$7.00

CAT SRM II

INSTRUCTION MANUAL; Series II synthesizers

Kitten II

OEI
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Contents;

CAT SRM II/ Kitten II Owners' manual
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The original Cat Synthesizer was designed in 1975 and became an immediate success because of it's reasonable price and exceptional versatility. As newer design techniques and more efficient integrated components became available to the electronic music industry, a successor instrument, the Cat S.R.M.(Series Revised Model) was introduced in 1977, along with a smaller version, the Kitten. These instruments were as successful as their predecessor since they featured improved internal electronics and additional features not found on the earlier model, while still maintaining a very competitive price/performance ratio. The S.R.M. featured a new two-note memory keyboard circuit which "cleaned up" the glitches normally associated with analog two-note keyboards. In addition to other improvements, both units featured an L.F.O. delay circuit and L.F.O. monitor to enhance their live-performance versatility.

Now, after producing thousands of instruments, we have introduced a second generation Cat SRM II and Kitten II in an effort to constantly maintain our products as reflections of the most recent advances in electronic music technology. These new designs are based on a "hybrid concept" whereby both analog and digital electronics have been applied to provide the most cost effective approach to state-of-the-art synthesizer design.

Both instruments utilize a digital keyboard system incorporating a unique single-contact-per-key design, dramatically reducing keyboard related malfunctions due to dirty or misaligned contacts. This digital system also allows the keyboard to be processed by a properly interfaced digital computer.

The second generation instruments are based on highly advanced single-chip integrated oscillators and filter to provide the utmost in reliability, performance capability and ease of repair.

Interestingly enough, the original CAT panel design has proven to be so functionally efficient, that it was difficult to improve upon. As such, only minor cosmetic changes have been made to the revised units. In the second generation series, the patch bay has been expanded so that the interfacing capabilities are greatly enhanced.

It is our hope that you derive as much enjoyment and satisfaction from the use of our instruments as we have in designing and manufacturing them.

Carmine J. Bonanno,
President,
Octave-plateau Electronics, Inc.

INTRODUCTION;

The average person knows the sound of a piano or guitar when he hears it, but rarely is the presence of a synthesizer as obvious as these well-known instruments. In fact, few people realize just how many sounds they hear on TV themes, commercials, movie sound tracks and recorded music are actually made by synthesizers. From the sound of pouring coffee, to the background sounds of headaches on aspirin commercials, to the melodic and dramatic sounds of popular rock music, synthesizers have made their mark and are here to stay!

Synthesizers look frightening at first glance. In fact, they're frightening even after two or three glances. Their myriad of control knobs, switches and symbols evoke visions of "Future Shock" rather than of a practical musical instrument. A piano is easy to cope with. You sit down, learn which keys are which and start playing. You always know what to expect and what it will sound like night after night.

A synthesizer, on the other hand, is touchy. The twist of a control can turn it from a bass into a twinkle. And before you can get any predictable sound out of it you have to memorize an owners manual or copy a bunch of patch sheets to show you how to set up the sound. And if one control is set wrong- you'll go nuts!

So why would anyone bother with a synthesizer? Why not just stick with the old-faithful conventional instruments?

Well think about this; the difference between a synthesizer and a conventional instrument, such as a piano, is like the difference between a car and a bicycle. The bicycle is great for short trips, but when you want to travel far beyond your immediate area, forget it. Pianos are great for their intended purpose, but synthesizers open up an entirely new and boundless outlook on your musical creativity. With a synthesizer, you create the sound as well as the music. Of course a piano is much easier to use than a synthesizer, just as a bicycle is easier to use than learning how to drive a car. But think about how far you can go after learning to drive a car. The same goes for synthesizers, a little work goes a long way.

HOW DOES IT WORK?

If you're a musician, don't be frustrated if you're not familiar with how synthesizers work- it's really not as hard as it looks. Remember, it's probably easier to teach a musician how to use a synthesizer than it is to teach an engineer who knows nothing about music how to play an instrument.

So let's begin by making a statement that will probably sound dubious: you were born with a synthesizer in your body. Ridiculous, right? OK, let's explain. When you speak or sing you synthesize words. Your lungs push air through your vocal cords which send vibrating air through your mouth. Your mouth changes its shape and forms different sounds which we call speech. The mechanism consisting of your lungs, vocal cords, mouth and facial muscles can be considered a biological (rather than electronic) synthesizer because it synthesizes words from air in your lungs. An electronic synthesizer creates sounds from the electricity in your wall socket.

The biological synthesizer has control over certain parts of the sounds it synthesizes. If you scream, your lungs push out more air than if you whisper and as such they control the LOUDNESS of the final sound. If you tighten your vocal cords, your speech would sound higher than if you loosened them to sound deeper. Thus, the vocal cords control the PITCH of the final sound. Finally, if you say "eeeee", or "mmmmmm" or "wow", your mouth would take three different shapes resulting in three different tones. So the mouth controls the TONE or TIMBRE of the final sound.

What controls your mouth, vocal cords and lungs? Your muscles, of course! So, your muscles are CONTROLLERS that let your biological synthesizer vary the three parameters of the final sound: pitch, timbre and loudness.

Now we want to do this electrically. Obviously, we need electronic devices that let us change pitch, timbre and loudness and a way of controlling these devices. Enter the synthesizer.

In a synthesizer, the pitch is determined by an electronic device called an OSCILLATOR, the timbre by a FILTER and the loudness by an AMPLIFIER. Now, just as the vocal cords, mouth and lungs need muscles to change them, the corresponding devices in a synthesizer need something to change them. So they are designed in such a way that their functions follow something called a voltage. Don't worry about what a voltage is, it doesn't matter. You could substitute the word "water" and it wouldn't change the concept. In fact, just so you won't feel threatened by it, we'll use both terms for a while to make the transition.

The synthesizer oscillator determines the pitch of the sound produced. The specific pitch produced by the oscillator depends upon how much water (voltage) you pour into it. Give it a lot and the pitch goes higher, a little and the pitch goes lower.

The filter determines the timbre of the final sound. The actual tone produced depends upon how much water (voltage) you pour into the filter. Pour in a lot and the tone gets sharper, a little and the tone gets mellower.

The amplifier (the one on the synthesizer, not the one with the speakers in it!)determines the loudness of the sound. It makes the sound louder if you pour a lot of water (voltage) into it and softer if you pour a little into it- and it turns the sound off altogether if you don't put any water (voltage) into it at all.

Now, if the oscillator were really controlled by water, we could call it a Water-Controlled-Oscillator, or W.C.O.. But in real-life, it's controlled by this thing called a Voltage, so instead we call it a Voltage-Controlled-Oscillator, or V.C.O.. The same goes for the filter (V.C.F.) and amplifier (V.C.A.).

So the synthesizer has an assembly line made up of a V.C.O, V.C.F., and V.C.A., each responsible for the pitch, timbre and loudness, respectively, and each following external voltages that tell them exactly how to respond. These voltages come from different CONTROLLERS on the synthesizer.

The most common controller is the keyboard. Each key on the keyboard corresponds to a different voltage. For instance, the low C key could be 1Volt,the next octave C key could be 2Volts, and the next octave C key could correspond to 3Volts.If the keyboard voltage is connected to a V.C.O., the pitch produced by the V.C.O. would correspond to the voltage coming from the keyboard, which depends upon which key is down. So if 1Volt comes out and goes to the V.C.O. the pitch corresponding to that key would come out. If the next octave C key corresponding to 2Volts is pressed, the pitch produced would be one octave higher. That's why a synthesizer can only play as many notes as it has oscillators-each oscillator can only play one pitch at a time. So a Cat SRM can play a maximum of two notes at a time because it has two V.C.O's, while a Kitten with only one V.C.O. can play only one note at a time.

Some other controllers available on a synthesizer are ADSR, Sample and Hold, Low Frequency Oscillator, and others, but let's not get too involved right now. The point to remember is that the actual sound producing devices are the Oscillators, Filter and Amplifier while the devices that manipulate these three are the controllers. The controllers put out voltages that tell the sound manipulating devices how to change. When a V.C.O. (or V.C.F or V.C.A.) is being manipulated by a controller it is said to be MODULATED by the controller. The process of connecting controllers, V.C.O.'s, V.C.F.'s and V.C.A.'s together to form a sound is called a PATCH.

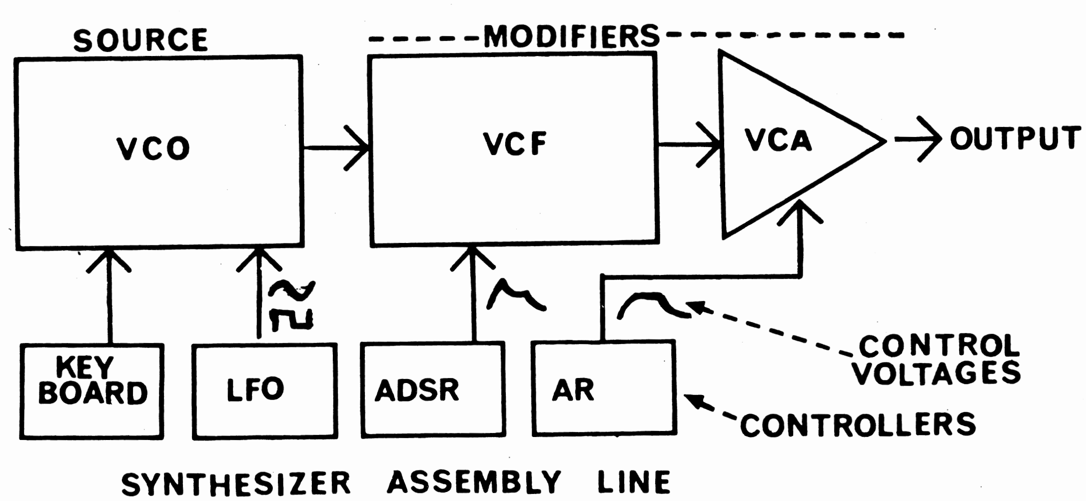


Fig. 1: The major components of a typical synthesizer are categorized as sources, modifiers and controllers.

WHAT YOU CAN AND CAN'T EXPECT FROM YOUR SYNTHESIZER.

The synthesizer is a tool for manipulating the three basic parameters of sound- pitch, timbre and loudness. It conveniently places these parameters on a panel in the form of electronic devices called oscillators, filters, amplifiers, LFO's, etc. which are controlled by turning knobs, flipping switches, pushing sliders, playing keyboards, pressing buttons and moving pedals.

The fact that a synthesizer allows sounds to be manipulated does not imply that ANY sound can be duplicated- some sounds can't. This is because some sounds are very complex in that they are made up of multiple variations in pitch, timbre and loudness all occurring simultanously in short periods of time. Reproducing such sounds requires complex electronic devices far beyond the capability of a performance synthesizer. For example, a human voice is difficult, if not impossible, to accurately simulate with a performance synthesizer. Other sounds, however, are simple enough to imitate so exactly that it often becomes difficult to distinguish the real from the synthetic. For example, a bass guitar, flute, tuba and other instruments can be effectively simulated with the manipulation techniques available on today's performance synthesizers.

Although the synthesizer possesses a phenominal ability to imitate conventional sounds, it's greatest potential lies in it's ability to create unconventional sounds that cannot be obtained naturally. For example, it can be set to imitate a tuba and then be changed to better suit a particular application. The point is that the sound is under YOUR control and can be tapered to suit your needs- this being a quality that no other instrument can offer.

The creative possibilities of the synthesizer are almost infinite- limited only by the complexity of the device and your knowledge of it's operation. Taking the time to fully understand the purpose of every control on the front panel will allow the greatest degree of versatility to be obtained from the instrument, as well as prevent much frustration during the creative process.

BASICS; getting technical

The first thing that should be pointed out is that sound produced by a synthesizer is manipulated within the instrument as a group of "voltages" or electrical signals. When the output of the synthesizer is connected to an amplifier-speaker combination, the speaker turns the synthesizer voltages into vibrating air. This air vibration travels through the room, and when it reaches our ears, our brain perceives it as sound. This perceived sound is an audible replica of the signals within the synthesizer.

Since we must differentiate between the sound we hear and its equivalent electrical counterpart in the synthesizer, there are names given to parts of sound as they exist in their electrical form and in their physical form. For instance "pitch" is something the brain perceives, while the corresponding electrical signals in the synthesizer are said to have a particualr "frequency". Thus, a "high pitched" sound corresponds to a "high frequency" signal in the synthesizer. Frequency is measured in "cyles per second" or "hertz" (Hz) so that "A above middle C", for instance, has a frequency of 440Hz (hence the term A-440).

Another example of synthesizer terminology is the term "loudness". Inside the Synthesizer, the signals are said to have a particular "amplitude" which corresponds to the resulting sound's loudness.

When we refer to the "tone" of the final sound, inside the synthesizer the signals are said to have a particular "harmonic content" which corresponds to the "timbre" of the sound. A signal containing few low-order harmonics will have a characteristically mellower sound than one which has many higher order harmonics. Our brain distinguishes between the various types of harmonic structures of sound and tells us this information in the form of timbre or tone.

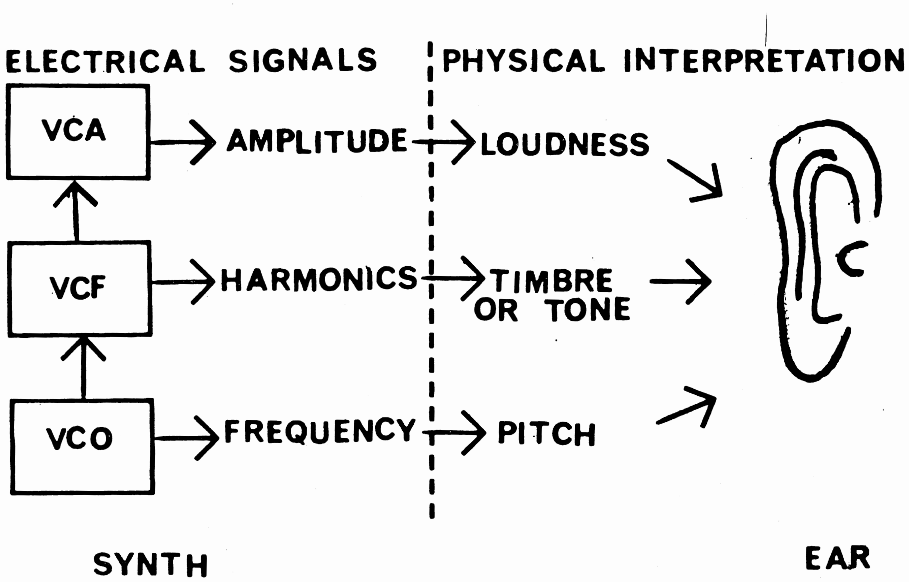


Fig. 2: Relationship between electrical terminology in synthesizers and physical terminology in music.

OSCILLATOR (VCO)

The signals inside the synthesizer are produced by the oscillator in the form of "waveforms" at different frequencies. A waveform refers to the shape of the electrical signal produced by the oscillator- each different type of waveform will have its' own particular harmonic structure and so its' timbre will sound different.

The waveforms available on the Cat SRM II and KITTEN II are called square, triangle, sawtooth, and pulse. Each waveform has a different tone and so by mixing them together in different pŕoportions you can combine tones.

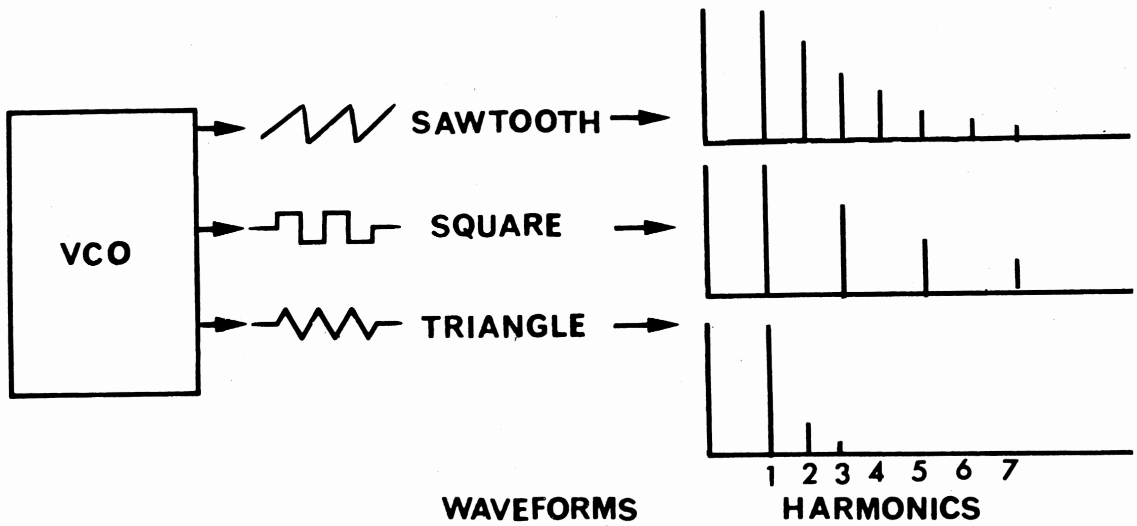


Fig. 3: The initial determinant of synthesizer timbre is the waveform selected from the VCO.

FILTER (VCF)

The basic harmonic content of a synthesizer signal is determined by the waveshape you select from the oscillator. The filter allows us to take away and/or emphasize certain harmonics from this signal to furthur modify the timbre of the final sound. The Cat SRM II and KITTEN II each have a "low pass filter" which means that it only allows harmonics below a certain frequency to pass through it. This frequency, above which no harmonics may pass, is called the "cutoff frequency" or Fc. Varying the cutoff frequency will modify the timbre of the waveform coming out of the oscillator.

Another interesting effect is called "resonance" or Q. This refers to an emphasizing of the harmonics that lie near the cutoff frequency. The result of varying the cutoff frequency with the resonance turned up sounds something like the typical "waa-waa" effect often heard in recordings. If the filter resonance is turned up to maximum position, the filter will turn into an osillator and not let any V.C.O. waveforms through.

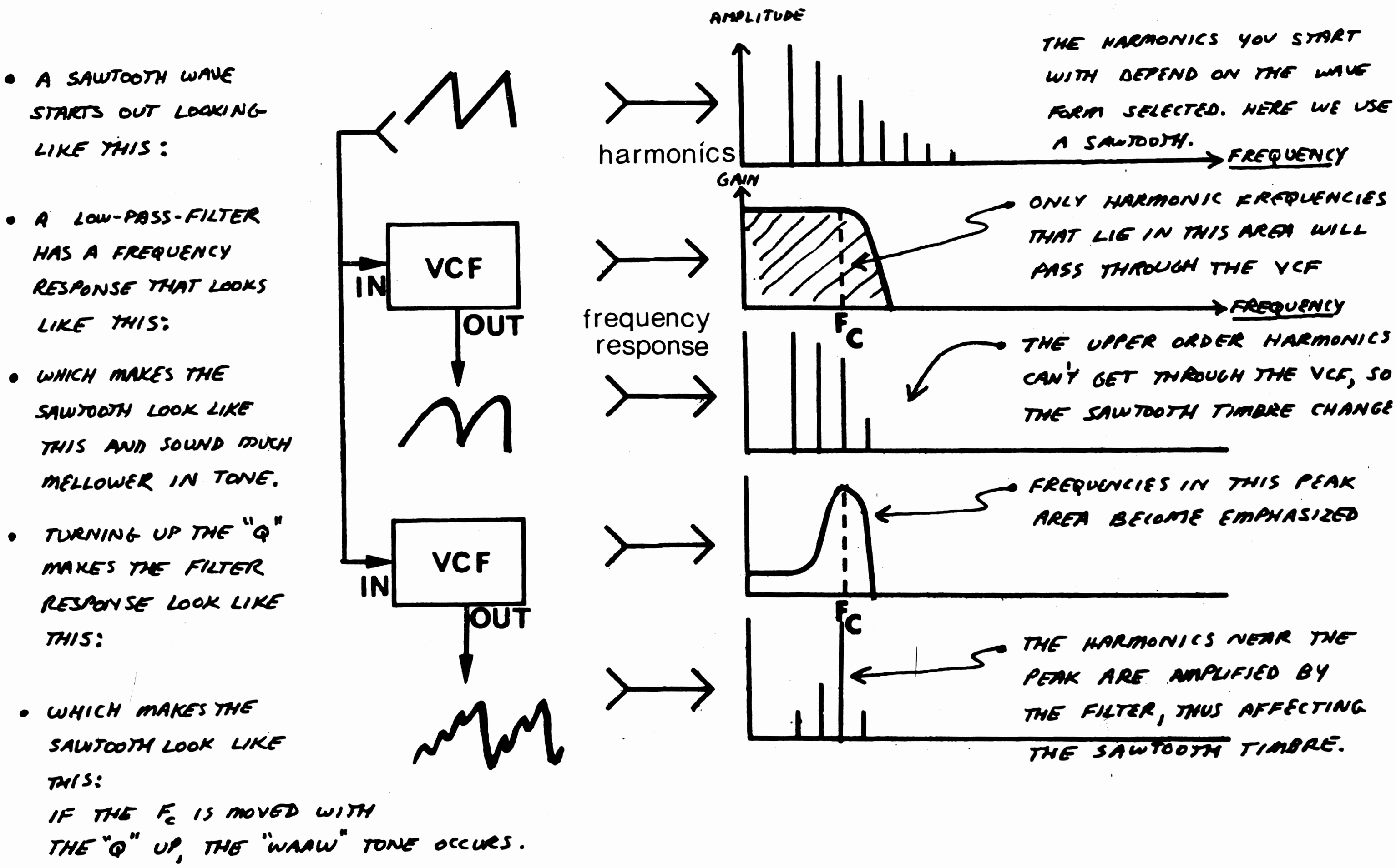


Fig. 4: The VCF modifies the harmonic spectrum of the input waveform by taking away and/or emphasizing harmonics to alter the final timbre.

