that the graphic equalizer can be considered second only to a reverberation unit on the list of priorities for accessories for the new electronic music studio. Nevertheless, "equalization fever" is still easy to contract and care should be taken not to overdo it. Too much of a good thing is still too much.

Fig. 4-1 Graphic Equalizers

ⓐ Boss GE-10 (10 Bands)



ⓑ The Roland Rack Series SEQ-331 (31 Bands)



ⓒ The Roland Rack Series SEQ-315 (Stereo; 15 Bands each Channel)

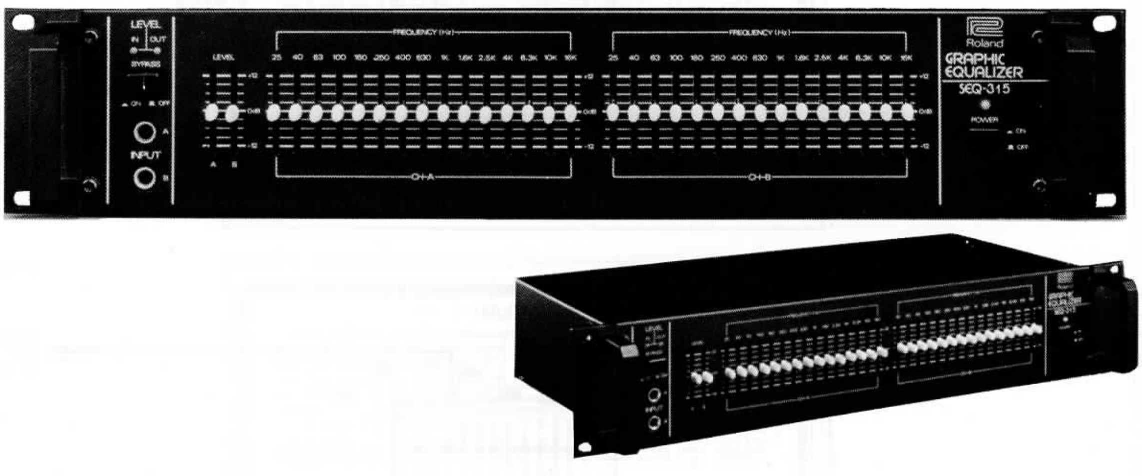


Fig. 4-1 shows some graphic equalizers. The name graphic equalizer comes from the fact that when the sliders on the front panel are set, they give a graphic representation of the frequency response produced by those settings. The block diagram of Fig. 4-2 shows that this variable frequency response is obtained by means of a group of band pass/reject filters (pass or reject, depending on whether the slider is up or down) in parallel.

Fig. 4-3 shows how a graphic equalizer can be connected to become a part of the synthesis chain. Often it makes little difference whether the equalizer comes before or after the final VCA, but if the higher frequency bands are set for high levels, the equalizer will have a tendency to generate more noise than normal, thus the arrangement in (a) may prove better. Fig. 4-4 shows four sample settings. These settings can be used for improving the quality of the string sounds of Fig. 2-2.

A **parametric equalizer** is an equalizer in which the center frequency of each band and the band width are adjustable, as well as the frequency levels. With the graphic equalizer we must be satisfied with adjusting the level of the frequency closest to the desired point, whereas with a parametric equalizer, it is possible to fine tune each band to exact frequencies.

Fig. 4-2 Graphic Equalizer Block Diagram

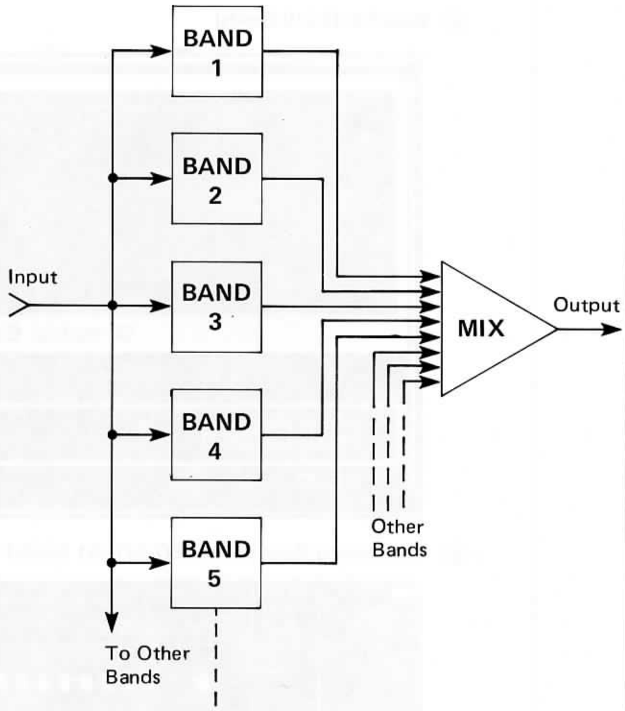


Fig. 4-3 Equalizer Connections

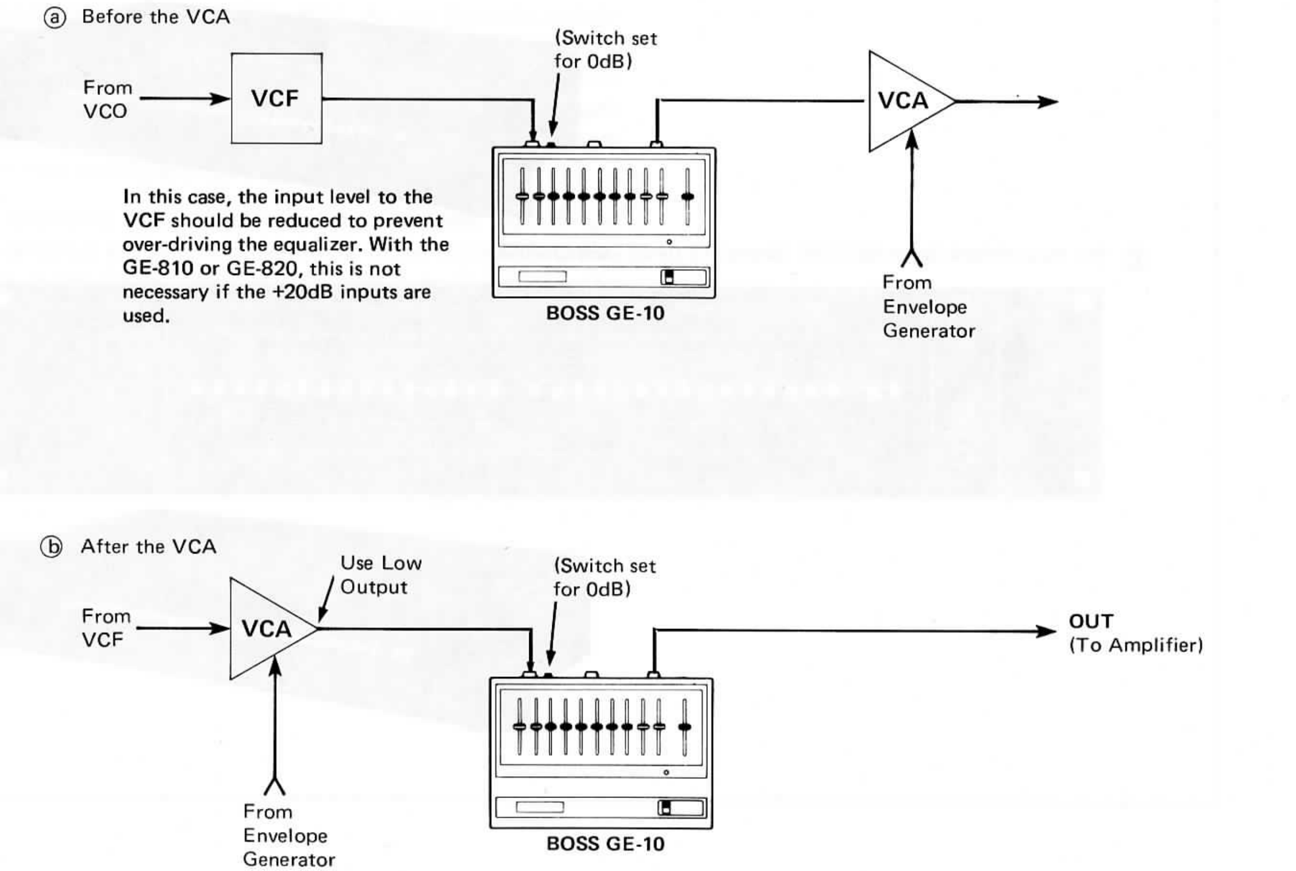
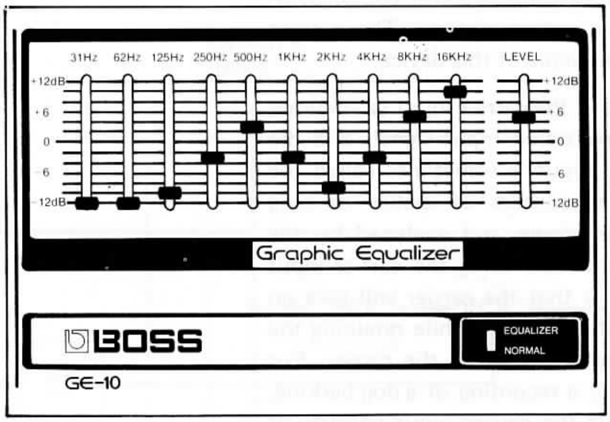
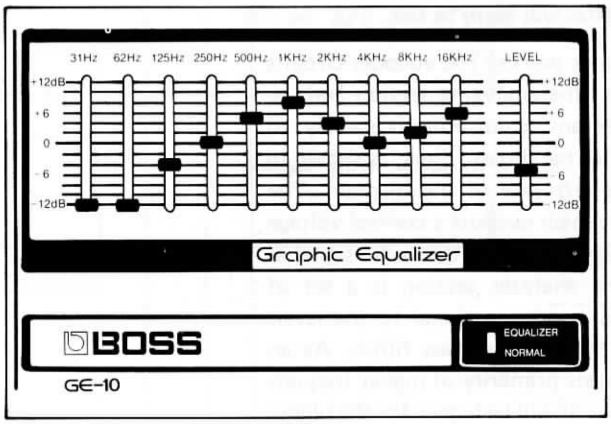