AMPLIFIER (VCA)

The internal amplitude of the synthesizer signal is determined by an electronic device called an amplifier. Although it has the same name as the common, everyday amplifiers we're used to seeing, there's a big difference- the least of which is that it can't drive speakers. The purpose of the synthesizer amplifier is to vary the signal amplitude according to a control voltage. If no control voltage exists, no signal will pass.

If we put a signal of constant amplitude into the V.C.A. and then change the input control voltage, the amplitude of the signal coming out of the V.C.A. will follow the shape of the voltage variation. The contour of the signal amplitude is called the "envelope".

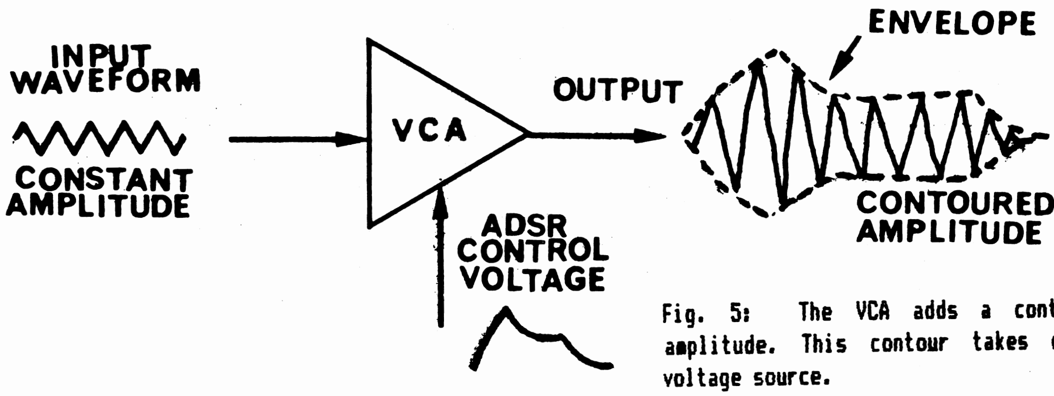


Fig. 5: The VCA adds a contour to an otherwise constant amplitude. This contour takes on the shape of the control voltage source.

Voltage Control

Suppose you want to synthesize the sound of a trumpet. Assume you start with a pitch of middle A, or 440Hz. The harmonic content starts off with a burst of bright timbre and then mellows out for the remainder of the note. The envelope varies in a similar manner, first having a loud burst and then reaching a plateau as the player blows softer into the trumpet. Now, consider the controls on the synthesizer and think about how many hands you would need to vary the parameters of the sound involved in the synthesized trumpet sound. You would need one hand to contol the filter cutoff, Fc, another to control the amplitude variation, and if you want some vibrato you'd also need a hand to vary the oscillator frequency. Oh yes, you'd also need a hand to play the keyboard. That's a total of four hands. This can be a problem for the average human being- so what do you do? How can you simultaneously (and quickly) control several synthesizer parameters without a multitude of hands? We can let some electronic devices do the work- this is where the principle of "voltage control" steps in.

The synthesizer is designed so that the parameters of frequency (pitch), filter cutoff (timbre), and amplitude (loudness) can all be varied by voltages. These control voltages change in the same way you would vary the parameter being controlled so that you free your hands and have it done automatically. So the control of synthesizer parameters reduces to a matter of finding the right devices that give you the variations in voltage you need for a particular application. Doing it this way frees you from a lot of knob twiddling and converts the problem of synthesizing a trumpet to that of setting up the proper connections between the control voltage sources and the VCO, VCF and VCA.

Controllers

The parts of a synthesizer that are the source of these "control voltages" are logically called "controllers". The controllers on a Cat SRM II are the transient generators (ADSR and AR), the sample and hold (S+H), the low frequency oscillator (L.F.O) and the keyboard. The KITTEN II has all of these except for the AR. All of these controllers put out different voltages to vary synthesizer parameters.

Sources

Since we're classifying different parts of the synthesizer, we should note at this point that since the oscillator is a "source" of sound, it is referred to as a "source". The noise generator is another "source" since it is a generator of sound, which in this case is non-pitched.

Modifiers

Also, since the V.C.F. modifies the harmonic structure and the V.C.A. modifies the amplitude of the signals, they are both classified as "modifiers".

Patching

The way in which the sources, controllers and modifiers are connected together is called "patching". The panel has switches and knobs that let you connect the different sections together in different proportions to form a "patch" for a particular sound.

Modulation

When a controller is patched to a source or modifier, it is said to "modulate" or vary the particular parameter it is affecting. for instance, if the LFO sine wave is connected to the VCO control voltage input to get vibrato on the VCO pitch, this is termed as "the VCO being modulated by the LFO" so that the LFO is the "modulation source" while the VCO is the "destination" The amount of pitch variation caused by the LFO is called the"modulation depth" and is set by a control in the VCO section.

PATCHING SYSTEM;

The functional layout of the Cat SRM II and the KITTEN II is based on the combination of slide pots, rotary pots, and slide switches situated in such a way that many functions are visually grouped. A system of slide switches select the source of modulation (the controller), while the rotary attenuators beneath each switch adjusts the depth of modulation. The functional grouping on the front panel can be generalized as follows:

1)Each slider within a V.C.O. section controls an audio signal level. The level for the noise source is also a slider.

2)All of the modulation attenuators are rotary pots located in a common row to distinguish their function.

3)Wherever possible, symbols rather than words are used to indicate synthesizer functions in a simple visual manner.

4)The grouping of the V.C.O., V.C.F. and V.C.A. modules is from left to right to visually correspond with the signal flow within the system.

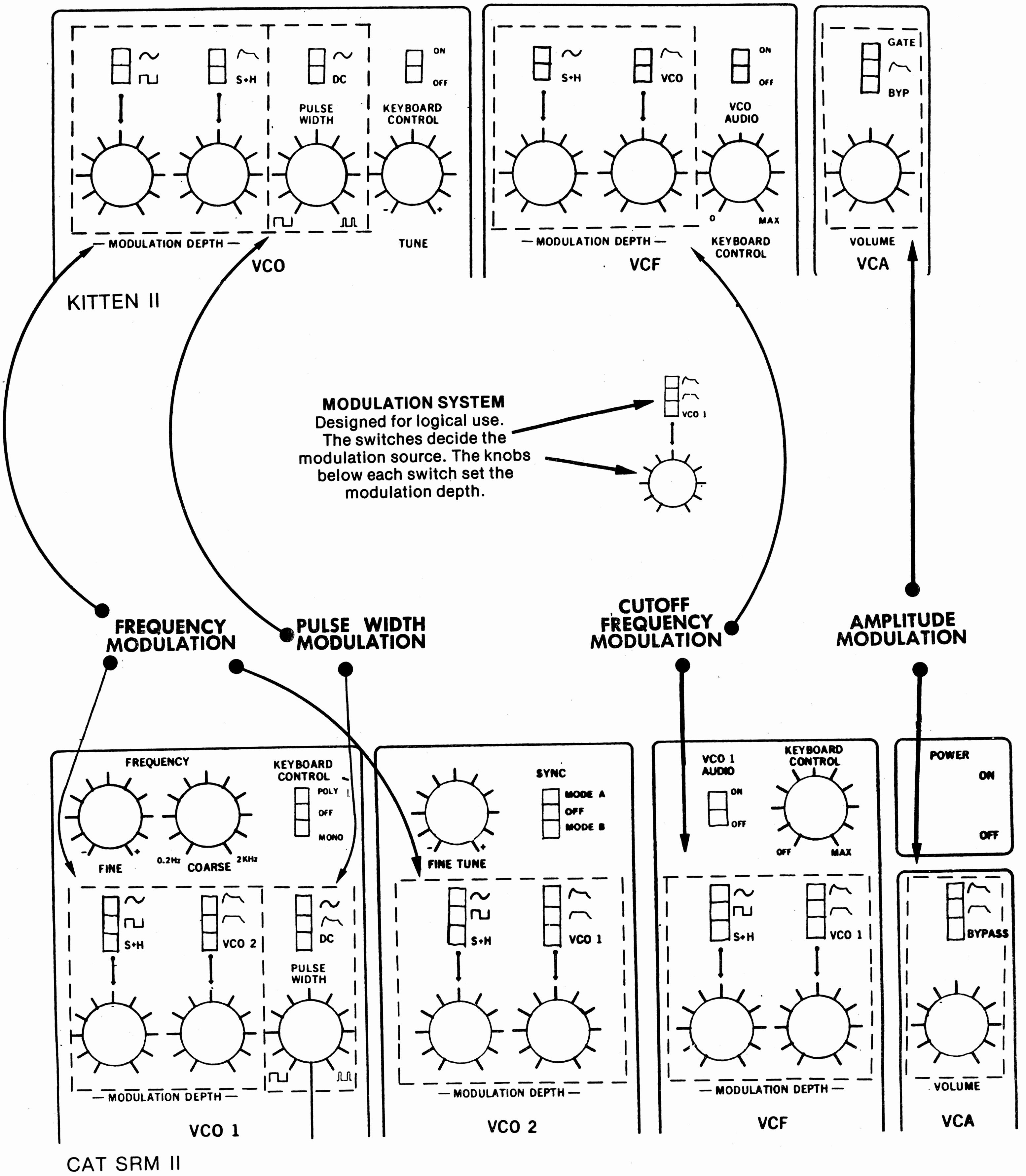
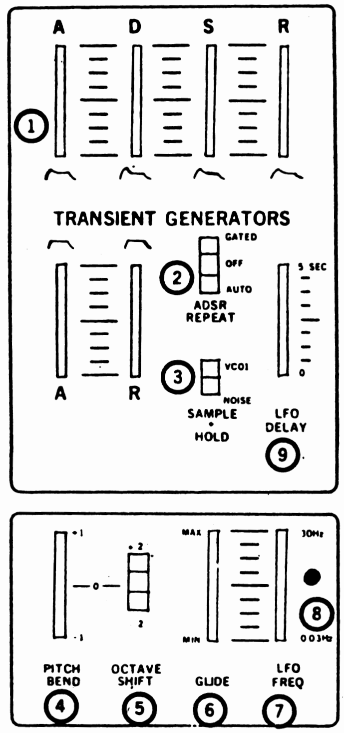


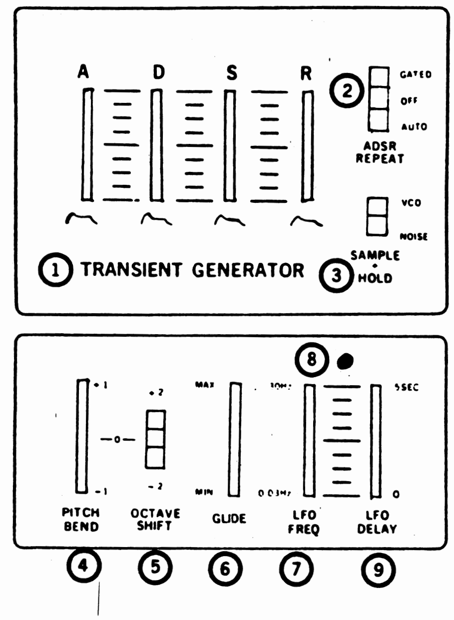
Fig. 6: The patching system used on the Cat SRM II and Kittren II.

CAT SRM II



AR TRANSIENT GENERATOR
The attack of the AR Transient is controlled by the A slider and the release by the R slider.

KITTEN II



1 **ADSR TRANSIENT GENERATOR**
Each slider controls different portions of ADSR as labeled below each slider. The "A" is for attack, "D" for decay, "S" for sustain, "R" for release.

2 **ADSR REPEAT**
OFF is the normal mode where the ADSR is triggered by the keyboard. AUTO makes the ADSR repeat at the LFO speed. GATED makes the ADSR repeat only when a key is down.

3 **AUTOMATIC SAMPLE AND HOLD**
samples either the noise or the mixed VCO1 waveforms for complex modulation effects. The output of the SAMPLE and HOLD is available at all of the patch switch positions labeled S & H.

4 **PITCH BEND**
has a center locking dead zone and allows transposing the unit by +1 or -1 octave.

5 **OCTAVE SHIFT**
transposes the unit by + 2 or -2 octaves.

6 **AUTOMATIC GLIDE**
allows the glide to complete even if the key is lifted.

7 **LFO FREQUENCY**
controls the speed of the LFO sine ∿ and square ⊓⊔ waves that are available at all of the patch switch locations labelled ∿ and ⊓⊔

8 **LFO MONITOR**
gives a visual indication of the LFO speed.

9 **LFO DELAY**
lets you delay the effect of the LFO sine wave (∿) for delayed modulation effects

PANEL FUNCTIONS REFERENCE

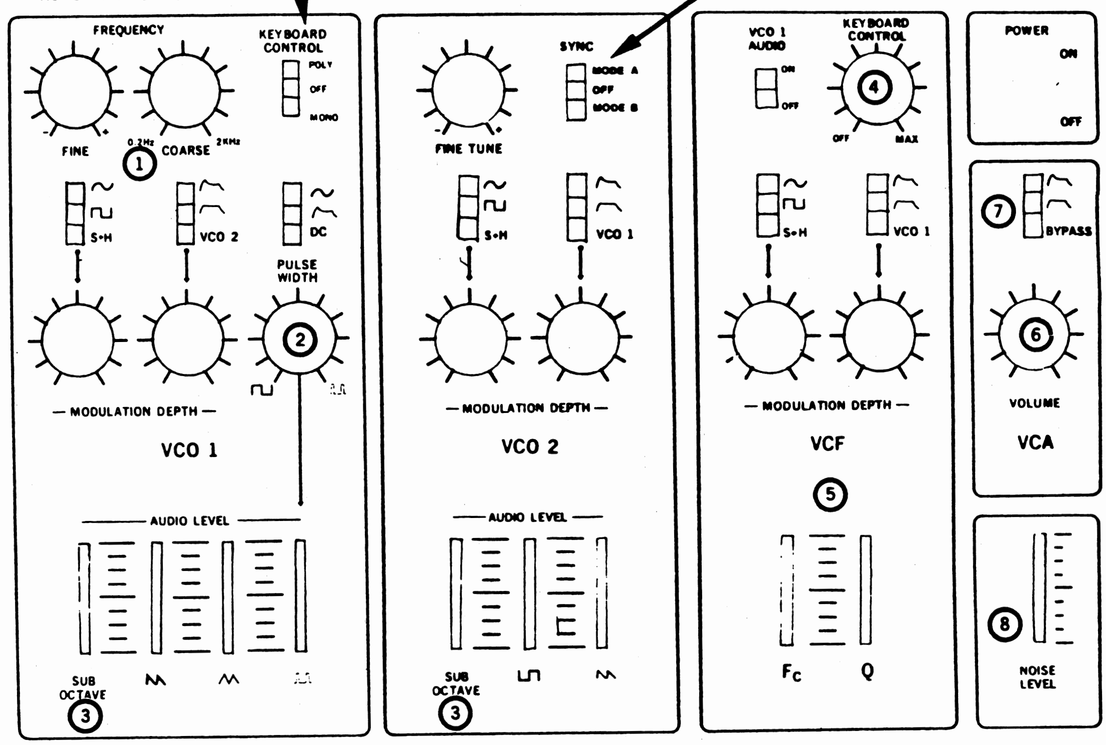
Fig. 7: Panel descriptions. Refer to the text for furthur details.

POLY/MONO
MONO makes VCO1 play the lowest note depressed on the keyboard.
POLY makes VCO1 play the highest note depressed on the keyboard.
Use this setting for 2 note memory effects.
OFF makes the keyboard have no effect on VCO1

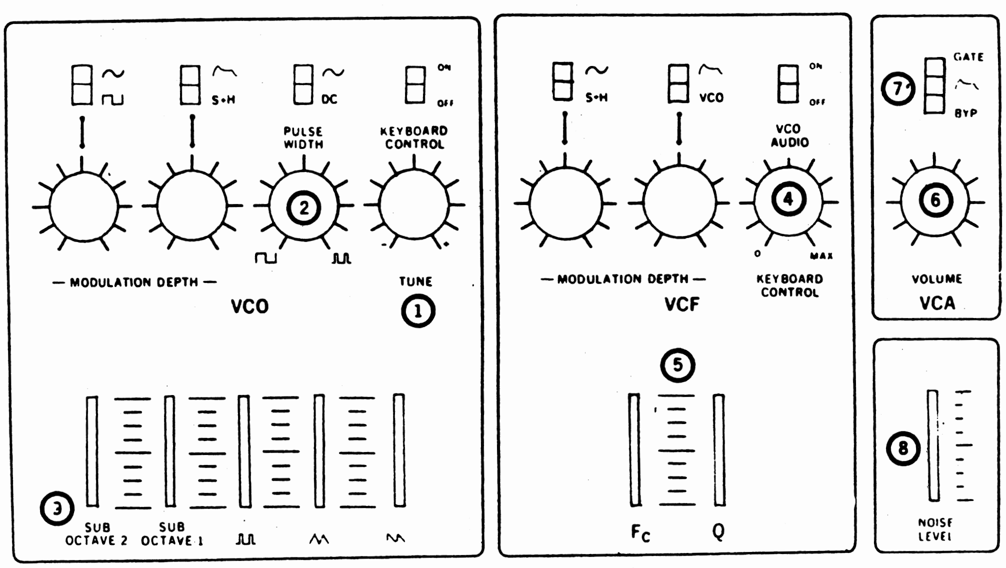
VCO2 FINE TUNE
is used to tune VCO2.
VCO2 always plays the lowest note depressed on the keyboard.

SYNC
Locks VCO1 onto VCO2 to from complex timbres from VCO1

SRM II



KITTEN II



1 **VCO TUNE CONTROLS:**
Use to tune VCO

2 **PULSE WIDTH MODULATION**
allows varying the pulse width for different timbres.
Use the switch above this knob for dynamic pulse width modulation (∼) or manual pulse width setting (DC).

3 **SUB OCTAVES**
on both VCO'S make 2 oscillators sound like 4.
MIXABLE WAVEFORMS gives each oscillator waveform its own volume slider for complex timbres.

4 **VCF KEYBOARD CONTROL**
allows continuous variation of the filter keyboard response.

5 **— 24dB/OCTAVE VCF**
The Fc adjusts the filter cutoff and the Q adjusts the resonance.
When the Q is up all the way, the VCF turns into a sinewave oscillator and won't let the oscillator sounds through.

6 **VOLUME CONTROL**
adjusts the volume of the entire unit.

7 **VCA PATCH SWITCH**
In the ADSR position, the volume of the unit follows the ADSR transient.
In the AR position, the volume follows the AR transient.
In the BYPASS mode, the volume is constant.
This is used for setting up a patch without holding a key down to hear the sound.

8 **WHITE NOISE GENERATOR**
Turning up this slider brings in white noise to the VCF.

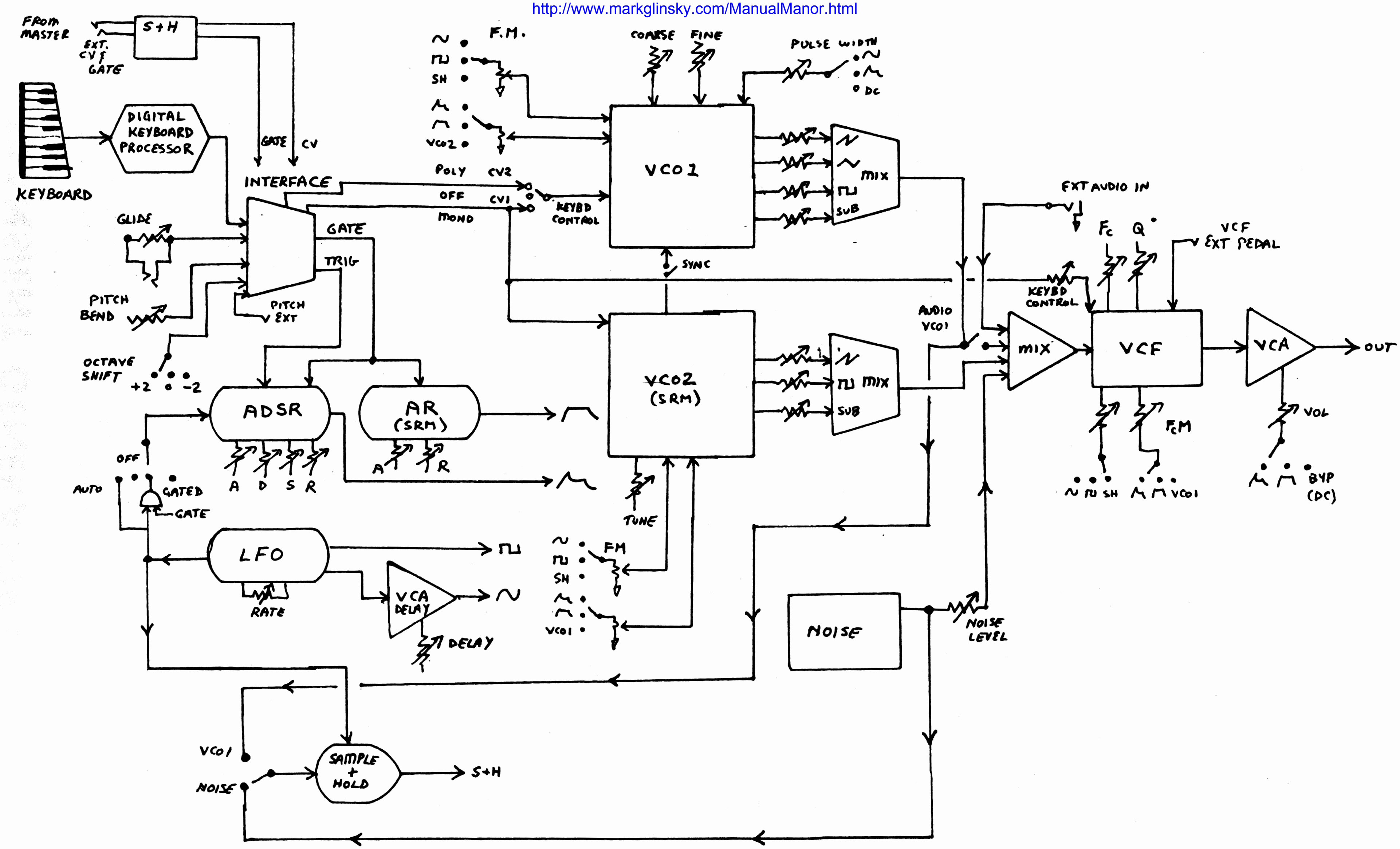


Fig. 8: **SYSTEM FLOW DIAGRAM (srm II / kitten II)**

TRANSIENT GENERATORS

The transient generators on the Cat SRM II are the ADSR and AR controllers. The KITTEN II has a GATE signal in place of the AR.

ADSR

The ADSR transient generator generates a four-part voltage shape whenever a key is depressed. The four parts are called Attack, Decay, Sustain and Release (thus the abbreviation ADSR) each controlled by a front panel slider. The attack can be short (fast) or long (slow), meaning that the output voltage starts at zero, and when a key is pressed, increases at a rate that depends upon the "A" slider setting. After reaching a peak value, this voltage decays down to a plateau voltage called the "sustain" level. The decay time and sustain voltage are each controlled by the sliders marked "D" and "S" respectively. The output voltage remains at the sustain level until the key is released, after which it goes back down to its' original zero value at a rate determined by the setting of the front panel "R" (release) slider.

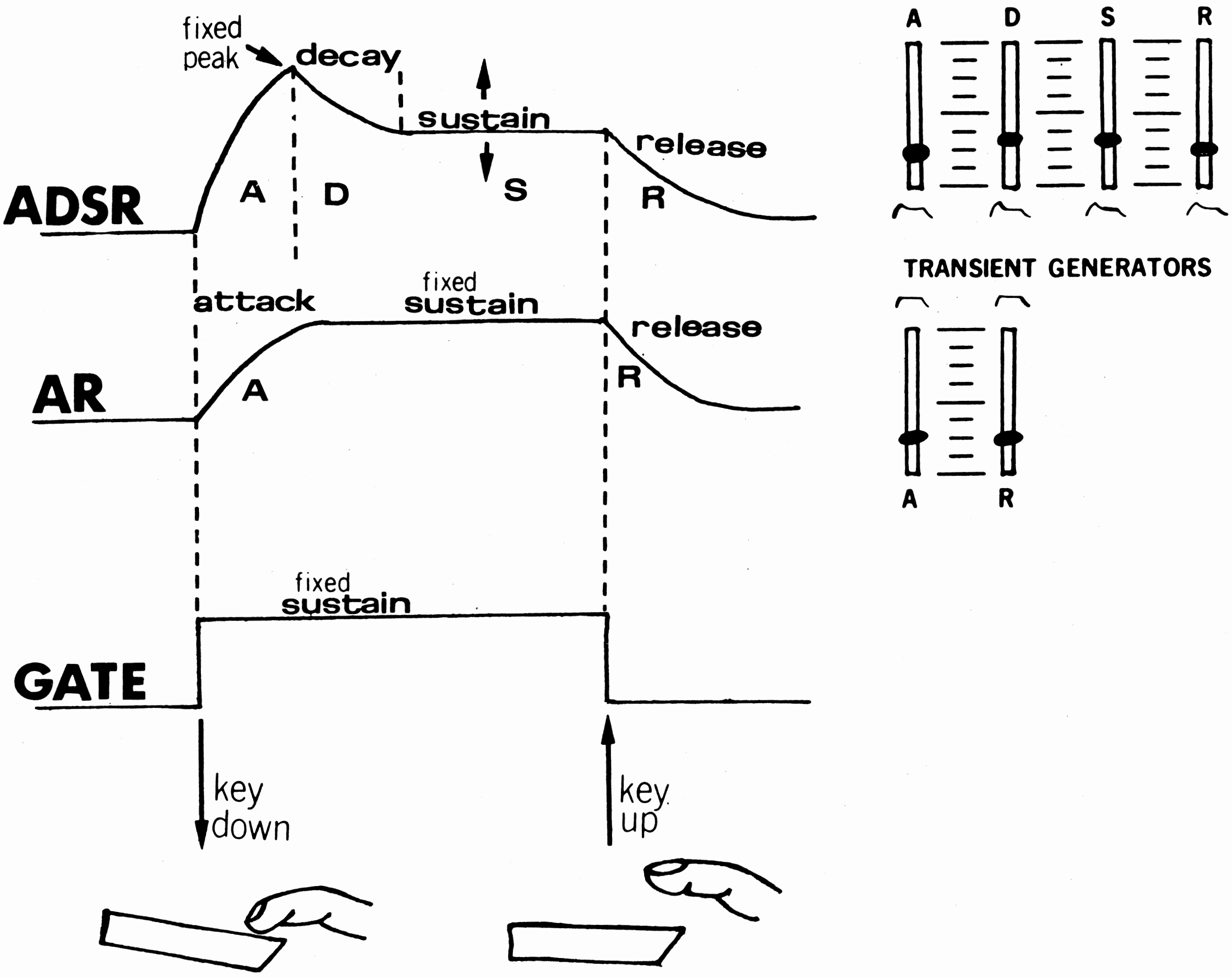


Fig. 9: Relationship between ADSR, AR and GATE control voltages.

We can use this ADSR transient voltage in a number of ways. The most common use is to control loudness by connecting it to the V.C.A. control voltage input. For instance, if a piano-type envelope is desired, the attack would be set to the minimum point (fast attack), the decay and release would be medium (slider up about 1/4 of the way) and the sustain would be set to minimum because a piano does not continue to play the note if the key is held down. The voltage transient resulting from this setting, when patched to control the V.C.A., would produce a percussive piano-like envelope.

The same ADSR control voltage used to control the loudness when connected to the VCA can be used to control pitch and timbre by connecting it to the VCO and VCF.

These three examples illustrate the imprtant point that a controller, such as the ADSR transient generator is nothing more than a voltage source whose effect on the final sound depends upon where it is connected. If it is connected to the V.C.A., the envelope of the final sound will follow the contour of the controller output voltage; if it is connected to the V.C.O., the pitch of the final sound will follow the contour of the controller output voltage; and if it is connected to the V.C.F., the timbre of the final sound will follow the contour of the controller output voltage.

The output of the ADSR transient generator is available at all switch positions labelled (⌃).

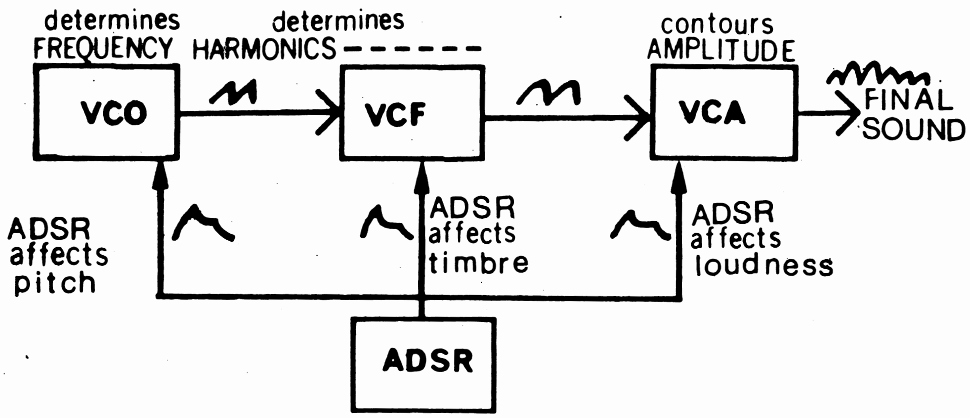
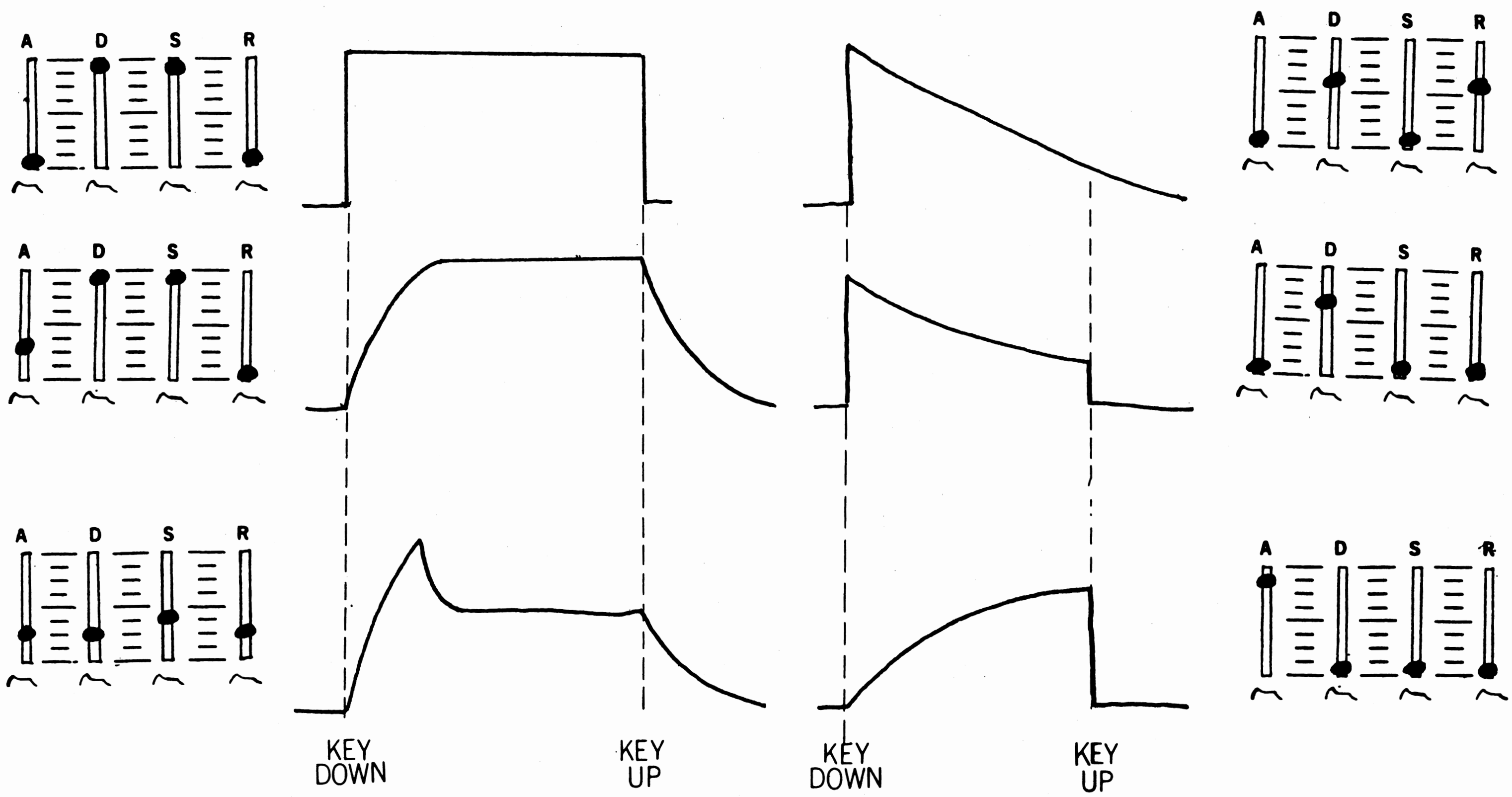


Fig. 10: A controllers' effect on the final sound depends upon where it is connected in the synthesizer assembly line.

Fig. 11: **Typical ADSR Settings:**



Touch Sensitive ADSR Response;

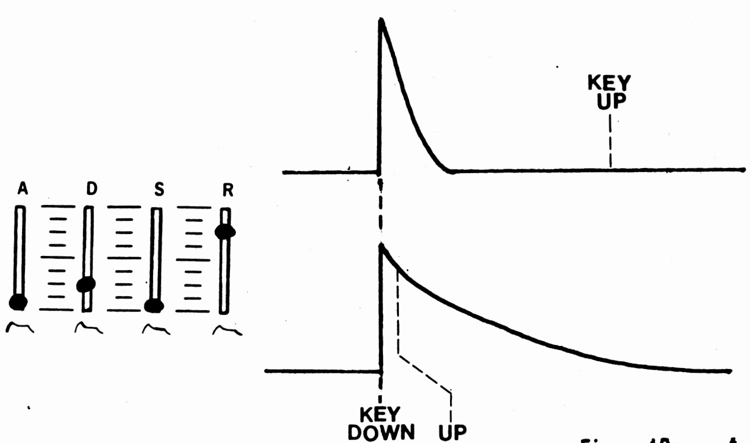


Fig. 12: A touch sensitive ADSR response can be obtained by utilizing the fact that the release (R) portion of the ADSR takes over only when a key is released. If you set a fast decay and hold down a note, the transient will decay at a rate determined by the position of the decay (D) slider. However if the key is quickly hit so that it is only down for a short time, the setting of the R slider will take over upon release of the key. Since this release will take over at the voltage point existing at the time the key is released, the release time of the transient will depend upon how long the key was pressed, or alternately, how quick it was hit. Thus, by using this setting, different transients will occur for different key hits. this is very useful for expressive percussive patches.

EXPERIMENTING WITH THE ADSR

To see how the transient generator can be used to control pitch, timbre and loudness, set up the patch shown in the following figure.

First turn up the knob labelled "A" and then hit any key. The pitch of the sound follows the ADSR shape. The depth of this pitch change is controlled by the knob, while the shape of the pitch change is controlled by the slider settings on the ADSR. What is happening here is that the V.C.O. is following the voltage shape produced by the ADSR controller each time a key is hit.

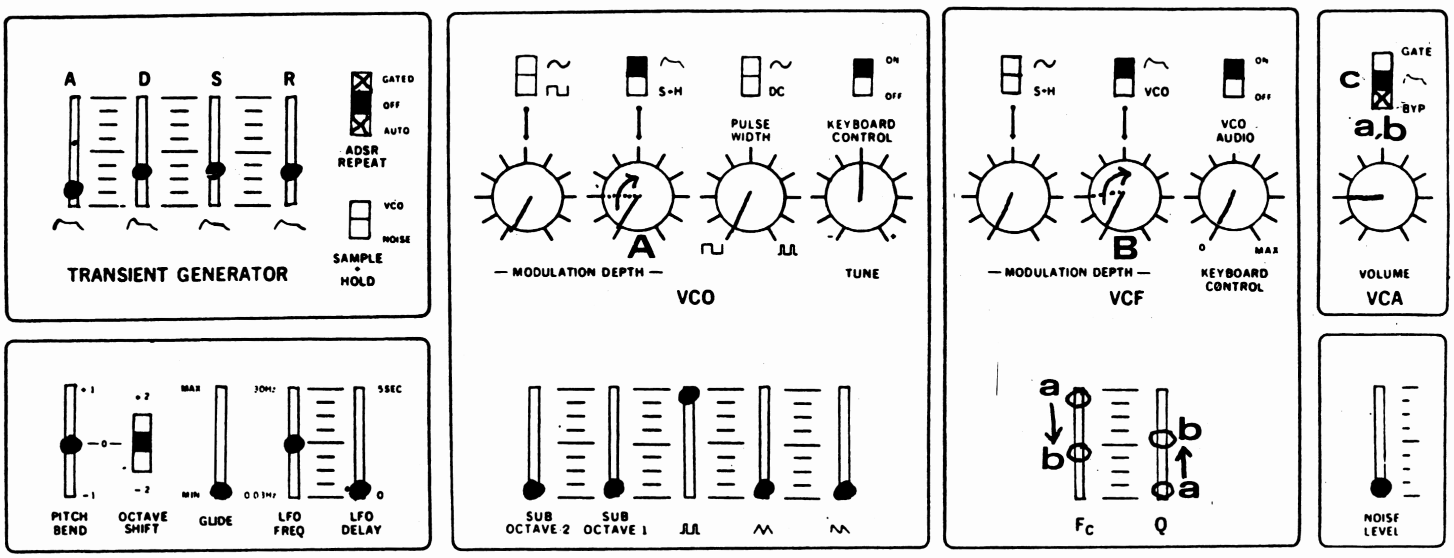
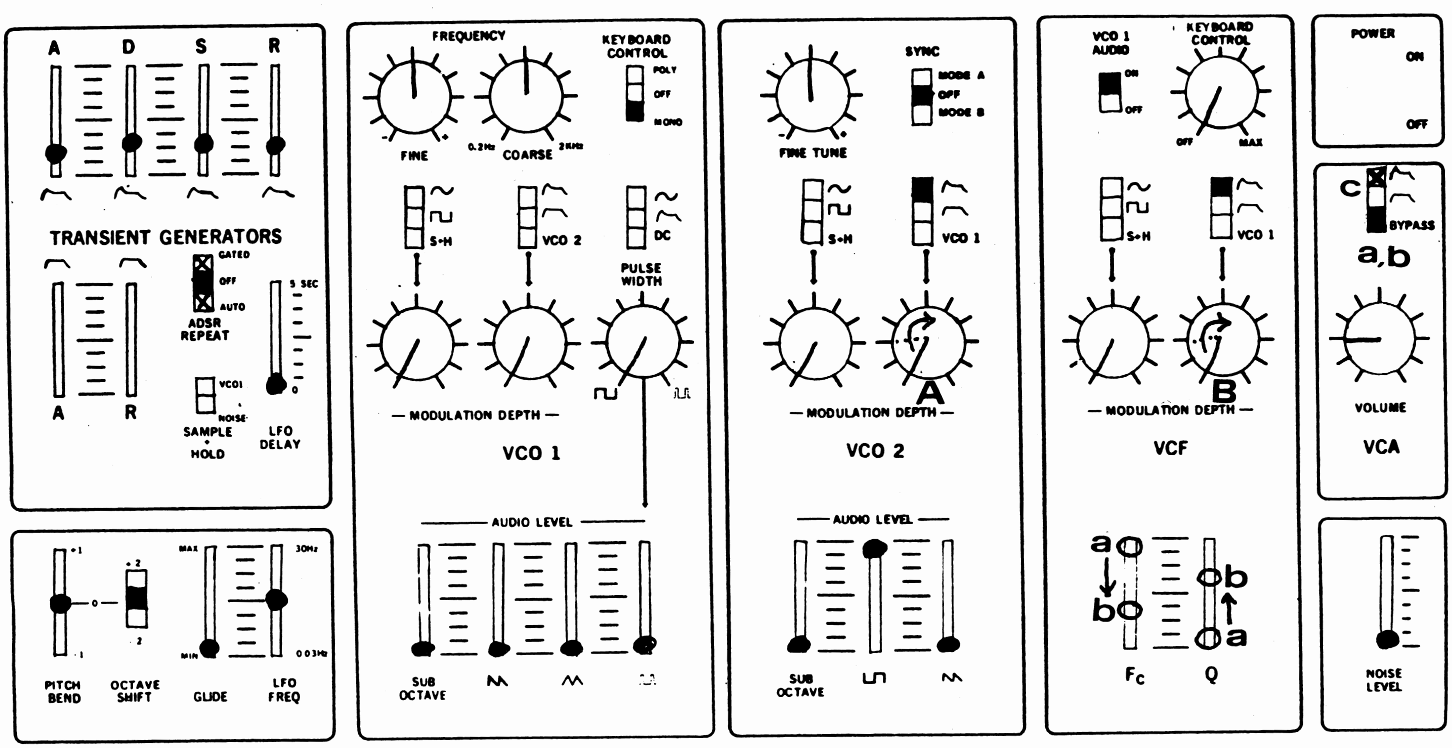


Fig. 13: Panel settings for the ADSR experiment.



Try putting the ADSR repeat switch in the "AUTO" mode and then in the "GATED" mode. The speed of this effect is controlled by the LFO speed setting. What happens here is that the triggering effect of the keyboard is replaced with the automatic triggering of the LFO so that continuous ADSR control voltage transients are generated.

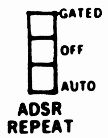
Now turn down the "A" knob and turn up the "B" knob. set the "Fc" and "Q" controls as shown in the "B" position change.