Fig. 4-4 Typical Equalizer Settings for String Sounds

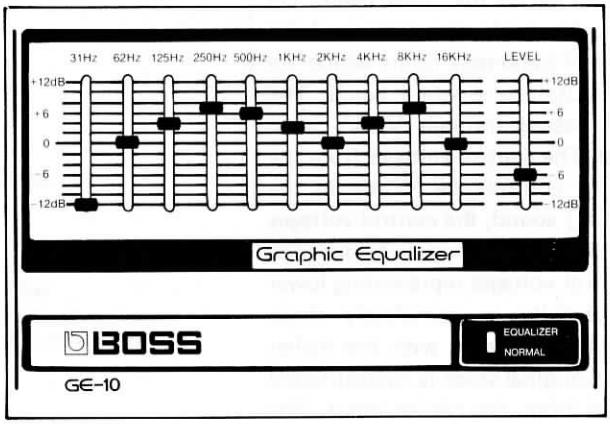
ⓐ Violin



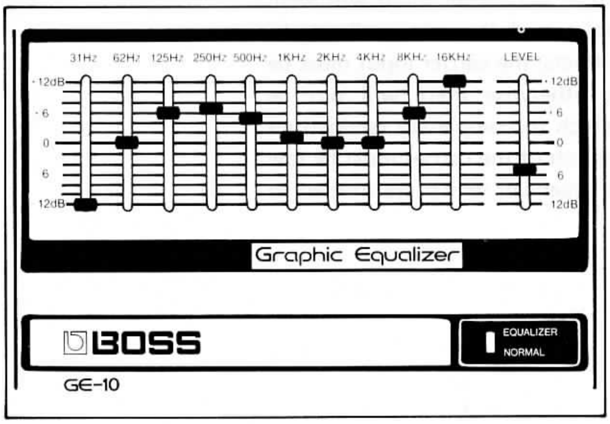
ⓑ Viola
(Change VCF CUTOFF FREQUENCY to "6".)



ⓒ 'Cello



ⓓ Bass



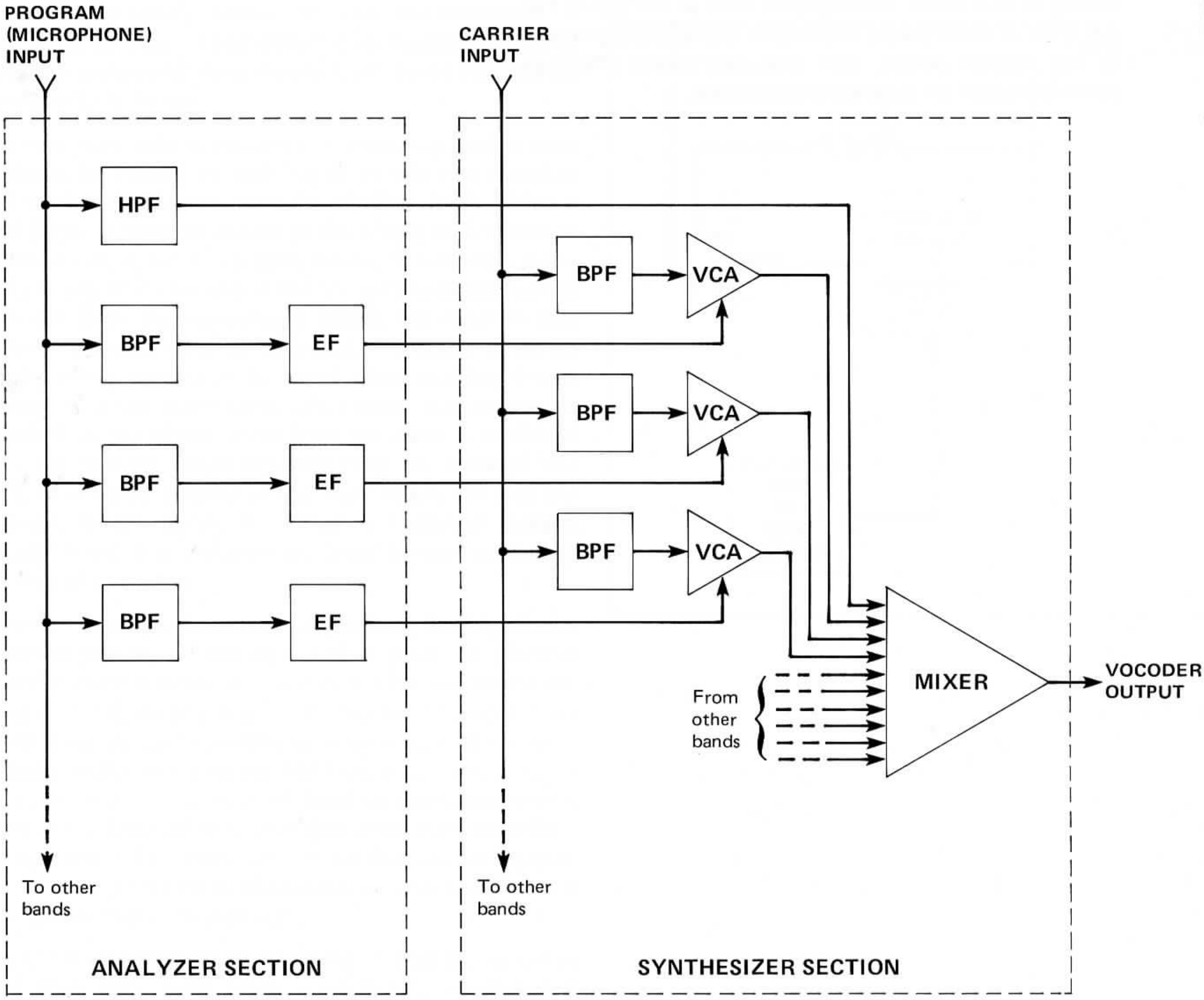
4-3 The Vocoder

In 1939, H. Dudlay announced a band width compression device for use in telecommunication systems. The present day vocoder is based on the principle of this device.

The **vocoder** requires two inputs: the **carrier** input (sometimes called the **replacement** or **excitation** input signal) and the **program** input. The program input is sometimes called the **speech** input because it often consists of spoken or sung words. This input is broken down and analyzed by the vocoder circuits, then reconstructed using the carrier input for raw material. The result is that the carrier will take on the qualities of human speech or singing while retaining the original pitch and tone color character of the carrier. For example, if the carrier input is a recording of a dog barking, the dog will seem to talk. If the carrier input consists of recorded street noises, the traffic will seem to talk.

Fig. 4-5 shows how the vocoder works. The vocoder circuits consist of two major sections: the analyzing section and the synthesizing section. The program or voice input is analyzed by passing it through a series of band-pass filters, as shown in the analyzer section on the left side of the diagram. The envelope follower is a circuit which outputs a control voltage whose level is proportional to the level of the audio signal at its input. The output of the analyzer section is a set of control voltages whose levels are proportional to the levels of the signals being passed by the band-pass filters. As an example, the sound [*ee*] consists primarily of higher frequencies, thus the control voltage levels will be higher for the higher frequency bands than they will be for the lower bands. On the right side of the diagram, the synthesizer section of the vocoder also contains a series of band-pass filters which are used to divide the carrier input into separate bands. The outputs of these filters are passed through VCA's, then mixed for the vocoder output. The control voltages from the analyzing section are used to control the VCA's in the synthesizer section. With the [*ee*] sound, the control voltages from the analyzing section that represent higher frequencies will be higher than those control voltages representing lower frequencies thus opening partially or completely those synthesizer section filter VCA's associated with the higher frequencies. By this means, the original voice is reconstructed using the tone color material from the carrier input. The sound of the instrument used as the carrier input will seem to speak or pronounce the [*ee*] sound. It can immediately be seen that except for special effects, the carrier input must be a sound rich in harmonics, otherwise there will not be enough raw material from which to completely reconstruct the program input. Also, the pitch of the carrier must be low enough to provide the low frequencies which are needed to build the new sound.

Fig. 4-5 Vocoder Block Diagram



HPF = High Pass Filter (See text)
BPF = Band Pass Filter
EF = Envelope Follower
VCA = Voltage Controlled Amplifier

Most carrier inputs will not contain enough of the higher frequencies to reproduce accurate consonant sounds such as the hissing of an *s* sound. The high-pass filter at the top of Fig. 4-5 is set so that it passes these higher hissing sounds directly to the vocoder output, thus producing speech effects in which the words are more easily understood.

4-4 Phase Shifters and Flangers

Phase shifters and **flangers** (audio delay lines) are more often thought of as effects devices, but they can sometimes be used in the synthesis of sounds with a complicated spectrum. To better understand them, however, we should first look at them as effects devices.

The first such effects produced in recording studios were produced by feeding an audio signal to two tape recorders and mixing the outputs from the playback heads. When a light finger pressure is applied to the supply reel on one of the machines, it will run slightly slower, thus giving a slight delay to one of the outputs. When this slightly delayed sound is mixed with the non-delayed sound, the result is that certain frequencies will be cancelled. The effect is similar to the effect which can be heard when standing near a runway as a jet plane takes off. Certain frequencies are cancelled as the direct sound from the plane is combined with the delayed sound reflected from the runway. The amount of delay changes as the plane moves off into the distance. In the studio, this effect is known as **flanging** because it was first produced by finger pressure applied to the flage of a tape reel.

In earlier attempts to electronically imitate flanging effects, electronic phasing systems were used. In these, the sound is passed through a series of phase shift networks, then combined with the original sound. The number of cancellations which occur due to this combination depends on the number of phase shift networks used. The frequency ratios between the cancellations are a result of circuit constants and retain a constant harmonic relation when the cancellations are shifted up and down. The result, then, is not the same as flanging, but in more recent times phasing has become accepted as a very useful effect in its own right.

Modern digital technology has made it possible to design solid state delay lines which more closely resemble the original tape flanging effects.

The block diagram in Fig. 4-7 can represent either a phase shifter or a flanger, depending on whether the block in the center of the diagram is a phase shift network or an audio delay line. The shift/delay control is usually a low frequency sine wave oscillator. In the electronic music studio, the most useful units for synthesis of sound are those which allow optional external control of the function by means of an external control voltage. Fig. 4-8 shows such units. The three jacks near the bottom of the panels are for the external control input. Fig. 4-9 shows the phase shifter and audio delay designed for the System 100M. At a quick glance, phase shifters and audio delays like these seem to be exactly alike, but they are different externally in that one has a SHIFT FREQUENCY control while the other has a DELAY TIME control. In synthesis, they would be patched into the system in exactly the same manner.