Notice how the same ADSR transient is now affecting the V.C.F. in the same way as it did the V.C.O. except that now the timbre of the sound is changing while the pitch remains the same. That's because the V.C.F. is following the ADSR control voltage output so that the filter cutoff frequency "Fc" moves along with the ADSR shape. The depth is controlled by control "B", just as the V.C.O. depth was controlled by "A". Try diferent ADSR settings as in the V.C.O. example to see how the same transient patched to a different point can control a diffeent part of the final sound.

Up until now, the loudness of the sound has been constant. Bring the "Fc" control and the "Q" control back to their original positions and turn down the "A" and "B" controls so that the ADSR will not affect the V.C.O. or V.C.F.. Now put the V.C.A. switch "C" in the ADSR (/\) position. This connects the ADSR output voltage to the V.C.A. control voltage input. The volume of the sound will now follow the contour of the ADSR voltage shape. Vary the "AUTO", "OFF" and "GATED" positions on the ADSR repeat switch to see how the same settings that affected pitch and timbre are now affecting loudness.

This little experiment served to illustrate how a controller can affect pitch, timbre and loudness in similar ways. This is why voltage control is such a powerful concept. Because of this, it is important that the controllers should be able to be connected to as many places as possible for the greatest versatility in sound manipulation capability. This is the basis behind the patching system used on Octave-plateau synthesizers.

ADSR Repeat



The ADSR transient will begin on each key depression if the "ADSR REPEAT" switch is in the "OFF" position. In the "AUTO" position, the ADSR constantly repeats at a rate determined by the "LFO RATE" slider setting. In the "GATED" position, the repeating will only occur when a key is held down.

If the attack is set too long, the repeat mode will appear not to be functioning because the time it takes the ADSR to rise from it's zero value may be much longer than the LFO speed setting. Also, if the release is set too long, the transients may run into each other and seem as though they are at a constant level. Because of this, when using the repeat mode, it is necessary to set the LFO rate to a speed compatible with the ADSR settings.

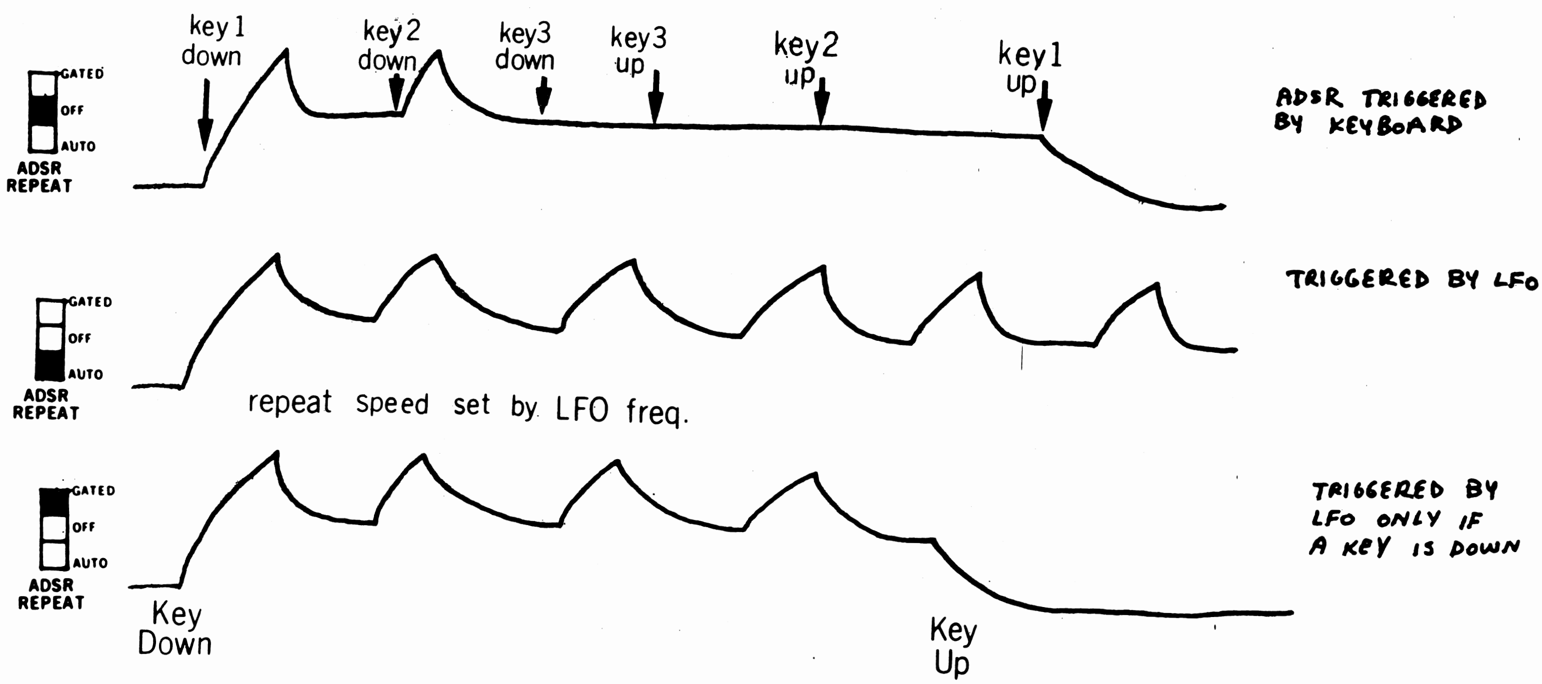


Fig. 14: Illustration of three ADSR triggering modes.

GATE Function

The keyboard generates a voltage called a "GATE" whenever any key is depressed. If no keys are down, the gate is zero, if one or more keys are down, the gate goes to a positive level. On the KITTEN II, this gate exists at the V.C.A. switch so that the V.C.A. turns the volume on full whenever any key is down and turns it off when no keys are down.

AR Generator

On the SRM II, the gate signal is run through an "AR" generator which allows us to slow down the rise of this voltage according to the setting of the front panel slider marked "A" (attack); as well as slowing down the fall back to zero after the key is released according to the setting of the slider marked "R" (release). This voltage exists at the points marked (/‾\) on the SRM II front panel switches and is a useful transient in applications similar to those illustrating the use of the ADSR transient.

voltage controlled
VCO; oscillator

The V.C.O.'s on Octave-plateau synthesizers have several different output waveforms that are all simltaneously mixable- a feature that is very useful and yet rarely found on many synthesizers. Each waveform has its' own slider so that the amplitude can be adjusted independently. This allows different waveshapes to be mixed in any ratio to form complex waveforms useful as both controllers and sound sources.

The waveforms available are sawtooth, square, triangle, variable width pulse and sub-octave (the kitten II also has a sub-octave 2 waveform). Each waveform has its' own charactaristic tone and as such, mixing them together allows blending of these tones.

The tone of a particular waveform is determined by the "overtone" or "harmonic structure" that defines it. For instance, the purest waveform, the "sine" wave, has no overtones or harmonics. If we add a series of sinewaves together, each one octave higher than the previous one, the resulting waveshape will have a timbre and waveform that is charactaaristic of the number of sinewaves added together. By carefully combining sine wave multiples in particular proportions we can come up with the sawtooth, squarewave, etc found on a synthesizer. Since each one has its' own particular tone, each is useful for particular types of synthesis.

SAWTOOTH ( ): This waveform has a charactaristic "brassy" timbre and is most often used as a starting point for horn and chorus strings synthesis. As can be seen by its' overtone structure, it contains both odd and even harmonics.

When used as a controller, the sawtooth can be used to create "wooop wooop" sweeping oscillator or sweeping filter effects.

SQUARE ( ): This waveform has a "hollow" or "reedy" sound and is useful for clarinet or other reed instrument simulations. The overtone structure has only odd harmonics.

When used as a controller, the square wave can be used to generate "trill" effects whereby a pitch alternates between two specific values as the squarewave goes high and low.

TRIANGLE ( ): This waveform has the fewest harmonics and so sounds the mellowist or closest to a pure sine wave. It is often used for flute or percussion synthesis.

As a controller, a triangle wave can be used to generate "vibrato" effects.

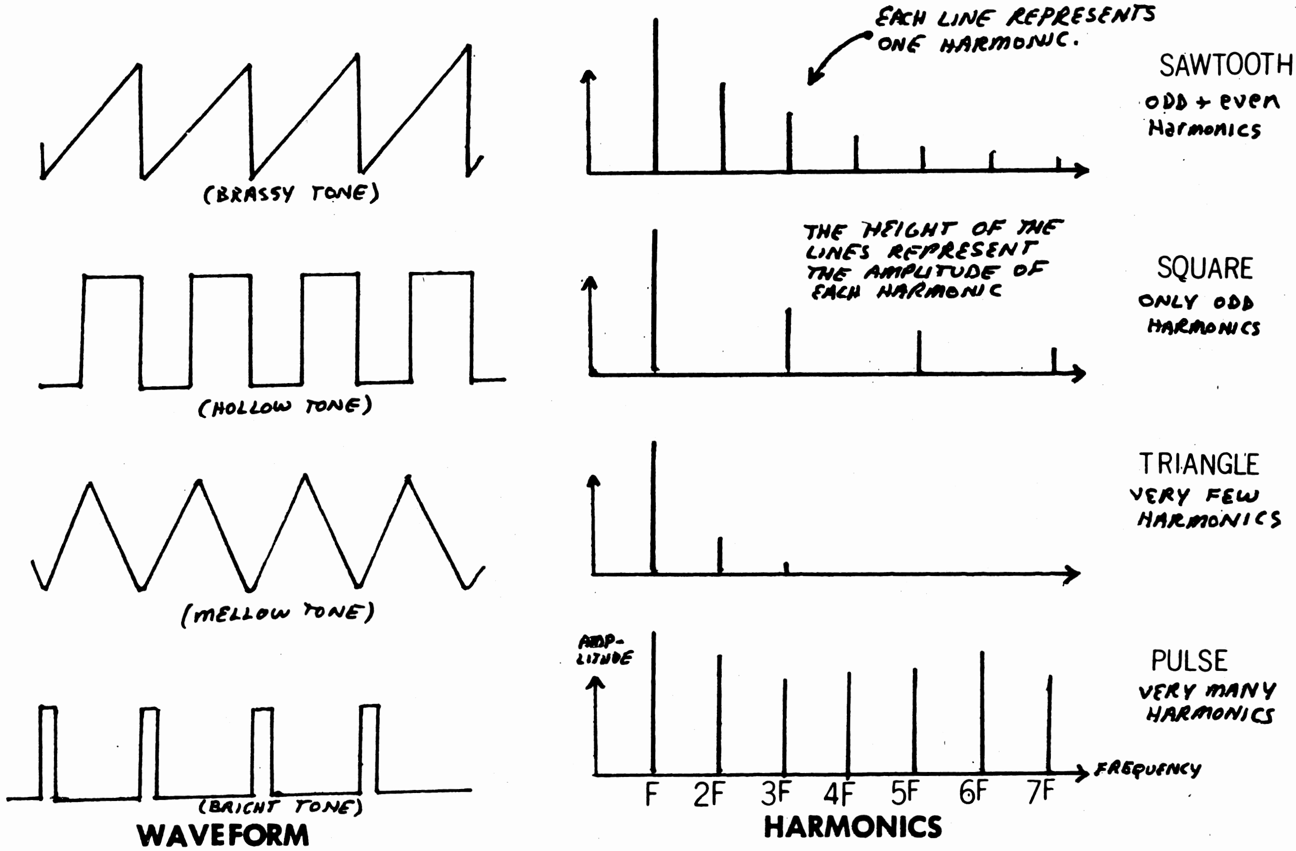


Fig. 15: The harmonic spectrum of a waveform determines its' particular timbre.

PULSE WAVE (⊓⊓): The pulse is a waveform whose shape (and thus timbre) is controlled by the position of the "PULSE WIDTH" control knob located above the pulse width volume slider. When the modulation switch above the knob is in the "DC" position, the width of the pulse is manually set by the knob so that when it is fully counter clockwise, the pulse becomes a square wave (only odd harmonics), and when the knob is fully clockwise, the pulse becomes narrow in width (many odd and even harmonics). The tone continulously varies for every position of the "pulse width" knob.

By varying the pulse width, the tone can be changed to that of a "twangy" guitar or clavinet simulation.

The pulse width can be automatically changed with the LFO sine wave output, resulting in a "chorusing" simulation as the pulse wave varys in width and alters its' harmonic structure at a rate determined by the LFO frequency. To select this mode, simply change the switch above the pulse width knob to the LFO sine (∿) position. The depth of the effect is set by the pulse width knob.

On the SRM II, the pulse width can also be modulated by the ADSR waveform. This allows interesting "plucked" string effects and is very useful for synthesizing guitar type sounds.

As a controller, the pulse width allows "trill" effects like the square wave, except that the variable pulse width allows changing of the trill time for the high and low portions of the trill.

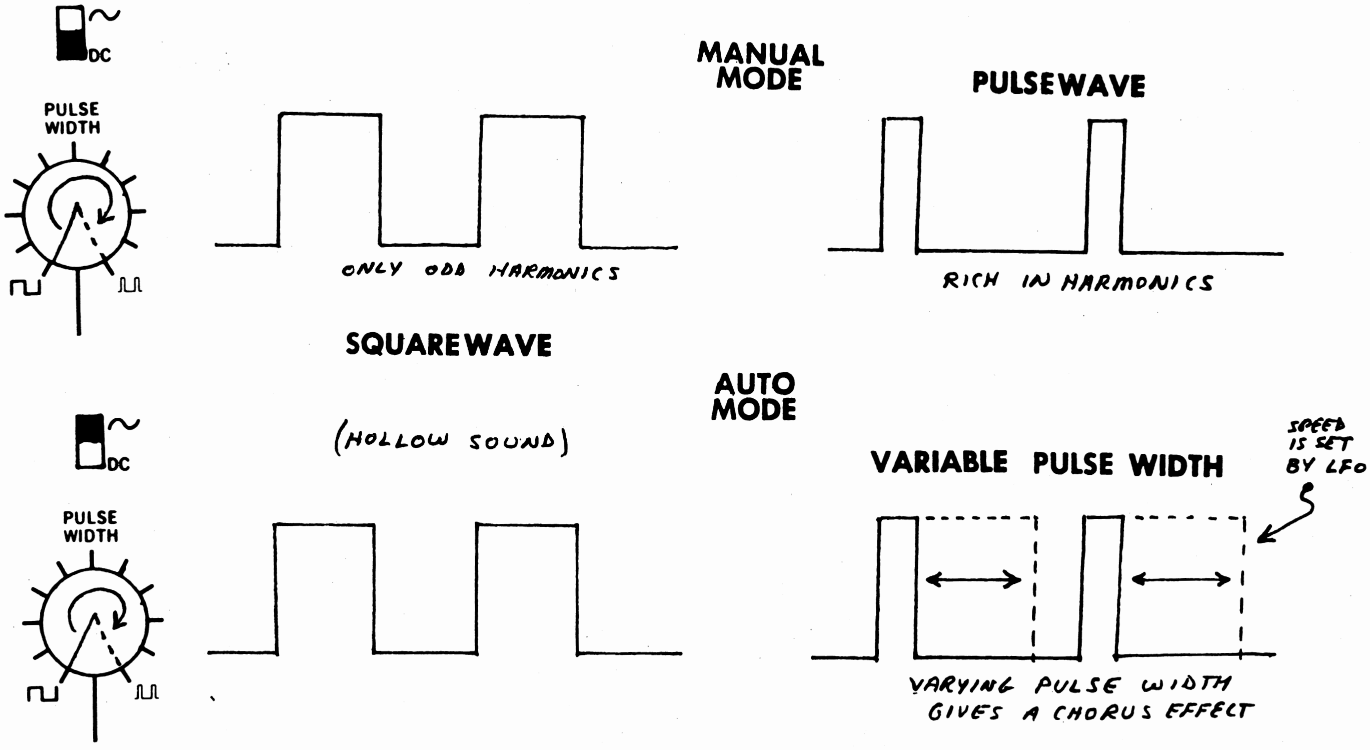


Fig. 16: The pulse width can be varied manually or automatically for a chorusing effect.

SUB-OCTAVE : The sub-octave is a square wave that is one octave below the normal oscillator waveforms. It posseses the same tonal characteristics as the normal square wave except for its' pitch. The sub octave is useful for adding a fuller sound to the oscillator because it simulates the presence of a second oscillator tuned one octave below the original sound.

SUB-OCTAVE 2: (KITTEN ONLY) This is an additional sub-octave square wave that is two octaves below the normal oscillator frequency. When used in conjunction with the sub-octave 1 waveform, the sound of three oscillators can be simulated with the single Kitten oscillator.

USING THE WAVEFORM MIXER AS A SEQUENCER

When we refer to the VCO waveforms as modulation sources we mean that the output of a VCO can be used to modulate another VCO (as on the SRM) or the VCF. By connecting the VCO1 output to the VCF modulation input, a "sequencer effect" can be obtained. To do this, we use the VCF as an oscillator by turning the "Q" up all the way. By setting the patch shown in the accompanying figure, the output of VCO1 is connected to the VCF modulation input so that the filter will follow the VCO waveform selected on the audio mixer.

By mixing the sub-octave square wave with the pulse wave, a four-step wave is produced. The shape of this waveform depends on the volume slider settings. On the Kitten, the sub-octave 2 waveform can also be used to get an eight-step waveform. When the oscillator is brought down to sub-audio frequency, the pitch of the oscillating filter will change according to the voltage level of each step in the waveform, resulting in a sequencing-note effect. The speed of the sequence will be set by the VCO frequency, the particular notes will be set by the waveform mixer settings and the overall range of notes is set by the modulation depth of the VCF control voltage input. Different rhythms can be obtained by varying the VCO pulse width control.