Fig. 4-7 Block Diagram for Phase Shifter or Flanger

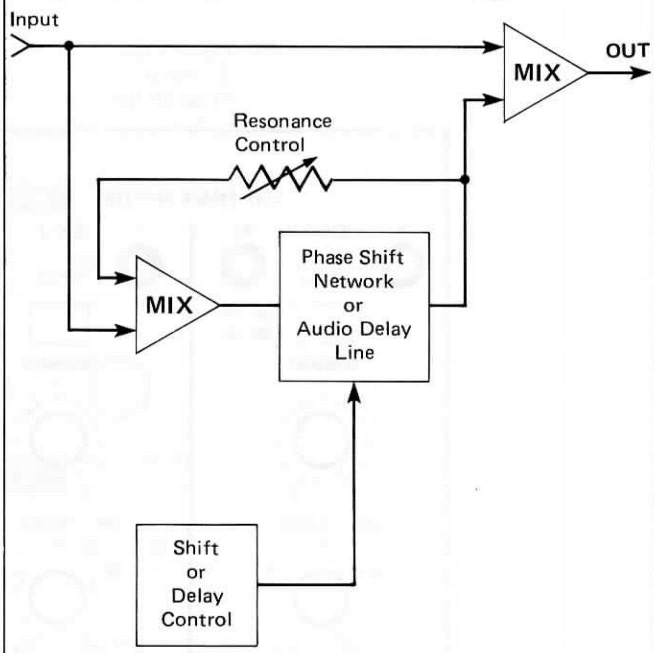


Fig. 4-8 Roland System 700 Phase Shifter and Audio Delay

(a). Model 720B
2-Channel
Phase Shifter

(b). Model 721A
2-Channel
Audio Delay

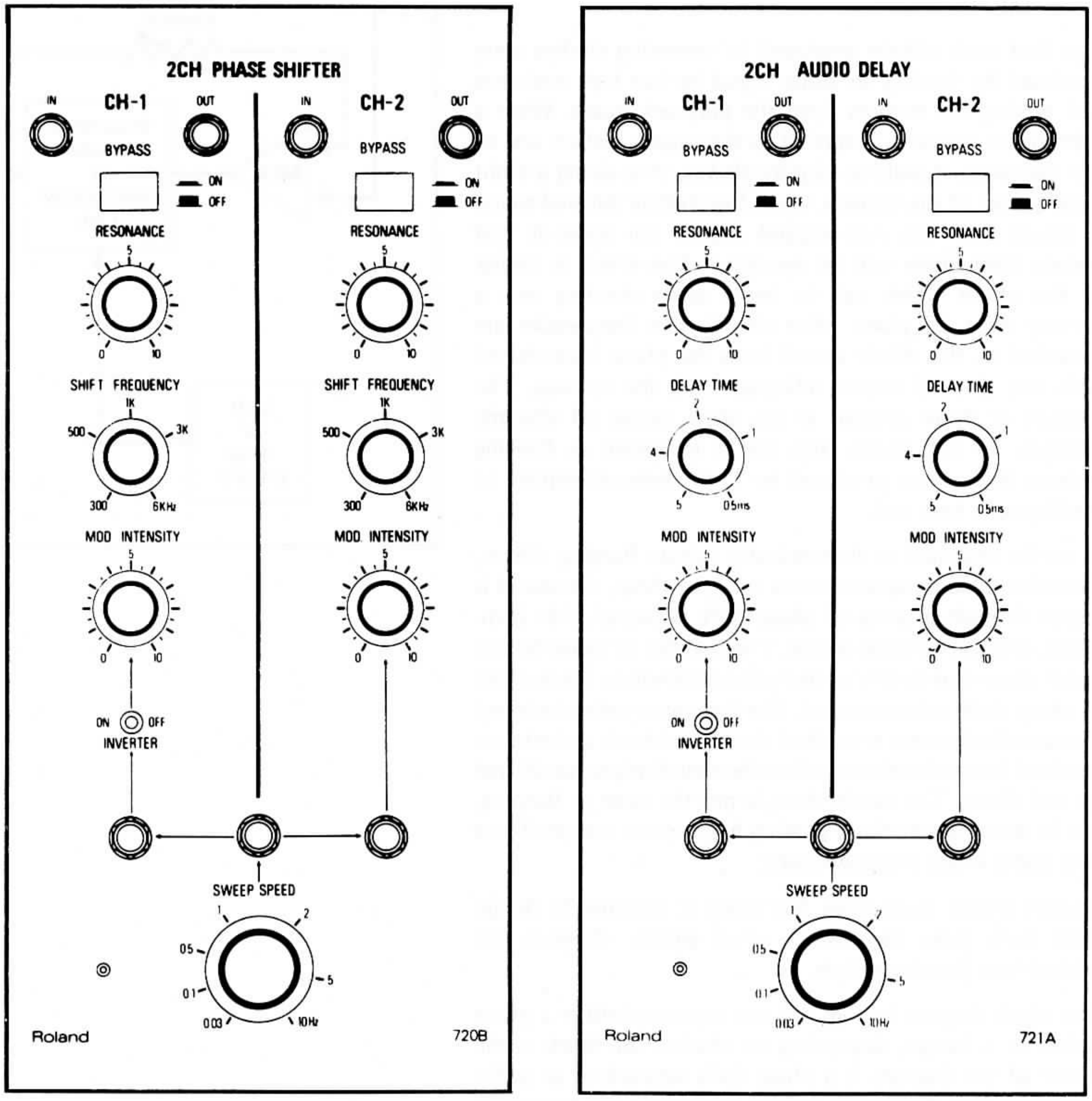


Fig. 4-9 System 100M Phase Shifter and Audio Delay

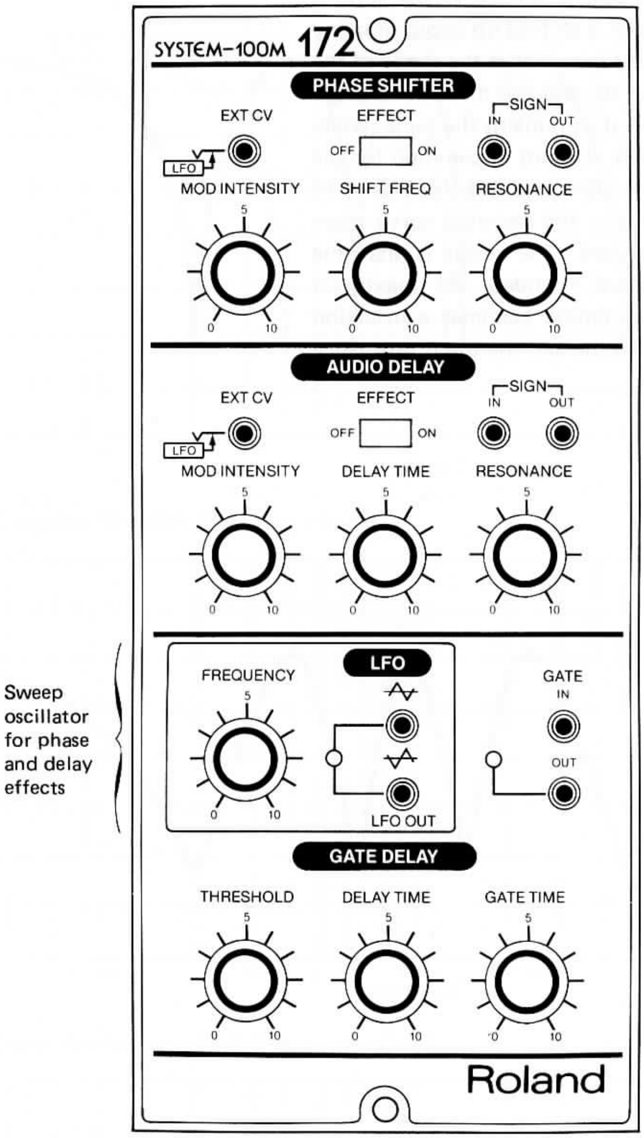
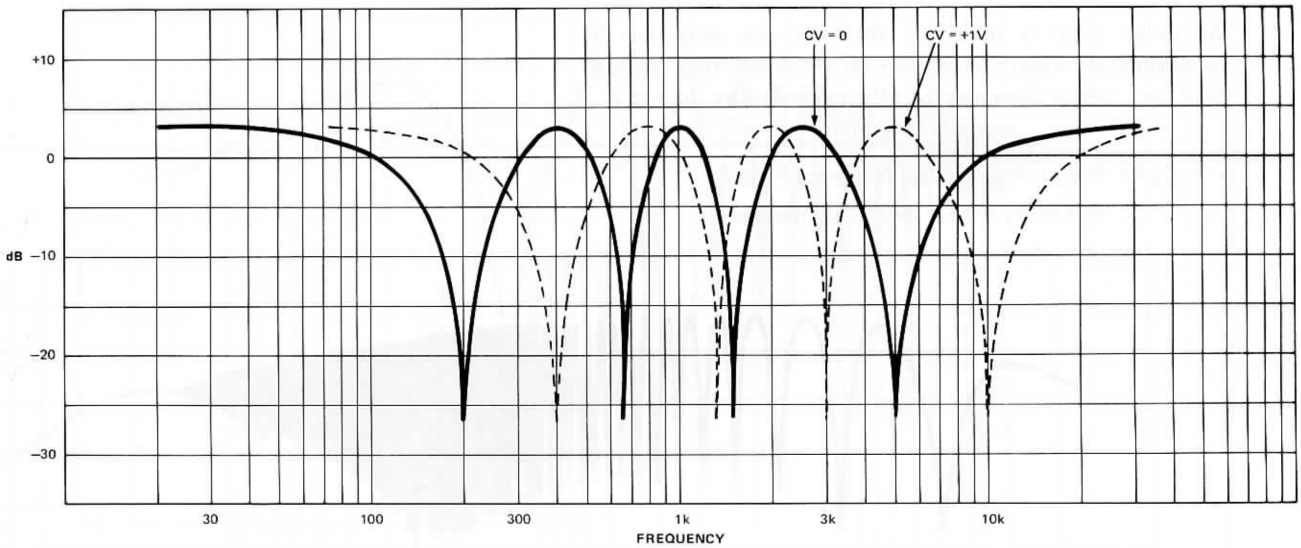


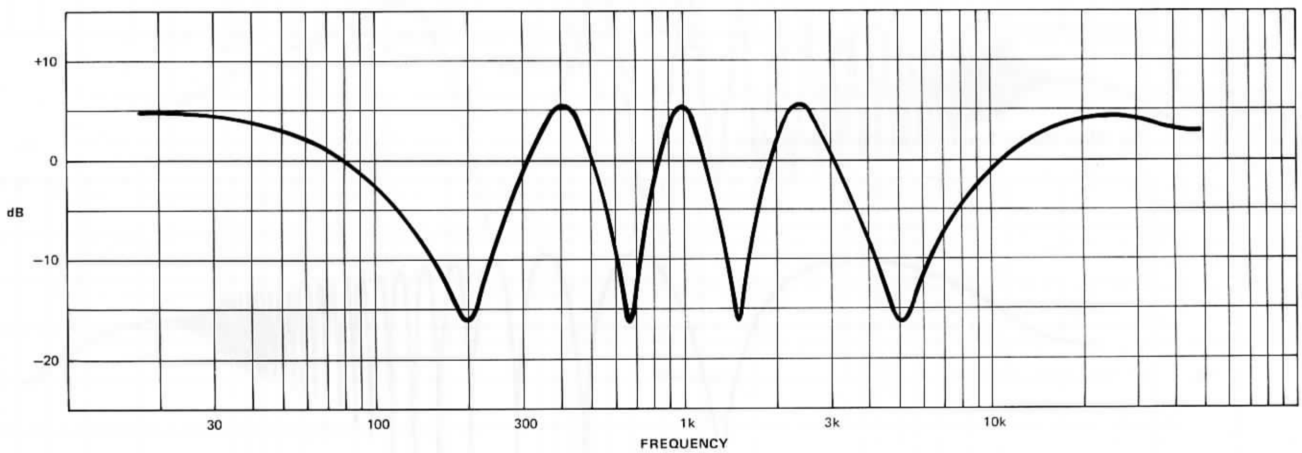
Fig. 4-10 (a) shows the frequency response of a typical phase shifter. The solid line shows the response with no control voltage input and the SHIFT FREQUENCY control set at 1kHz (note that the highest portion of the center hump occurs at 1kHz). The dotted line shows the response when an external control voltage of +1 volt is applied to the control input. The entire response curve is shifted up approximately one octave. The important point here is that the shape of the curve changes very little, if at all, and the distance in *pitch* between each of the peaks and dips remains the same. When using the phase shift as an effect, it is not uncommon for the center hump to sweep between 30Hz and 10kHz. Figs. 4-10 (b) and (c) show what happens to the response curve when the RESONANCE control is used. The result is that the humps and dips become more rounded. At maximum resonance, the response curve almost becomes a inversion of the normal response curve. To the ear, the use if resonance enhances the phase shift effect.