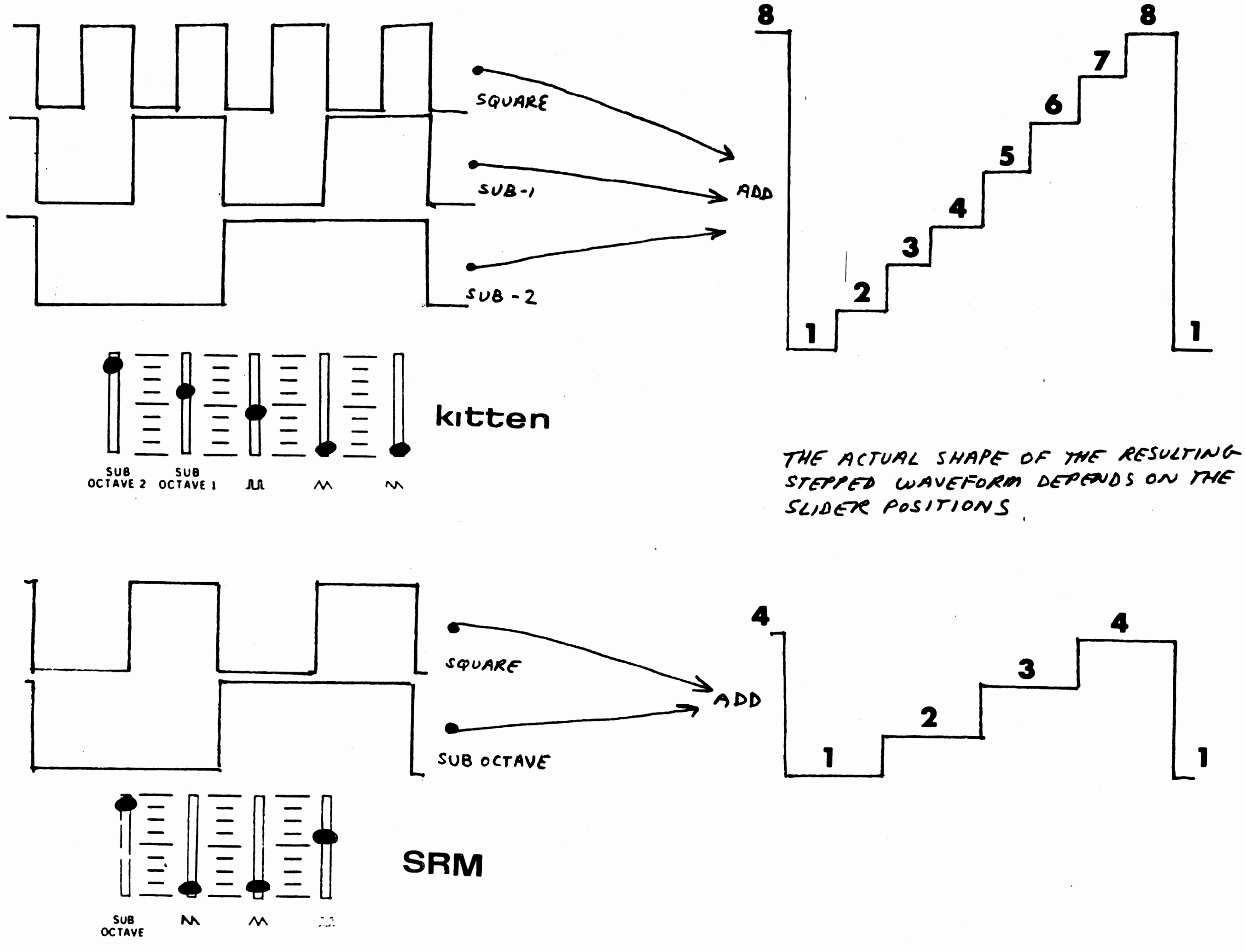
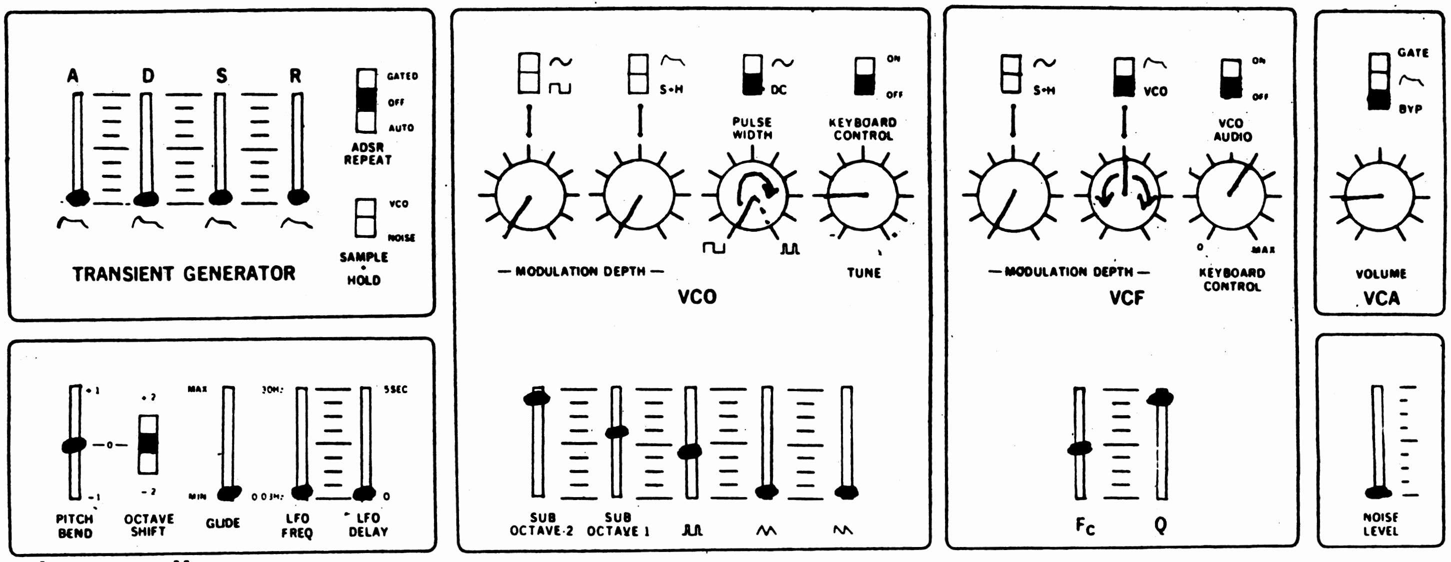
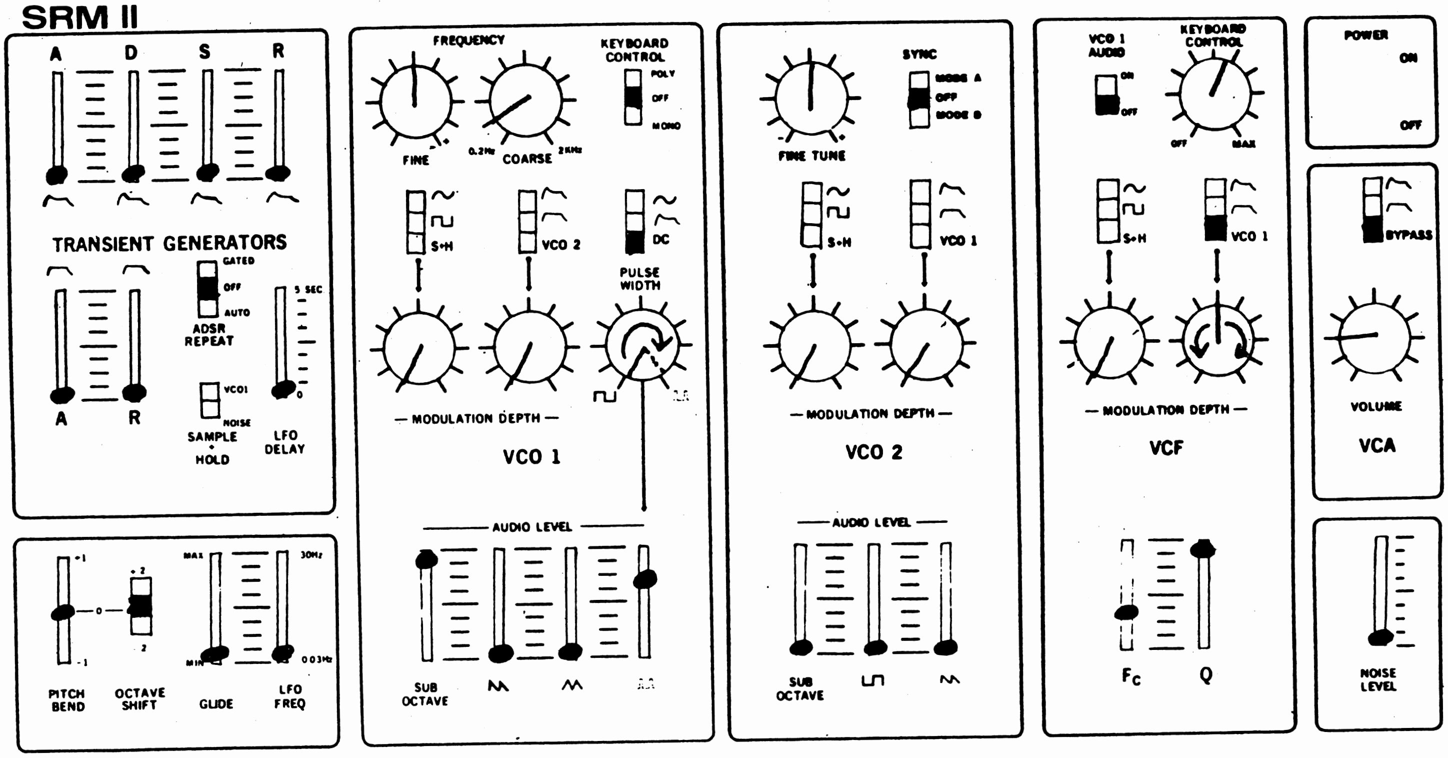


Fig. 17: By mixing waveforms together, a sequencer effect can be obtained when the VCO is used as a controller with the VCF in oscillation mode (i.e., Q up all of the way).



kitten II

SRM II



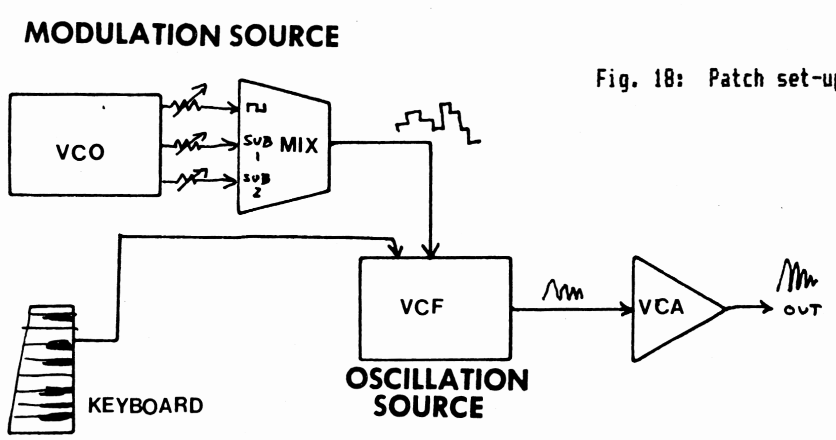


Fig. 18: Patch set-up for the sequencer effect.

VCO SYNC (srm II)

The "SYNC" switch on the SRM II VCO 2 allows both SRM oscillators to be locked together to form one complex oscillator. VCO 2 is unaffected by the "SYNC" switch, however, there is a drastic change in the tonal quality of the VCO 1 waveforms.
When the "SYNC" is turned on, VCO 1 is forced to follow VCO 2's frequency, resulting in complex waveforms whose shape depend upon the frequency spacing between the two oscillators. Since VCO1 has four different waveforms, each one will have its' own particular "sync timbre". The reason for this is that VCO 1 is forced to "lock" on or SYNCronize to a multiple of VCO2's frequency. From the figure, we can see that in the normal mode (sync off), the VCO2 and VCO1 waveforms can be at any position with relation to each other. This is the typical "oscillator chorusing" effect. In the sync mode, the waveforms generated by VCO1 are forced to fall within the constraints of VCO2's waveforms, so that trying to tune VCO1 away from VCO2 will result in a complex waveform from VCO1. From this, it is obvious that if VCO1 is tuned to a frequency below that of VCO2, the output waveform from VCO1 will be greatly diminished in volume because it is forced to reset at the VCO2 frequency and can not rise to its' normal level.

Two modes of sync are available:

Mode A is a "hard sync" in that VCO2 has a drastic effect on the VCO1 waveforms. Here, VCO1 is actually turned off for half of each VCO2 cycle, resulting in the waveform shown in the figure.

Mode B is the standard sync mode found on most synthesizers. It produces a charactaristic "vowel-like" sound. Technically, VCO1 is merely reset on each VCO2 reset pulse, resulting in a less complex waveform, as shown in the figure.

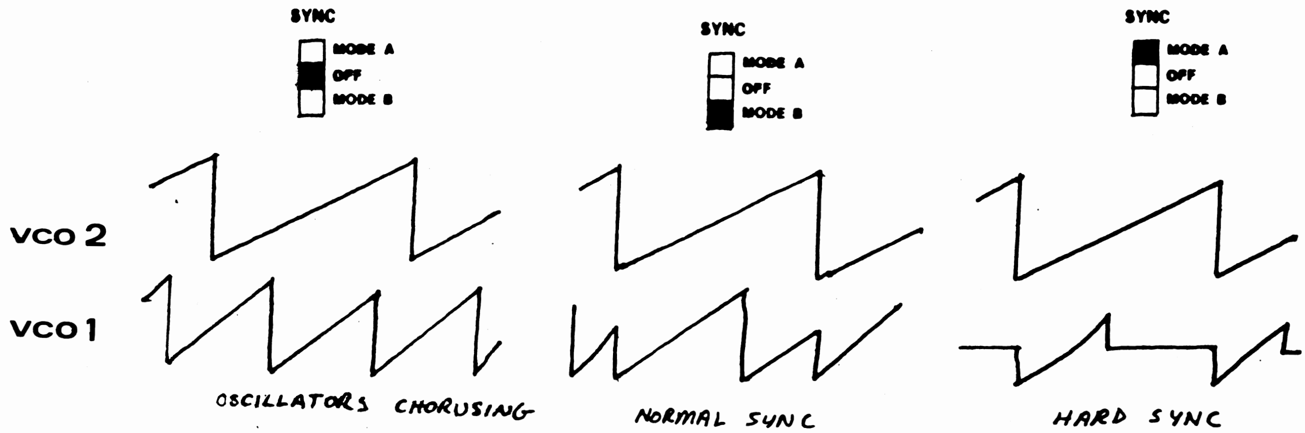


Fig. 19: Comparison of sync mode A and B on the SRM II.

KEYBOARD CONTROL SWITCH

srm II
Poly/Mono switch

VCO2 always plays the lowest note depressed on the keyboard. VCO1 can be made to follow the lowest or highest note on the keyboard.

In the "MONO" position, VCO1 plays the low notes along with VCO2 and the instrument becomes a standard monophonic synthesizer.

In the "POLY" position, VCO1 plays the highest note depressed on the keyboard. Thus, if a chord is played, VCO1 will play the highest note of the chord while VCO2 will play the lowest note of the chord. All of the notes inbetween will be ignored.

A special "TWO NOTE MEMORY" system comes into play when using the "POLY" position. On many "two note" synthesizers, VCO1 will drop to the lowest note down when the upper key is released. On the SRM II, a memory system is built into the second note so that it is remembered after being released. To use this feature, release the upper note before releasing the lower note so that your hand gradually "rolls" over the keys. An opposite "roll" (thumb up first) will cause the lower note to jump to the higher note- a technique useful for "trilling" between notes in the poly mode.

With the keyboard control switch in the "OFF" position VCO1 does not follow the keyboard at all- it is in a "drone" mode. This position is useful when using VCO1 as a modulation oscillator (when using the abovementioned sequencer effect, for instance), since the oscillator frequency would often be set to a point and left alone.

kitten II
Keyboard Control switch

On the Kitten II, the frequency control knob has a range of approximately +/-1 octave when the keyboard control is switched "ON". The VCO will follow the lowest note depressed, just as does VCO2 on the SRM II.

When the switch is in the "OFF" positon, the tune control becomes a wide range tuning control that spans well over 10 octaves. This allows low frequency operation of the VCO when using it as a modulation oscillator. The keyboard has no effect on the VCO frequency when in this mode.

voltage controlled VCF; filter

The VCF is a -24db/octave (four-pole) lowpass type with adjustable resonance (Q). This type of filter attenuates all frequencies above the cutoff frequency (Fc) setting.

With the "Q" control at minimum, moving the Fc slider up will let more harmonics pass so that the timbre of the sound becomes brighter. As the filter cutoff is brought down, less harmonics can pass through so that the sound becomes mellower.

The resonance, or "Q", emphasizes any harmonics that lie around the Fc setting. If the "Q" is turned up to a bit more than midway, moving the Fc slider will produce a characteristic synthesizer "waaoow" sound. If the "Q" is turned up to maximum, the VCF turns into an oscillator, producing a sine wave output whose frequency depends upon the Fc setting. No oscillator sound will pass through the filter in this mode of operation.

In the oscillation mode, the filter can be used as an oscillator since it can be made to track the keyboard with the "KEYBOARD CONTROL" knob.

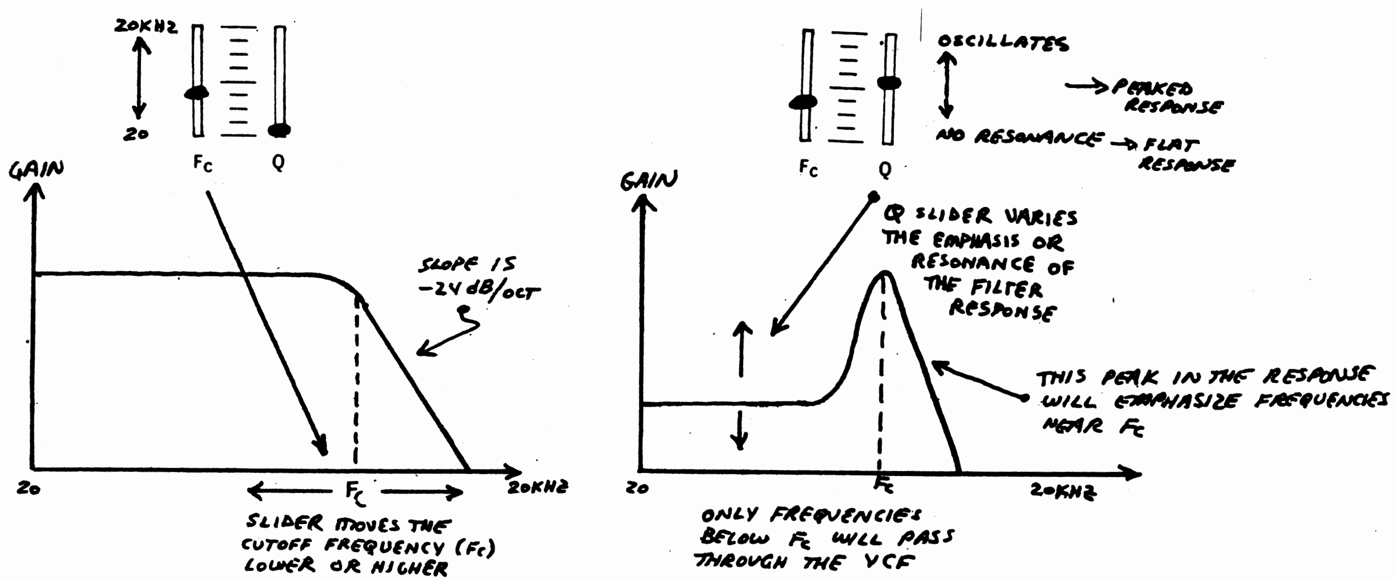


Fig. 20: The effect of the Fc and Q controls on the VCF.

VCO Audio on/off

The VCO on/off switch located in the VCF section, allows the audio from VCO1 (VCO for the Kitten) to be silenced. This is a useful function when the VCO is being used as a modulation oscillator because it allows the low frequency "clicks" to be silenced. Also, in a situation where VCO 2 is being modulated by VCO 1 to obtain complex FM sounds, VCO1 can be silenced so that only the modulated VCO 2 sound will be heard.

FILTER KEYBOARD TRACKING

When the "KEYBOARD CONTROL" knob is in the "0" position, the filter cutoff frequency will remain at the Fc slider setting, unaffected by the keyboard. As the knob is turned up, the keyboard will affect the Fc more and more until at maximum position, the cutoff frequency will shift by approximately two octaves for each change of one octave on the keyboard. For the settings inbetween, the keyboard response will be proportional to the knob setting. At settings below the "1 o'clock" position, a one octave change on the keyboard will produce a change in Fc of less than one octave, while settings greater than the 1 O'clock position will cause greater than one octave changes in Fc for each one octave keyboard change.

To tune the Fc reponse to one octave keyboard tracking, first hit the lowest C note on the keyboard and then set the filter in the oscillation mode at a frequency that is comfortable (e.g. C below middle A). Next, hit an octave C note and turn the "Keyboard Control" knob until the Fc is one octave above the original C. The position of the knob should be about "1 o'clock" when the 1 volt/octave setting is finally found. After setting this tracking adjustment, the VCF can be played as an oscillator with the standard keyboard reponse.

Why have a variable keyboard tracking control?

The filter is a very important part of the synthesizer because it determines the dynamic timbral changes of the final sound. Imagine having the VCO set to a particular frequency with the low C key down. Now suppose you want to set the VCF cutoff frequency so that the VCO produces a mellow sound. To do this, the Fc must be set slightly above the VCO frequency so that most of the harmonic overtones are filtered out. Consider what happens if the filter cannot track the keyboard. If the Fc is set to the VCO frequency when the low C is down, the tone becomes mellow, as desired. But what happens if the next octave C key is hit? The VCO frequency jumps up by one octave, but the filter cutoff remains at its' original value slightly above the low C note. Since the VCF has a rolloff of -24dB/octave, the new VCO note will be attenuated by -24dB, resulting in a noticably diminished amplitude. The next higher C key will be down by -48dB and so on. So, although the original note was mellow, the keyboard did not tell the VCF to change its' Fc along with the new notes, resulting in an unbalanced keyboard response.

To correct this, the VCF tracking should be set to correspond to the VCO tracking, which is 1 volt/octave. To do this, one must follow the adjustment explained above in the filter oscillation mode paragraph.

The filter "undertracking" mode is when the keyboard control is set to less than 1 volt per octave so that the sound would be brighter at the bottom of the keyboard and mellower at the top of the keyboard.

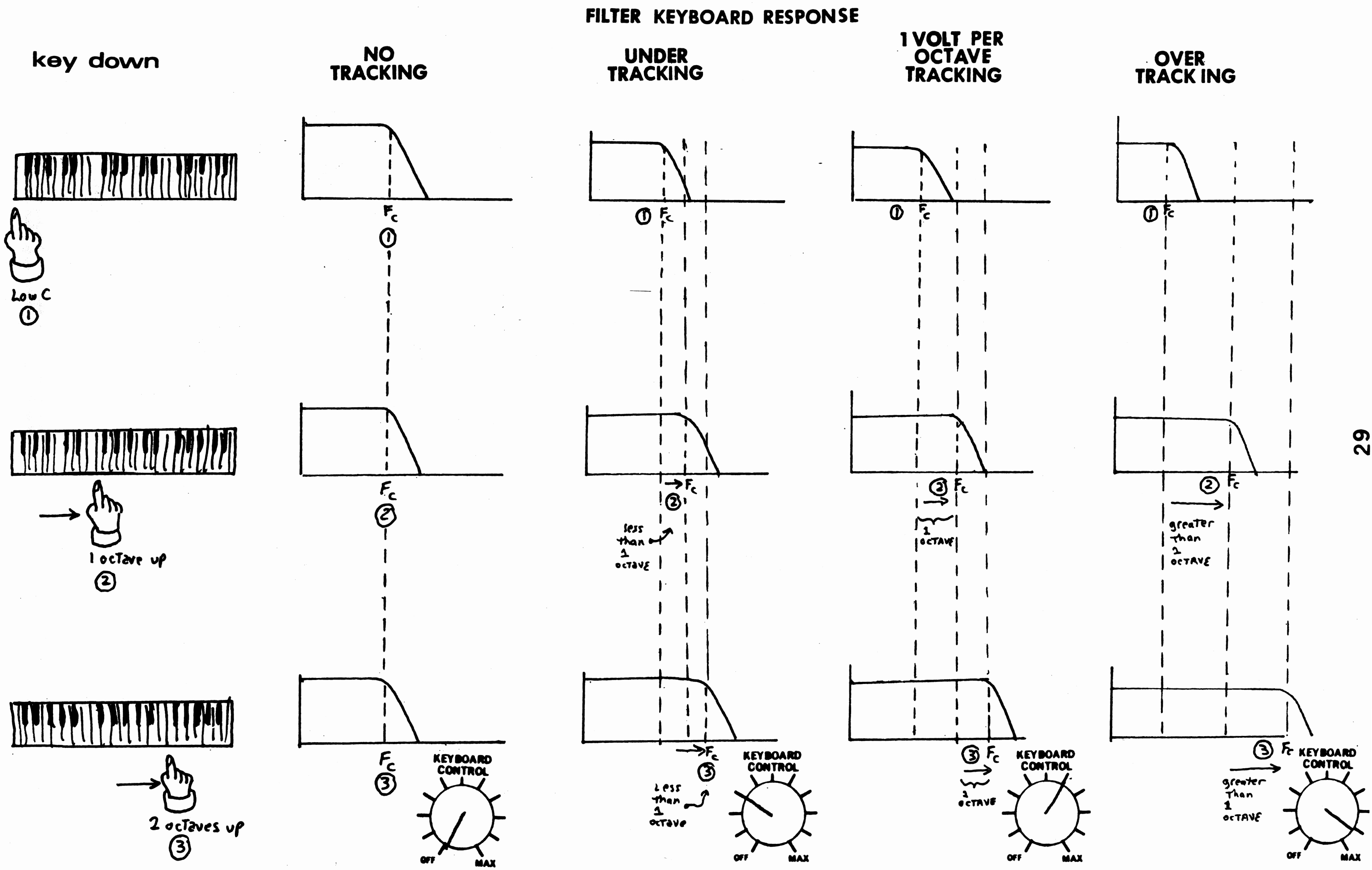


Fig. 21: Filter keyboard tracking depends upon the Keyboard Control setting.

The filter "overtracking" mode is when the control is set to grater than 1 volt per octave so that the sound becomes brighter at the top of the keyboard and mellower at the bottom of the keyboard.

The particular tracking used depends on the keyboard response desired.

VCA; voltage controlled amplifier

The VCA is the last link in the synthesizer chain since it controls the loudness contour of the final sound. The contour obtained depends on the modulation source used to vary the VCA reponse.

In an SRM II, the VCA can be modulated bv the ADSR or the AR by selecting the respective position on the VCA three-position modulation source switch. Repeated "mandolin" type effects can be obtained by modulating the VCA with the ADSR in the "auto" or "gated" mode. Straight keying effects can be obtained by using the AR.

The Kitten II uses a "GATE" signal instead of the AR. The gate is essentially an AR with fast attack and fast release. It can be used to turn the sound on when a key is down while the VCF is being modulated with the ADSR.

The "BYPASS" mode on the VCA turns the volume on full, regardless of any keys being down- as though the VCA were being "bypassed". This is useful when a sound is being set up so that you don't have to hold a key down to hear the results of your control manipulation.

The output level of the entire synthesizer is set by the "VOLUME" knob in the VCA section.

noise

The "NOISE LEVEL" slider sets the level of white noise mixed into the VCF. White noise is used for simulating "wind", "waves on an ocean", "explosions", etc., i.e. any "unpitched sound". By changing the Fc on the VCF, different noise timbres can be obtained.

The noise level does not affect the noise input to the Sample and Hold- this is internally connected to always be at maximum.

KEYBOARD and associated functions

DIGITAL KEYBOARD

The CAT SRM II and KITTEN II both utilize a new digital keyboard system. This design incorporates a single contact per key, dramatically reducing the cleaning and adjusting of dual contact-wires as is often necessary with standard analog keyboards. The digital design also provides scanned digital keyboard data output that allows keyboard information to be connected to a digital computer serial input data line when the proper interfacing circuitry is used.

Two Note Memory

The keyboard provides information for the highest and lowest notes down. This information is sent to VCO1 and VCO2 on the SRMII oscillators (only low-note information is sent to the Kitten II oscillator). The control voltage and gate for the low and high notes is available on the rear panel patch bay so that slave operation of synthesizers is possible with the slave playing either the high or low note. This allows a Kitten II to be played as a two note synthesizer when used with a slave instrument.

The two-note system requires a particular keyboard technique. When two or more notes are down, VCO1 will accept the higher note (if its' in the "POLY" mode) while VCO2 will accept the lower note. Any notes inbetween are ignored. To memorize both notes, the high note is released first. This results in VCO1 playing the higher note even after it is released. The reason for this is so that the common "glitching" problem associated with non-memorized two-note keyboards is eliminated. This glitch becomes apparent when an envelope with a long release is used so that the note contines for a short time after releasing the keys. Without the two-note memory, VCO1 would jump down to the last note played and both oscillators would decay playing the same note. If an interval was being played, this jumping effect of VCO1 would be very obvious and would make the technique sound erratic. By allowing VCO1 to hold the last note played, the interval continues to play even after releasing the notes, so that the decaying oscillators sound clean in the two note mode.

If the lower of the two notes is released first, VCO2 will jump to the next note down. If an interval is being played, this permits "trilling" between notes in the two-note mode.

Note that if you are in the two-note mode and play a passage that involves "walking" your fingers from the top to the bottom of the keyboard, it is possible that you may have a lower note down while releasing an upper note. Of course, this would cause the upper note to memorize while the lower note is playing, but the net result is that it sounds as though VCO1 is always trailing behind the notes played by VCO2. To overcome this, simply make sure that you fully release the last note pressed before hitting a new note so that only one note at a time is depressed while playing quick passages. If the two-note mode is not necessary, fastest keyboard technique is possible in the "MONO" mode.

PITCH BEND

This slider shifts the pitch of the entire unit up or down by one octave. The center position of the slider gives no pitch change and has a "locking" detent to allow "feeling" for the zero position.

OCTAVE SHIFT

The octave shift switch transposes the pitch of the entire unit up or down by 2 octaves.

Note that when the pitch bend and octave shift are both in their zero positions, the instrument will be tuned such that the second octave "A" note is approximately 440Hz with the VCO tuning controls in the 12 o'clock position.

GLIDE

As this slider is increased upwards, the glide time between notes played on the keyboard increases. The action of this slider is defeated by using the optional "GLIDE on/off FOOTSWITCH" that plugs into the rear of the unit. If a particular glide rate is set with the slider and the footswitch is plugged in, the glide will only occur when the footswitch is depressed. This allows the insertion of the glide effect only during small parts of long passages.

The glide feature is designed such that the pitch will glide to the new note even after the new note is released.

LFO and associated functions

The LFO is a special purpose oscillator which is dedicated to low frequency (.2Hz - 20Hz) operation. It is a dedicated control voltage source which determines the repeat rate of several synthesizer functions such as the ADSR repeat, sample and hold, LFO sine wave output (∿), and LFO square wave output(⊓⊔).

LFO Frequency

This slider sets the repeat rate of the LFO. The LFO L.E.D. monitor light provides a visual indication of the LFO speed.

The LFO produces both sine and square wave outputs which are available at the modulation switch positions labelled (∿) and (⊓⊔). The square wave is useful for trill effects when used to modulate the VCO frequency and for echo effects when used to modulate the VCF cutoff frequency. The sine wave is useful for vibrato and filter sweep effects as well as for pulse width chorusing effects (see VCO section for an explanation of pulse width effects).

LFO Delay

The LFO sine wave runs through an internal circuit which turns it off for a specified time period after each key is hit. This time period can be varied from zero to approximately five seconds, after which the sine wave gradually inceases to its full value. This effect is called "LFO DELAY" and is useful for delayed vibrato and delayed filter sweep effects. The delay function does not affect the LFO square wave.

SAMPLE & HOLD

The sample and hold is a circuit that takes a smooth or continuous input waveform and turns it into a stepped voltage. The waveforms that can be sampled come from either VCO1 or the noise source and the steps occur at the LFO frequency.

If the VCO1 position is selected on the S+H input switch, the waveforms selected on the VCO1 audio mixer will be sent to the S+H circuit. For example, if you want to turn the VCO1 sawtooth into a stepped waveform, put the S+H switch in the VCO1 position and turn up the sawtooth slider on the VCO1 mixer. This will cause a stepped sawtooth output to appear at all of the switch positions marked S+H. Any mixture of VCO 1 waveforms will enter the S+H circuitry in this mode, so that various stepped waveforms can be obtained. Note that if the sliders are all down, the S+H will have no input and there will be no signal at the S+H switch poisitions.

To use the noise, simply place the S+H input switch to the "NOISE" position. This will place a stepped random voltage at all of the S+H modulation switch positions.

S&H EXPERIMENTS

To see how the S+H operates set the front panel as shown in the accompanying figure. First place the modulation switch "A" in the VCO1 position. This allows VCO1 to modulate the filter, which is acting as an oscillator in this example. Turn up the VCO1 sawtooth slider and vary the VCO1 frequency (coarse) control. Notice that when the control is fully counter clockwise (low frequency) the filter produces a "woooop wooop" sound because its' frequency is following the vco sawtooth waveform. Try some other VCO1 waveform combinations to notice their effect on the filter frequency.

Now connect the S+H output to the filter modulation input by turning down knob "B" and turning up knob "C". The input to the S+H" is the VCO output, so if all of the VCO1 waveform sliders are down, the VCF frequency doesn't change. Turn up the VCO1 sawtooth and listen to the effect on the VCF frequency. By setting the VCO frequency very low, you can hear how the steps produced correspond to the sawtooth waveform. Vary the LFO frequency slider "D" to see how the steps are directly affected by it. Now alternate between modulation depth controls "B" and "C" to better understand the correlation between the unsampled and sampled VCO waveforms. Try the S+H switch in the "NOISE" position to see the randomness of the steps produced.

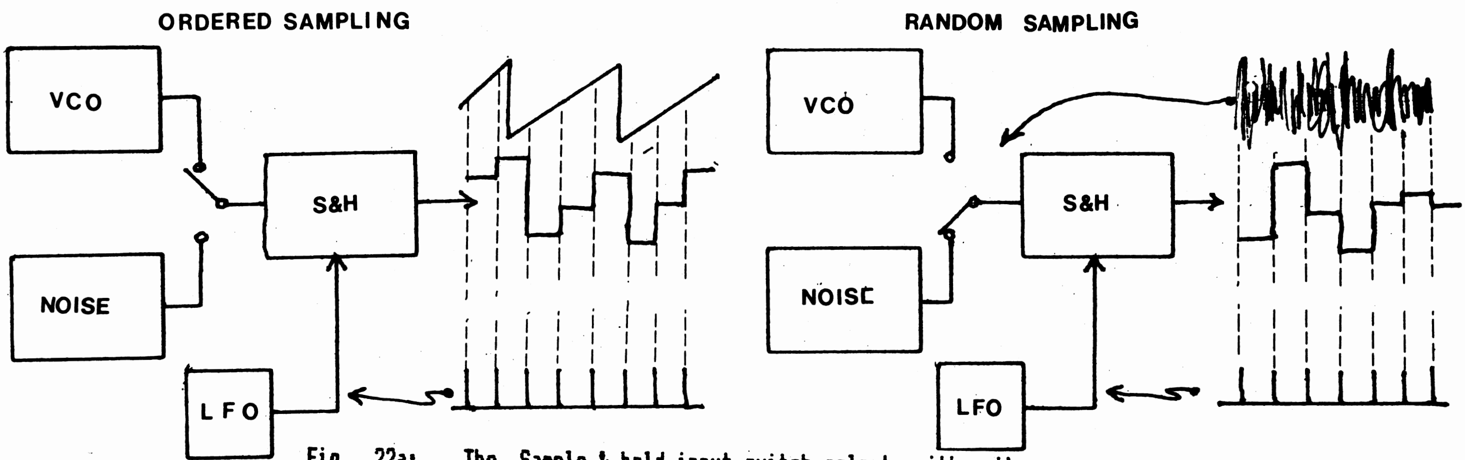


Fig. 22a: The Sample & hold input switch selects either the VCO or noise source. The sampled output waveform is available at all modulation switch positions labelled "S&H".

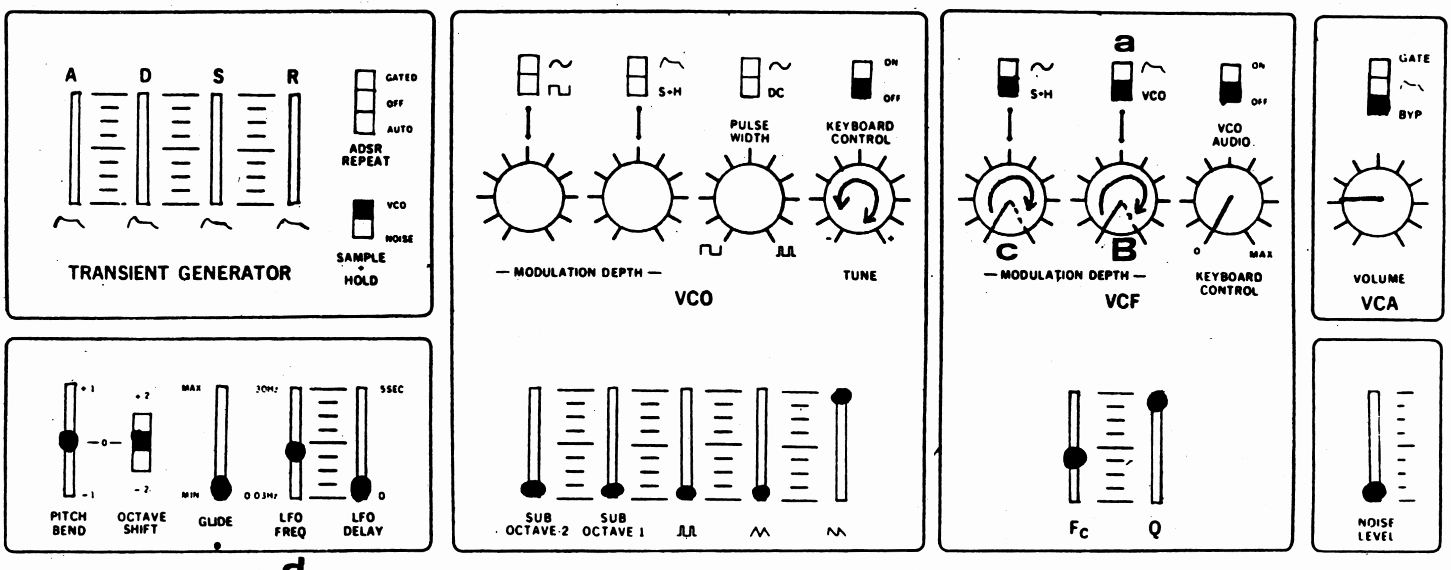
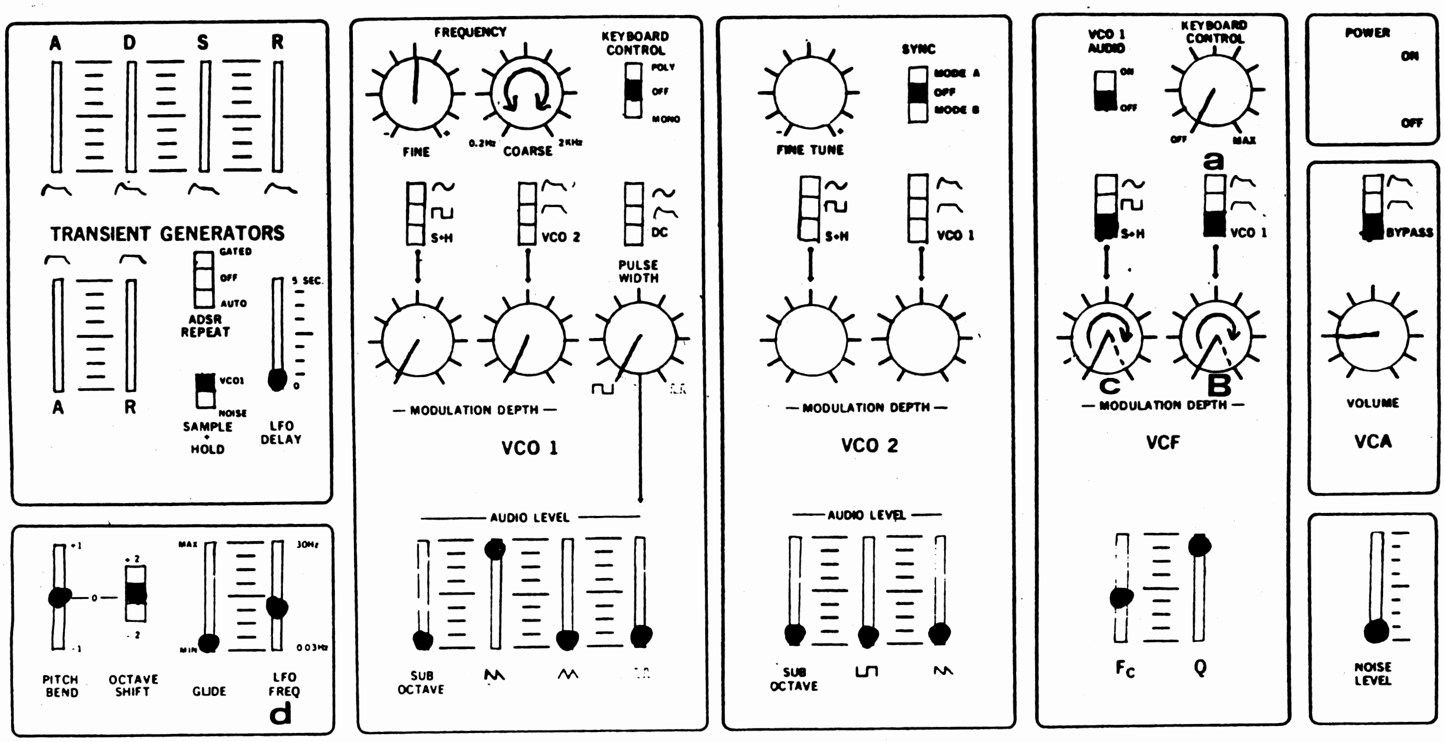


Fig. 22b: Panel setting for the Sample & Hold experiments.



PATCH BAY

Not all synthesizers have a rear panel patch bay. Many simply provide an audio output for connecting the instrument to an amplifer. Octave-plateau synthesizers use an extensive patch bay system which allows interfacing to synthesizer accessories, foot pedals and other synthesizers for future expansion or extended control capability.

AUDIO OUTPUTS

Two audio outputs are provided for connecting the instrument to an amplification system. They can be used independently or at the same time.

The "LOW LEVEL" output is the standard output to be used with preamplified systems such as guitar amplifiers, PA microphone inputs or other high-gain inputs.

The "HIGH LEVEL" output is useful for driving signal inputs that require large voltage swings, such as the auxiliary inputs on stereos and tape decks or the input of a non-preamplifed power amp. This output is also capable of driving headphones directly for monitoring the synthesizer output.

Both outputs are protected against short circuiting and are low impedance sources.

AUDIO INPUT

The external "AUDIO INPUT" allows an outside audio source to be connected to the input of the VCF for processing by the synthesizer. Instruments that are high level, such as other synthesizers, can be connected directly with no furthur amplification necessary. Low level devices such as electric guitars or microphones must be preamplified to increase their output level for proper processing.

GLIDE FOOT SWITCH

This jack allows a foot switch to be connected that will defeat the glide function until it is depressed. The proper footswitch for this application can be obtained from any authorized Octave-plateau dealer.

foot pedal inputs

The pedal inputs allow certain synthesizer parameters to be varied by an accessory footpedal or other control voltage source. These jacks are "stereo", meaning that they provide control voltage input on the tip and pedal power on the ring. The proper pedal can be obtained from any authorized Octave-plateau dealer. These pedals have a "sensitivity" control knob installed on one side along with a stereo input jack. The pedal connects to the instrument via a stereo cable

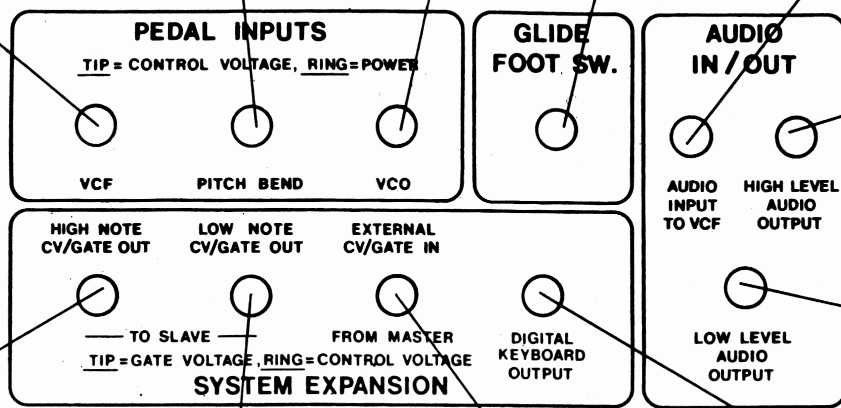
VCO1 PEDAL INPUT
Allows an external foot pedal to sweep VCO1 frequency for pitch bend or complex Sync effects (on SRM II)

EXTERNAL AUDIO INPUT
Allows other high-level instruments to be processed by the VCF/VCA.

PITCH BEND PEDAL INPUT
Allows an external foot pedal to sweep VCO1, VCO2 and the VCF simultaneously.

GLIDE FOOT SWITCH INPUT
Allows an external foot switch to turn the glide function on or off.

VCF PEDAL INPUT
Allows an external foot pedal to sweep the VCF cutoff frequency.