Fig. 4-10 Phase Shifter Frequency Response

ⓐ With Shift Frequency Control set at 1kHz.



ⓑ With Resonance Control at "5" (Shift Frequency = 1kHz)



ⓒ With Resonance Control at Maximum (Shift Frequency = 1kHz)

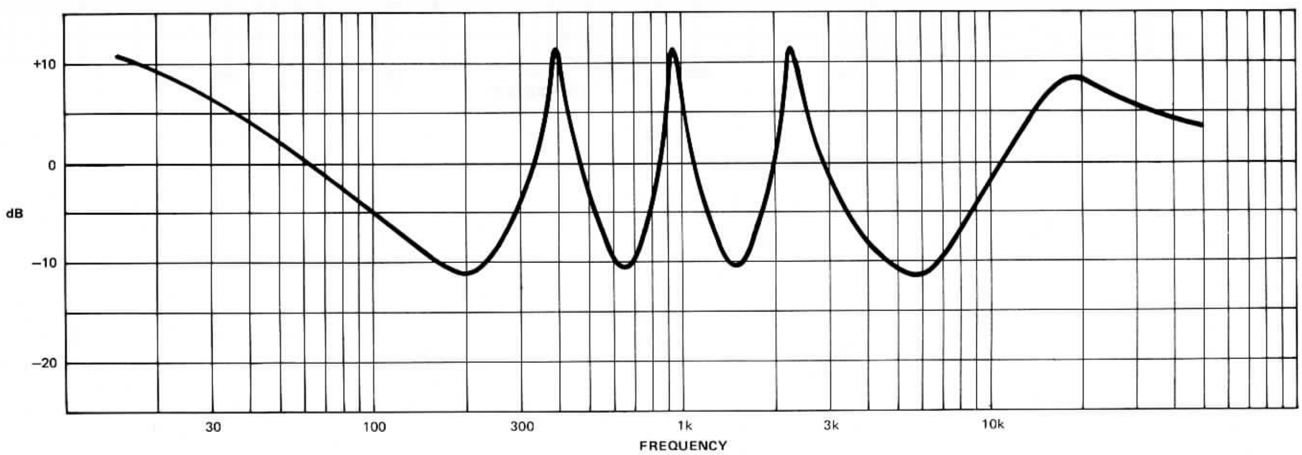
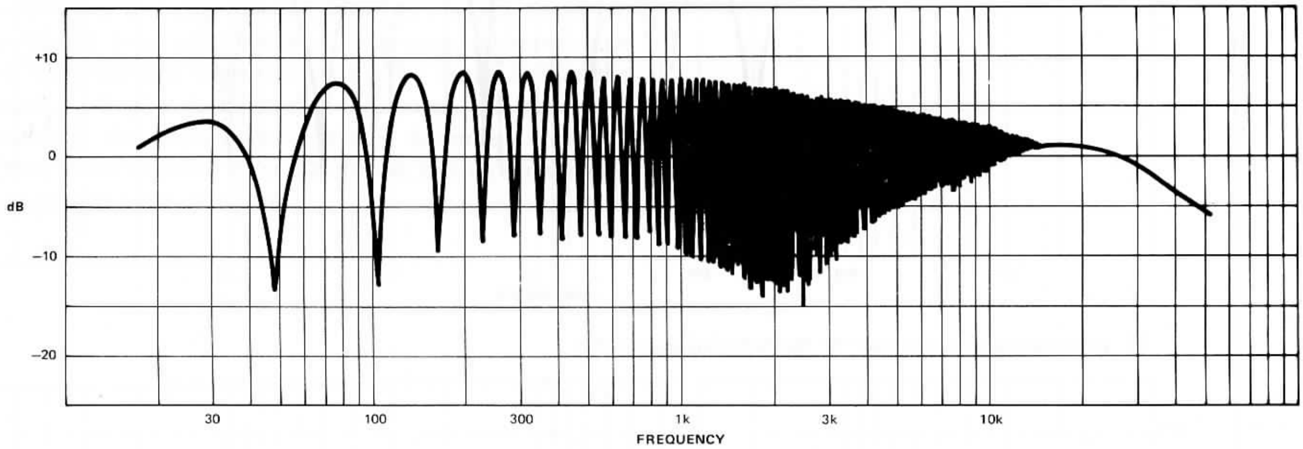


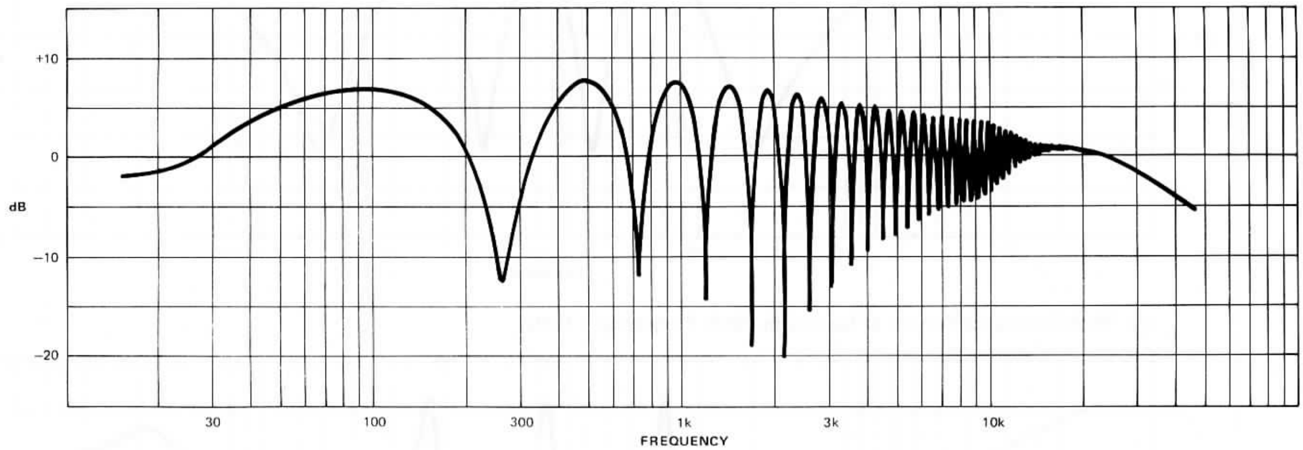
Fig. 4-11 shows the frequency response of a typical flanger. Note that the dips (cancellations) in the response curve occur at regular *frequency* intervals. In (a), the distance between any two adjacent dips is approximately 50Hz. In (b), where the delay time is different, the dips are separated by approximately 470Hz. In (c) and (d) are shown the effects of resonance, which enhances the effect as heard by the ear.

Fig. 4-11 Flanger (Audio Delay) Frequency Response

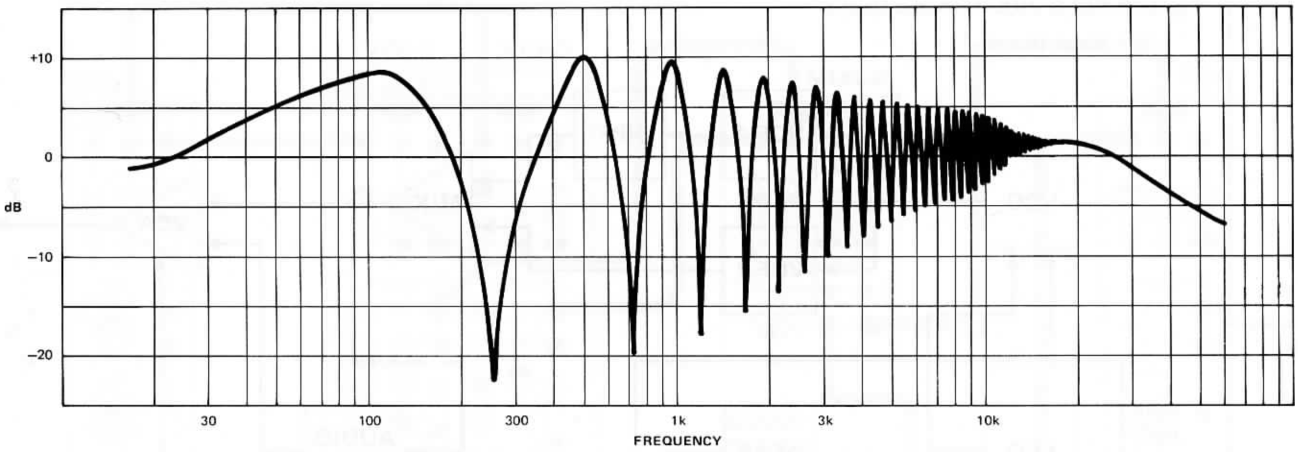
ⓐ With Delay Time = 16ms* (No Resonance)



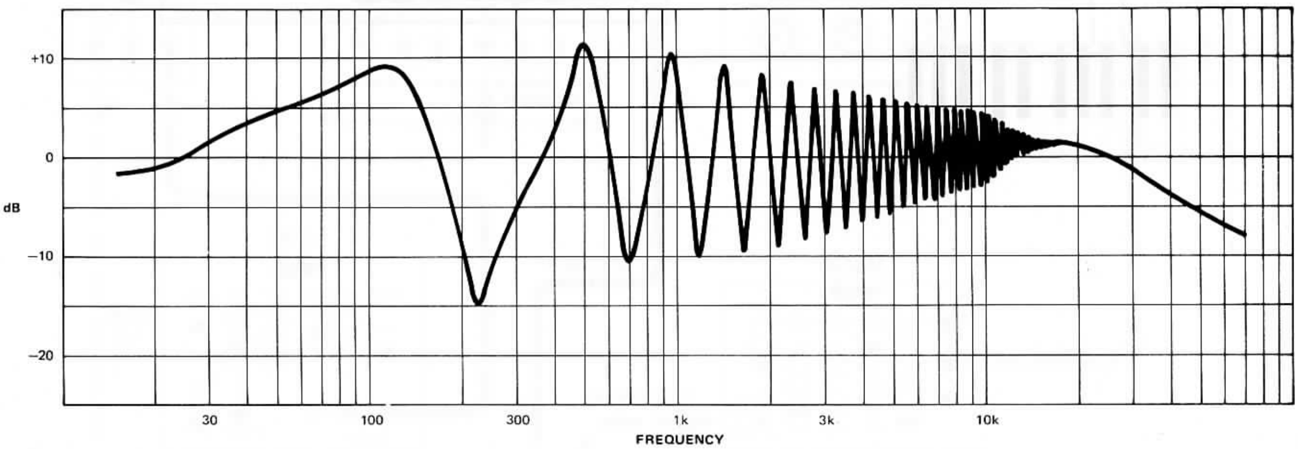
ⓑ With Delay Time = 2ms (No Resonance)



Ⓒ With Resonance at "5" (Delay Time = 2ms)



Ⓓ With Resonance at Maximum (Delay Time = 2ms)