HIGH LEVEL AUDIO OUTPUT
Provides a high-level drive output for directly driving line inputs on tape decks or mixing boards.

LOW LEVEL AUDIO OUTPUT
Provides a low-level drive output for connection to guitar amplifiers and other pre-amplified stages.

HIGH-NOTE TO SLAVE OUTPUT
Provides high-note control voltage and a gate signal (0 to + 15 volts) that goes high when two or more keys are depressed. This output allows a slave synthesizer to play the highest key down for two-voice polyphonic effects.

LOW-NOTE TO SLAVE OUTPUT
Provides low-note control voltage and a gate signal (0 to + 15 volts) that goes high when one or more keys are depressed.

FROM MASTER INPUT
Accepts control voltage and gate information from external sources (such as synthesizers or sequencers) for controlling the synthesizer as a slave. The keyboard remains connected to allow transposing.

DIGITAL KEYBOARD OUTPUT
Provides serial digital Keyboard data for processing by external devices such as digital computers or other polyphonic peripherals.

- The TO SLAVE/FROM MASTER system allows CATS and KITTENS to connect together in a MASTER/SLAVE arrangement with a common stero cable. This allows multiple synthesizers to be played off of one master keyboard.

- The pedal inputs allow interfacing to external voltage generating devices such as a CATSTICK. The tip of the pedal input jacks is the CV input so that a common mono guitar cable can be used for interfacing.

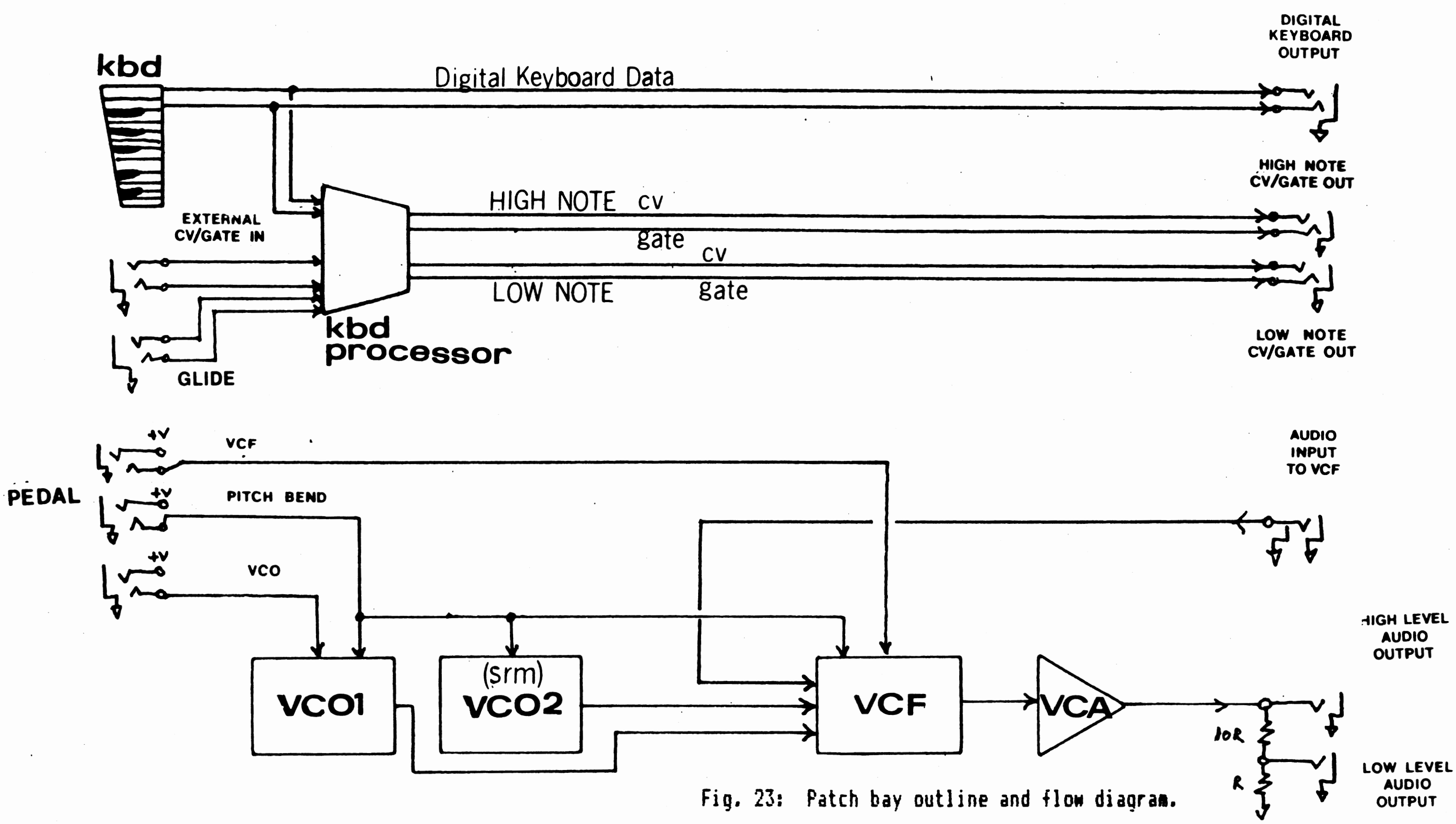


Fig. 23: Patch bay outline and flow diagram.

(provided with the pedal) and the maximum modulation depth is set by the control knob on the pedal.

For instance, if it is desired to pitch bend the unit up by one octave with the footpedal, the pedal would first be connected to the "pitch pedal" input jack. Then the knob on the side of the pedal would be adjusted for a one octave pitch change with the toe portion of the pedal fully depressed. This would cause no pitch change with the heel part down and a one octave pitch change with the toe part down. A set-up such as this would free one hand fron the task of pitch bending notes.

An external voltage generating device, such as a CATSTICK professional joystick controller, can be connected to the pedal inputs with a mono guitar cable. This cable will harmlessly short out the pedal power and access the control voltage input on its' tip.

VCO pedal

This input allows VCO1 on the SRMII (VCO on the Kitten II) to be varied upwards in pitch with a foot pedal. This is a useful effect when the VCO is placed in the "SYNC" mode (SRM only) and swept with the pedal.

PITCH pedal

This input allows pitch bending of the entire unit with an external footpedal. Wheras the "VCO pedal" input allows only VCO1 to be swept upwards in frequency, the "PITCH pedal" input allows both VCO's and the VCF to be swept simultaneously (VCO and VCF on Kitten II). This action is very similar to that of the "pitch bend" slider on the front panel.

VCF pedal

This input allows the VCF cutoff frequency (Fc) to be swept with an external pedal for "hands free" filter manipulation.

system expansion inputs

The system expansion inputs allow the synthesizer to be controlled by or to control other synthesizers.

DIGITAL KEYBOARD OUTPUT

This is a stereo jack which contains digital keyboard information in serial form. The tip of the stereo jack is a multiplexed clock/strobe line which provides reset and clock rate information to an external device such as a digital computer. The ring of the stereo jack is the keyboard data information which tells the external system which of the 37 keys are down. The accompanying figure illustrates the logic levels and phase relationship between these signals. The availability of this digitally scanned information allows polyphonic interfacing to a digital computer that has a compatible serial input line and proper level detection devices. Feel free to contact Octave-plateau for assistance in interfacing this output to your particular computer.

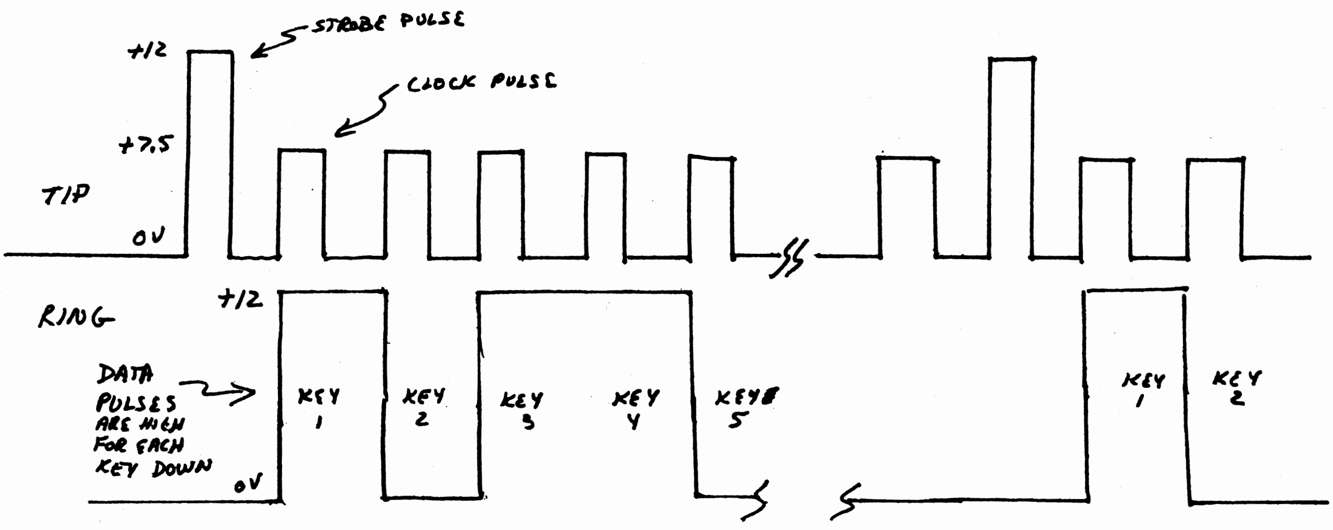


Fig. 24: Relationship between signals in the digital keyboard output.

cv/gate INPUT

This input allows external devices such as sequencers and other synthesizers to control the instrument by accepting control voltage (CV) and gate information. Connecting a control voltage and gate to this input makes the external controlling device act as a "master" and the instrument as a "slave". The keyboard remains connected when using this jack so that the control voltage coming into the "from master" jack is added to the original keyboard control voltage, allowing the keyboard to transpose any notes that are played on the master synthesizer or sequencer. For instance, if the synthesizer were being controlled by a sequencer, the keyboard could be used to transpose the sequence as it is playing. Or if were being controlled by another "master" synthesizer, the "slave" could be used for tuning intervals between both units since the slave can be played by using either the "master" or "slave" keyboards.

The control voltage input on the "from master" jack has a memory circuit connected to it that memorizes the last note coming in after the external gate goes low. This is so master synthesizers with unsampled control voltage outputs (like the early model Cat and Kittens) can be used as master controllers.

Remember when using this input that the last note played by the master will be held- even if the master is disconnected. This can cause the puzzling problem of having hit a high key on the master, disconnecting it from the slave, then trying to play the slave a little later on and wondering why it is totally out of key! The solution to this is to turn the slave power off and on to discharge the internal memory circuit and put the instrument back in its' original, unshifted key. Of course, this can be avoided by playing the lowest note on the master before disconnecting it.

HIGH/LOW NOTE TO-SLAVE OUTPUTS

These outputs provide control voltage and gate signals for controlling slave synthesizers or for memorizing passages played into an external sequencer.

The low-note gate will go from its' normally low (zero volts) state to a high state of approximately +12volts whenever one or more keys are down. The high note gate will make this transition when two or more keys are down. The low note control voltage output corresponds to the lowest note being depressed, regardless of the number of keys down, while the high note control voltage corresponds to the highest of all the keys depressed.

Connecting a slave to the low note output will allow both master and slave to play monophonically. Connecting the slave to the high note output allows a two-note polyphonic effect to be obtained from a Kitten II used as a master. The slave unit will only play when two or more notes are depressed, thus silencing its' output when only one key is down on the master.

EFFECTIVELY USING YOUR SYNTHESIZER

Proper Amplification:

The tonal range of a synthesizer is greater than that of any electrical or acoustical musical instrument. Keeping this in mind, it's important to remember that if this tonal range is to be realized, a high quality amplification system must be used.

For example, bass amplifiers are fine for reproducing synthesized bass guitar but terrible for most other sounds. The reason for this is that bass amplifiers are usually not made to reproduce high frequency sounds, thus making the synthesizer sound "dull" or "muddled".

The best type of amplification system to use for synthesizer reproduction is one that has both high frequency drivers and bass speakers so that both the high and low frequency sounds can be faithfully reproduced. For home or studio use, high fidelity speakers are an excellent choice. Of course, if such a system is not readily available, you should use whatever is handy; but keep in mind that the sound of your synthesizer will only be as good as the fidelity of your amplification system.

Using Sound Modifiers:

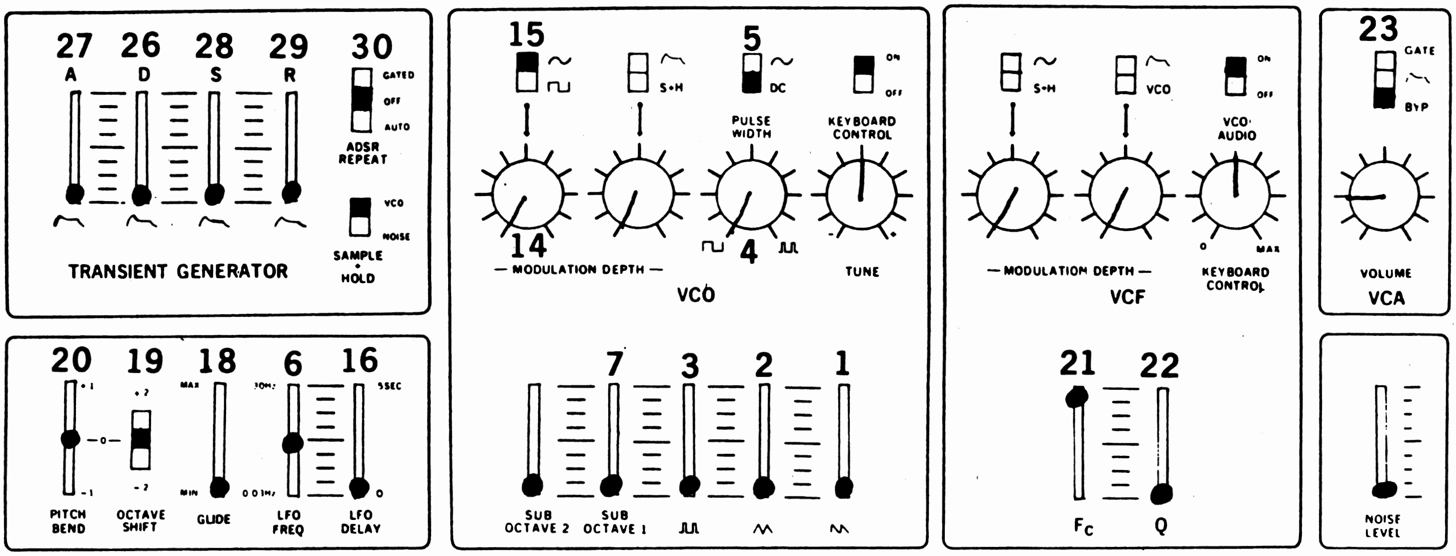
When external sound modification devices such as flangers, equalizers, distortion boosters, etc. are used to modify the synthesizer sounds, they should be used with discretion. For instance, don't attempt to synthesize a french horn while running the output of the synthesizer through a rotating speaker system meant for an organ. The synthesized sound of an organ would sound excellent through such a system, but the french horn would fall far short of your expectations. The point is that sound modifiers should be used with a specific purpose in mind. Know when to use them and why.

Reverberation and analog delay are among the more useful synthesizer accessories because they add fullness to sounds like synthesized strings. Equalizers also play an important role by allowing the tone to be tapered at the output to compensate for any deficiencies of the amplification system or to simulate the natural resonances of the instrument being simulated.

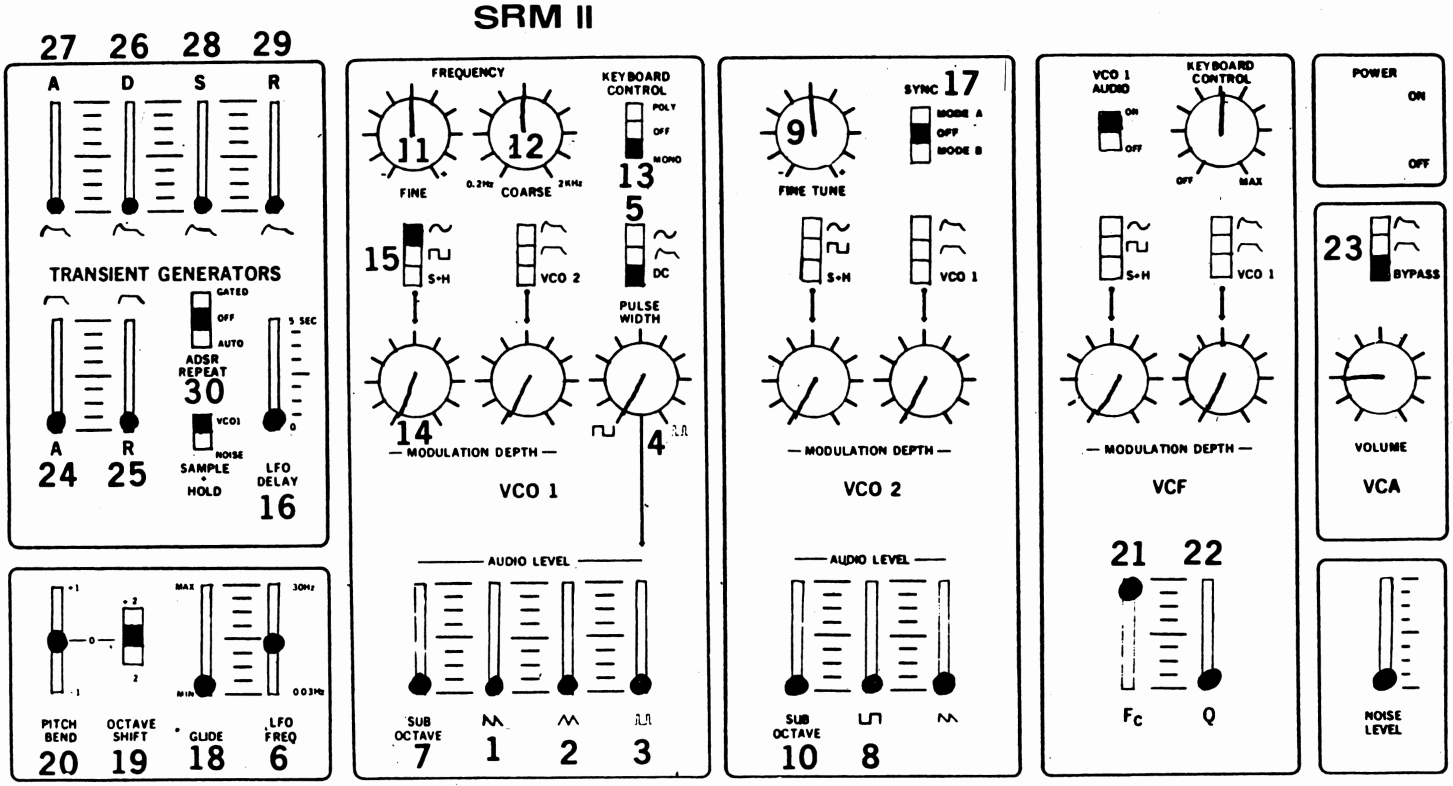
A synthesizer is often played with two hands- one on the keyboard and the other on the control panel. This enables you to add dynamics while playing the instrument so that the synthesized sounds are more interesting. Using pitch bending on guitar simulation or adding vibrato only during certain passages adds a feeling of realism to the final sound.The synthesizer is one of the few instruments that allow this type of dynamic variation and so the panel controls shouldn't be overlooked. When you sit down to play it, don't just play the keyboard, play the INSTRUMENT!

A Quick Demo and General Overview;

Kitten II



SRM II



- SET THE PANEL AS SHOWN BEFORE BEGINNING. DOUBLE CHECK TO SEE THAT ALL OF YOUR SETTINGS MATCH THE ILLUSTRATION.

- SET THE VOLUME CONTROL ON THE FAR RIGHT TO A COMFORTABLE LISTENING LEVEL.

• FOR A COMPLETE DEMO FOLLOW THESE STEPS AND REFER TO THE PANEL.

WAVEFORM MIXER

Turn up the slider marked ⩘(1) in the VCO1 section. Hit the lowest C note on the synthesizer. The sound you hear is a SAWTOOTH WAVEFORM. Now, turn this SAWTOOTH off and turn up the slider marked ⩘ (2) to the right of it. This is the sound of a TRIANGLE WAVEFORM. Turn this off and turn up the slider to the right of it labeled ⊓⊓ (3). This is the sound of a SQUAREWAVE. Vary the knob (4) above this slider and listen to how the tone changes. When the knob (4) is fully clockwise, the sound is that of a PULSE WAVEFORM. Place the switch (5) above knob (4) in the LFO SINEWAVE (∼) position. Now, the sound is automatically varying from a square wave to a pulse wave. This is called DYNAMIC PULSE WIDTH MODULATION. The speed of this effect is controlled by the LFO frequency (6). Do you see how it adds richness to the sound? Switch between the ∼ and DC position on switch (5) to better understand this effect.

Turn down all the audio level sliders except for the VCO1 ⩘ (1) slider. Now slowly turn up the Sub-Octave (7) slider. Notice how the sub-octave adds depth to the sound? Now try all of the VCO1 waveforms in different proportions by varying the VCO1 audio level sliders. This is called MIXING WAVEFORMS because each waveform has its own volume control slider. Since each waveform has its own charactaristic tone, the ability to mix them together is very important for creating complex tones.