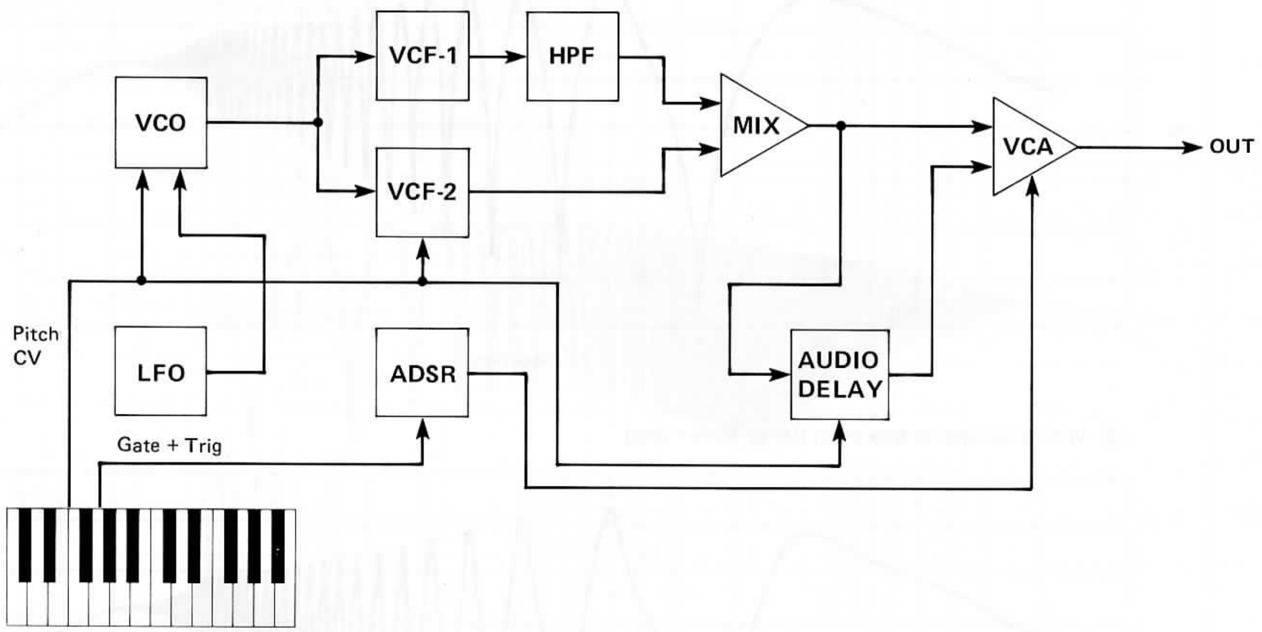


* ms = millisecond; 1 millisecond = 0.001 second

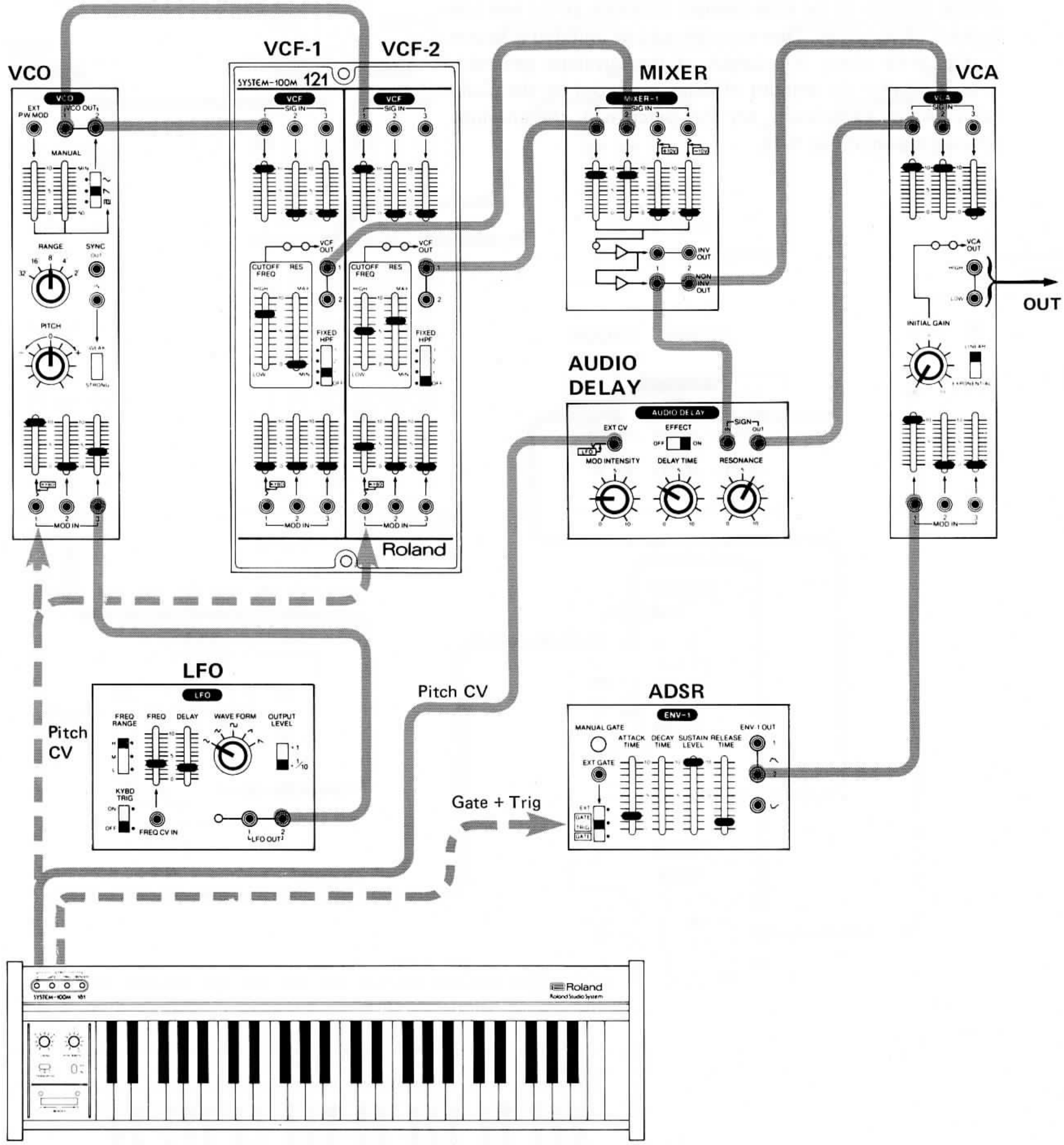
Fig. 4-12 shows an example of the use of an audio delay unit for the synthesis of a solo violin sound.

Fig. 4-12 SOLO VIOLIN (Audio Delay)

ⓐ Block Diagram



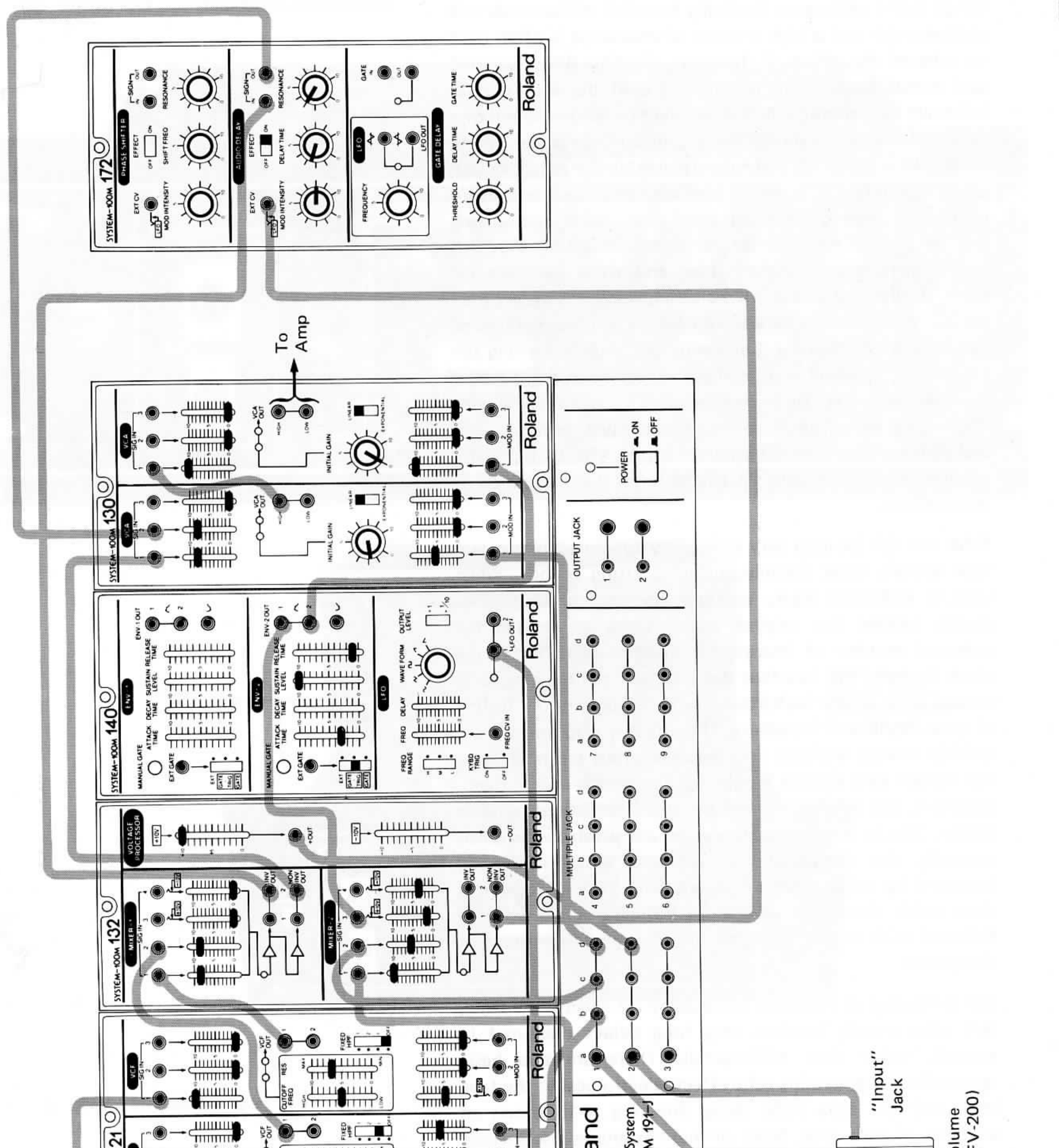
ⓑ Patch Diagram



One of the factors which determines the tone color produced by a violin is the speed and pressure of the bow as it passes across the string. It follows, then, that the tone color will change slightly as the bow changes direction at the top and bottom of its stroke. One way this can be imitated is shown in Fig. 4-13 where the output of the envelope generator is used to alter the amount of delay produced by the audio delay. As an experiment, try this patch with the envelope settings shown in Fig. 4-14.

Fig. 4-15 shows what might be termed a "complete" violin patch. It uses two VCF's and an audio delay to help produce the correct tone color. A foot pedal is used to control the loudness of the sound. The loudness, pitch, and envelope all affect the tone color.

Fig. 4-15 Complete **SOLO VIOLIN** Sound



4-5 Chorus and Echo

Normally with audio delay effects, the time delay is slowly varied over a wide range (typically from 0.5 milliseconds to 5 milliseconds) and a high amount of resonance is often used to enhance the effect. If, however, the time delay is varied over a small range and no resonance is used, the result is what is known as a **chorus** effect. The effect causes the sound of a solo instrument to sound like a group of the same instruments. In a group of instruments such as the string section of an orchestra, it is highly unlikely that each and every member is playing the exact same pitch; some instruments will be slightly flat and others slightly sharp. The overall sound, then, extends slightly above and below some average pitch to give the sound of the string section a very broad feeling. A fixed delay time would merely produce a sound of the same pitch, delayed. Constantly but slightly varying the time delay, however, causes slight variations in the pitch of the delayed sound. These slight variations, when mixed with the original sound, produce the chorus effect. Most flangers and audio delay units can generate chorus effects, but there are also machines designed particularly for the production of these effects.

Echo can also be used very effectively for broadening group type sounds, most notably sustained string sounds. When used in sustaining legato passages, an echo which follows closely behind the original sound tends to double the apparent number of instruments in the sound. When the pitch changes, the fact that the original pitch remains for a second or so in the background further increases the feeling of great depth and broadness. The effect is lost, however, in quickly moving melodic lines because often the pitches do not remain long enough for the echo to "catch up" to them, therefore, the original sounds are rarely reinforced by their echoes. This in itself, though, can be a desireable effect. For example, the complexity of arpeggio passages can be increased by using echo; or, as another example, imagine a short quick phrase played using a flute-like sound which is followed with a long rest filled with decaying repetitions of that phrase.

The principles of echo and audio delay are exactly the same, but echo usually involves very long delays measured in seconds rather than milliseconds. There are two basic approaches to generating echo effects electronically: the tape echo and the solid state delay line. Fig. 4-16 shows an example of each type. Both of these machines also include provisions for generation of chorus effects.

Fig. 4-16 Chorus and Echo Machines

ⓐ Roland SRE-555 (Tape Echo; with Chorus)

ⓑ Roland SDE-3000 (Digital Delay)

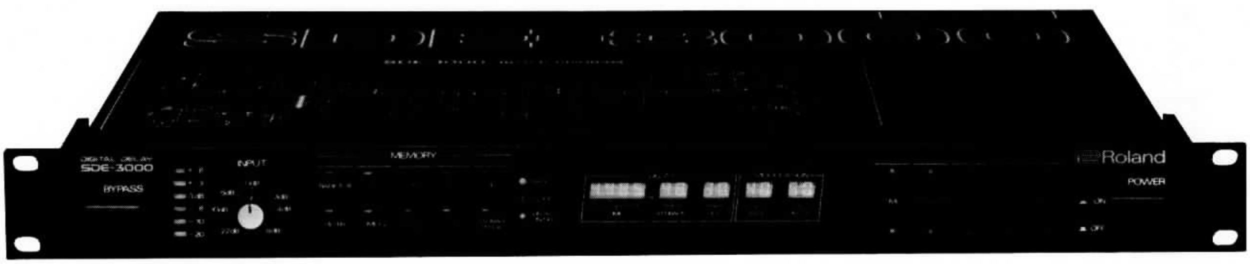


Fig. 4-17 shows how tape echo is generated. The delay is produced by the time required for the recorded sound to travel between the record head and the playback head. In most tape echo machines the tape speed is variable, or the position of the playback head can be changed so that the delay time can be set as desired. Many such machines also include more than one playback head so that many types of echo patterns are possible. An ordinary tape recorder with three heads can be used as an echo machine if it is connected as shown in Fig. 4-17. This is the method by which electronic echo was first generated in recording studios. The main disadvantage of using an ordinary tape recorder is that the amount of delay is usually limited by the number of playing speeds available. Also, unless a special tape loop is made for echo purposes, the tape will have to be rewound frequently.

Solid state echo machines work on exactly the same principles, the only difference being that the record-tape-play portion of the chain is replaced with a solid state delay line. The main advantage to such a system is that there are no tape heads or tape transport to wear out.

Fig. 4-17 Block Diagram of Tape Echo

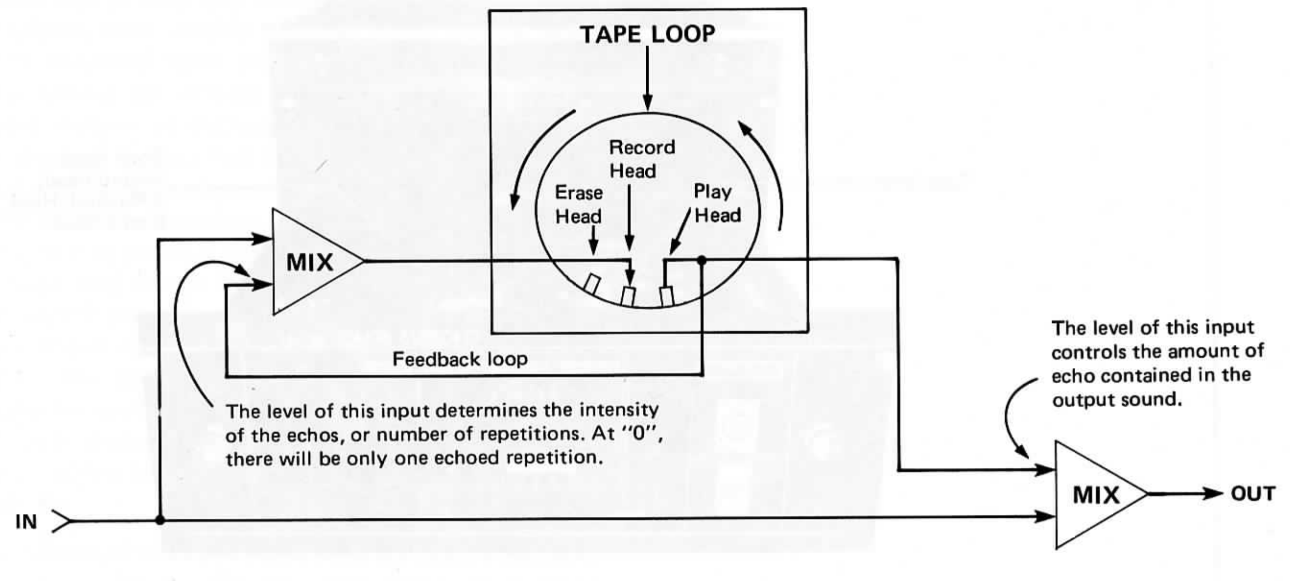
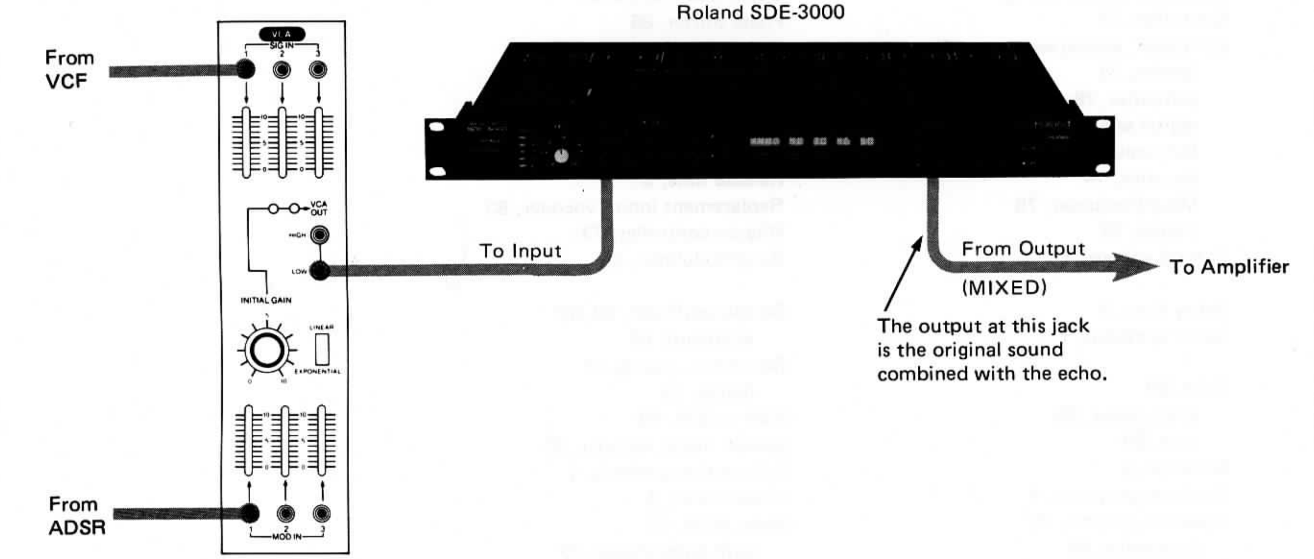


Fig. 4-18 shows how an echo machine can be patched between the synthesizer and the amplifier (or mixer) input. In experimenting with echo effects, try varying the echo repeat rate (tape speed in tape machines) while the machine is actually producing echoes. Varying the rate will cause the pitch of the returning echo to vary, which may be effective in certain cases. If it were possible to vary the echo rate relatively slowly and smoothly up and down at a rate comparable to vibrato, the result would be the chorus effect.

In discussing chorus and echo machines, we have crossed the hazy border between synthesis methods and effects.

Fig. 4-18 Using an Echo Machine



Index to Terms