SRM VCO2

Now try two oscillators. Turn the pulse width knob (4) fully clockwise. Turn up only the VCO1 ⊓⊓ (3) slider and the VCO2 ⊓⊔ (8) slider. Tune VCO1 and VCO2 together using the VCO2 fine tune (9). (A slight chorusing effect is desirable for added richness.) Now turn up the sub octave audio level on both VCO1 (7) and VCO2 (10). Do you hear the sub octaves make the 2 oscillators sound like 4? For a more pronounced effect tune VCO1 to one octave below VCO2 and experiment by playing some lines on the keyboard.

SRM poly/mono

Now turn down all of the audio level sliders except for the VCO1 ⩘ (1) slider and the VCO2 ⊓⊔ (8) slider. Bring the frequency control knobs (9), (11) and (12) to the 12 o'clock position. Tune VCO1 and VCO2 together using the VCO2 fine tune control (9). Hold down several notes at once on the keyboard. Both oscillators play the lowest key you have depressed. This is because the keyboard is set for **monophonic** response. To use the **2-note memory** system, place the keyboard control switch (13) in the poly position. Play **one** note. VCO1 and VCO2 are both playing that note. Now, while holding down this note, play another note above (to the right) of this note, so that **two** notes are down. VCO1 will play the highest note depressed and VCO2 will play the lowest note depressed. Any notes in-between are ignored. Now let go of the top note only, keeping the bottom note depressed. VCO1 is still playing the upper note, even though it's not down anymore. This is called a 2-note MEMORY and not all 2-note synthesizers have it. Most other synthesizers would have had VCO1 jump to the lower note rather than hold the interval.

TWO-NOTE KEYBOARD

With only the VCO1 ⩘ (1) slider and the VCO2 ⊓⊔ (8) slider up full, play any two notes on the keyboard. Let go of the lower note while keeping the higher note depressed. Now both oscillators will play this high note. Thus, the 2-note memory requires a particular fingering technique. To **memorize** two notes, let go of the upper one **first,** then the lower one. To **trill** between notes in the memory mode, let go of the lowest note first and both oscillators will jump to the upper note. The memory is designed this way to speed up the keyboard response in the 2-note mode.

PATCHING

Place the Poly/Mono switch (13) in the MONO position. Turn down all of the audio level sliders in the VCO1 and VCO2 sections except for the VCO1 ⩘ (1) slider which should be up all of the way. To add **vibrato** to this sound turn up the knob number (14) in the VCO1 section. You can hear how the steady oscillator tone varies in accordance with the sine wave (∼) next to the switch (15) above this knob. The depth of this vibrato is controlled by knob (14) and the speed of LFO frequency (6). Vary both of these to experiment with their effect.

The audio level sliders should all be off except for the VCO1 ⩘ (1) slider. With the modulation knob (14) up, you can hear a constant vibrato. Turn up the slider marked LFO DELAY (16) to the half way point. Now depress any key on the keyboard. Notice how the vibrato stops and comes on after a delay time. The setting of the LFO delay slider (16) determines the length of this delay. This effect occurs only on the LFO sine wave (∼).

SRM VCO SYNC

The audio level sliders should all be down except for the VCO1 ⩘ (1) slider. Turn off the vibrato by turning down the modulation depth knob (14). Turn on the sync switch (17) in the VCO2 section. VCO1 is now "SYNCHRONIZED" to VCO2. Sweep the VCO1 coarse frequency control (12) back and forth. Notice how the sync gives the oscillator a vowel-like tone. Try this with each of the 4 different VCO1 waveforms by turning up each of the sliders labeled sub-octave (7), ⩘ (1), ∼ (2) and ⩩ (3) one at a time. Notice how each waveform has its own particular sync tone.

GLIDE

Turn off the sync switch (17) and put all of the VCO tuning knobs (11) (12) (9) in the 12 o'clock position. Turn up only the VCO1 ⩘ (1) slider. Play a low C on the keyboard. Lift up this low C and play the highest C. Hear how the note changes immediately? Now turn up the GLIDE slider (18) to its fullest point. Play the low C note. Hear how the note glides? Now just touch the high C for a split second. Hear how the glide keeps on going until it reaches the high C note even after you release the key. This is the AUTOMATIC GLIDE feature that many other synthesizers **don't** have. They instead would stop the glide at the note it was at when you lifted the high C key.

OCTAVE SHIFT

All of the audio sliders should be down except for the VCO1 ⩘ (1) slider. Turn down the glide slider (18). Put the OCTAVE SHIFT switch (19) in the +2, -2 and back to 0 positions. See how it transposes the instrument by exactly 2 octaves?

PITCH BEND

Now try the pitch bend control (20). Feel the center-locking zone that lets you return to standard tuning by feeling for a "click".

vcf

All of the audio sliders should be down, except for the VCO1 (1) slider only. Slowly bring down the Fc slider (21) in the VCF section. Notice how the mellowness of the sound depends on where this slider is set. That's what the VCF, or voltage controlled filter does, it modifies the basic tone of the oscillator. Now, turn up the "Q" slider (22) to midway. Vary the Fc (21) slider. Hear a "wow" type of sound? That's what the "Q" does. It emphasizes overtones and gives a resonant sound to the oscillator.

VCF Q

SRM AR

All of the audio sliders should be down except for the VCO1 ⩘ (1) slider. Turn the Fc (21) fully up and the Q (21) fully down. Up until this point, the sound from the synthesizer has been constant. To make it respond to the keyboard, put the VCA patch switch (23) in the AR ⌒ position. Now, the sound comes out only when a key is depressed. To shape the sound, adjust the sliders labeled A (24) and R (25) in the TRANSIENT GENERATOR section. The A stands for "attack" and the R for "release". Experiment with them to see how these sliders affect the envelope of the sound.

ADSR

All of the audio level sliders should be down except in the VCO1 ⩘ (1) slider. Now put the VCA patch switch (23) in the ADSR ⌒ position. Depress a key. You should hear a "tick" sound. Turn up slider (26) labeled D for decay in the transient generator section to midway. Note how the envelope becomes percussive, almost like a piano. The higher you set the slider, the longer the sound takes to decay. Now try the other three sliders labelled A (attack) (27) S (sustain))28) and R (release) (29). Note how they control different parts of the envelope. The shape of the envelope is that of the ADSR transient.

ADSR REPEAT

All of the audio level sliders should be down except for the VCO1 ⩘ (1) slider. The VCA patch switch (23) should be in the ADSR ⌒ position.

Set the A (27), S (28) and R (29) slider fully off and the D (26) slider to a notch less than midway. Place the ADSR repeat switch (30) in the AUTO position. Vary the LFO frequency (6). Do you hear how the envelope repeats at a speed determined by the LFO frequency (6) setting? Place the ADSR repeat switch (30) in the GATED position. Now the repeat will only occur if a key is depressed. After you're through experimenting, place the ADSR REPEAT switch (30) back in the "off" position.

USING THE PATCH CHARTS

The synthesizer output should be connected to a suitable amplification system using the jack marked "Audio Output Low" for guitar amplifiers and other high gain inputs. The output marked "High Level" is intended for line inputs on P.A. systems, external inputs on home organs, and other high level inputs. By setting the front panel controls as shown on one of the patches included in this manual, the synthesizer will be ready to play. The volume control in the V.C.A. section will determine the overall output level.

THE SYNTHESIZER SHOULD BE LEFT ON FOR AT LEAST 10 MINUTES BEFORE FINAL TUNING SO THAT THE INTERNAL CIRCUITRY CAN THERMALLY STABILIZE. Tuning prior to warm up will result in a slightly drifting tune setting that may require retuning after the initial 10 minute interval. The instrument consumes very little power so that it may be left on continuously throughout a performance without overheating.

When setting up a patch, it is always best to start with all the controls at their minimum setting and then systematically scan the panel to set each control. Turn the volume off until you're ready to hear the patch so that you won't be distracted during the setting-up procedure.

Remember, every synthesizer manufactured will be slightly different due to normal parts variations, so all of the patch settings are approximate for your synthesizer. Use the patches as a guide- your ear should be the final judge of tonal quality and slight variations may have to be made to the original settings. If a control is not marked on the chart it should be set to minimum. Pay close attention to each setting- a sound can be completely changed by forgetting to set a control.

Finally, remember that a synthesized sound should be used in context. A trumpet sound should be played as though you were playing the actual instrument. Playing a trumpet patch with the style of a guitar will detract from the realism of the trumpet effect. Stay within the pitch range of the real instrument and add the expression used by a musician if he were using the real instrument. Synthesizing the sound is not enough for it to be convincing- the correct style is just as important.

technical SPECIFICATIONS

output level

High level- 25Vpp maximum
Low level- 2.5Vpp maximum

patching

TO SLAVE

stereo cable provides control voltage and gate signals to slave unit

RING: buffered control voltage output.
Output Z= 1Kohm
Range= 0.2V- 3.2V

TIP: buffered gate output
Gate High= +12V
Gate Low= 0V

FROM MASTER

NOTE: Trigger signal is internally derived from changes in gate and CV transitions. For proper triggering, input CV must have no glide.

Accepts sampled or unsampled control voltage. Has internal S&H circuitry. Accepts gate signal.

Ring: C.V. input
Input Z>2Megohm
Input Range: 0V min
10V max

Tip: Gate input
Input Z>300Kohm
Threshold= 2V (approx)

DIGITAL KEYBOARD OUTPUT

see text for specs

PEDAL INPUTS

Ring= +15V `470 ohm
Tip = CV input, +/- 10V max
>100k input Z.

ext. audio input

10Vpp maximum input
100K input Z

adsr ar

Attack time= 3msec-7sec
Decay time= 3msec-7sec
Release time: 3msec-7sec
Sustain: 0-100%peak

LFO

Freq= 0.03hz-30hz
Sine & Square wave outputs
Delay=0-5sec (approx)

VCO kitten	Range (kbd off)= 1Hz-2KHz (kbd on) = 6Hz-17KHz Tune (kbd off)= 11 octaves (kbd on) = 2 octaves Waveforms: saw, triangle, variable pulse, suboctave square-1, suboctave square-2
VCO1 srm	Range (kbd off) = 1Hz-9KHz (kbd on) = 1hz-30KHz coarse tune = 15 octaves fine tune = 1/2 octave Waveforms: saw, triangle, variable pulse, sub octave square
VCO2 srm	Range =10 Hz- 13KHz Tune =1 1/2 octaves
noise	White noise, uncalibrated.
VCF	Cutoff range= 5Hz- 40KHz Useable resonance= 40(approx) Keyboard response= 0- >2 Volts/octave (uncalibrated)
sample& hold	Samples VCO mixer or noise source Sampling rate determined by LFO.
pitch bend	+/- 1 octave, calibrated, with center band click stop
octave shift	+/- 2 octaves, calibrated
glide	variable from 0 to approx. 1.5 sec/oct
power	100-130 VAC 50/60Hz Internally modifiable for 200-250VAC 18Watts maximum consumption
physical	Kitten- 24"x17.4"x5.5" (61.0cm x43.8cm x14cm) 18 lbs (8.2kg) SRM- 24"x19.5"x6" (61.0cm x49.5cm x15.2cm) 23lbs (10.4kg)

simple problems and solutions

1)NO SOUND AT ALL:

a)Is the V.C.A. volume turned up?
b)If the V.C.F. cutoff frequency (Fc) is down all of the way, no sound will get through.
c)Are any of the V.C.O. or noise audio sliders turned up?
d)If the ADSR sliders are all down and the ADSR is patched into the V.C.A. you'll only hear a "click" when a key is hit.
e)Is the amplification system working correctly?

2)NO SOUND FROM THE V.C.O.

a)Is the audio "on/off" switch in the "on" position?
b)Is the V.C.O. frequency in the audio range?
c)Are the audio sliders up?

3)FILTER DOESN'T TRACK KEYBOARD

a)Is the "keyboard control" knob in the V.C.F. section turned up?

4)NO SAMPLE & HOLD

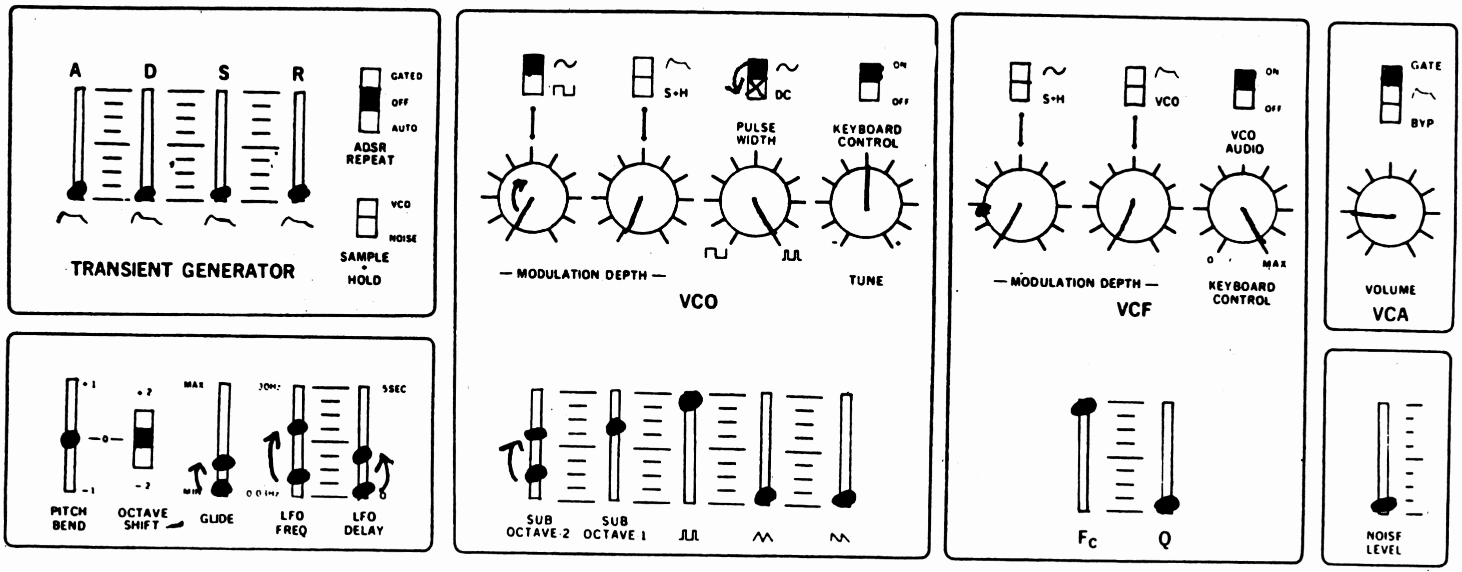
a)Is the S&H switch in the V.C.O. position with all of the V.C.O. audio sliders off? If so, the S&H won't have any input to sample.

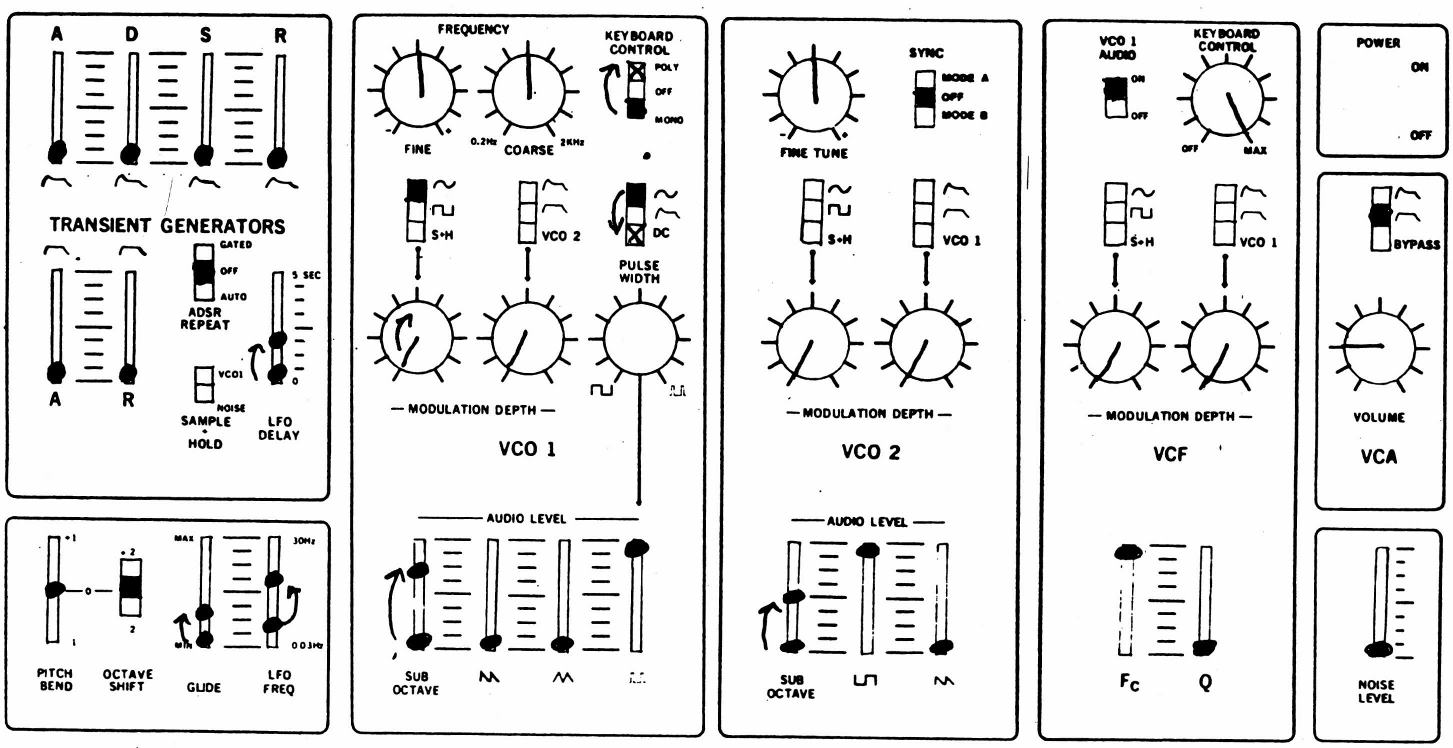
5)V.C.O. DOESN'T TRACK KEYBOARD

a)Is the "keyboard control" switch in the "off" position?

6)SOUND COMES THROUGH CONSTANTLY EVEN WITH NO KEYS DEPRESSED

a)Is the V.C.A. on "bypass" mode?
b)If the ADSR repeat switch is in the "auto" position with a short attack and long release, the transients will run into each other. This will cause a constant output if patched into the V.C.A..

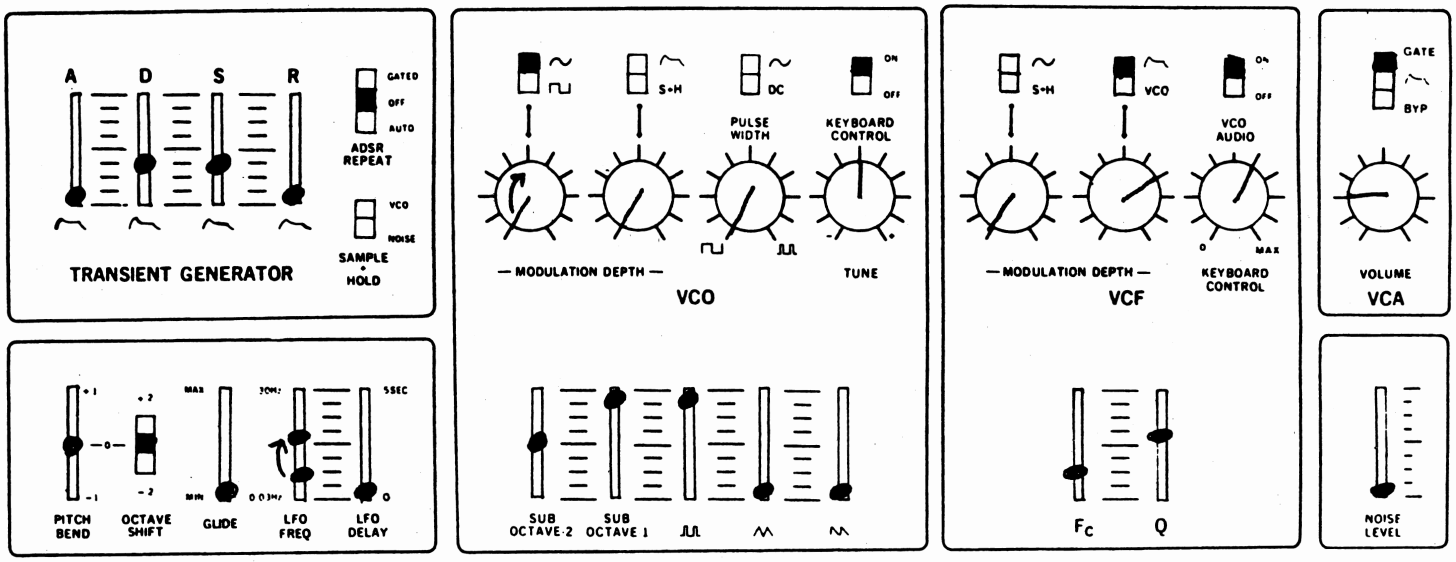


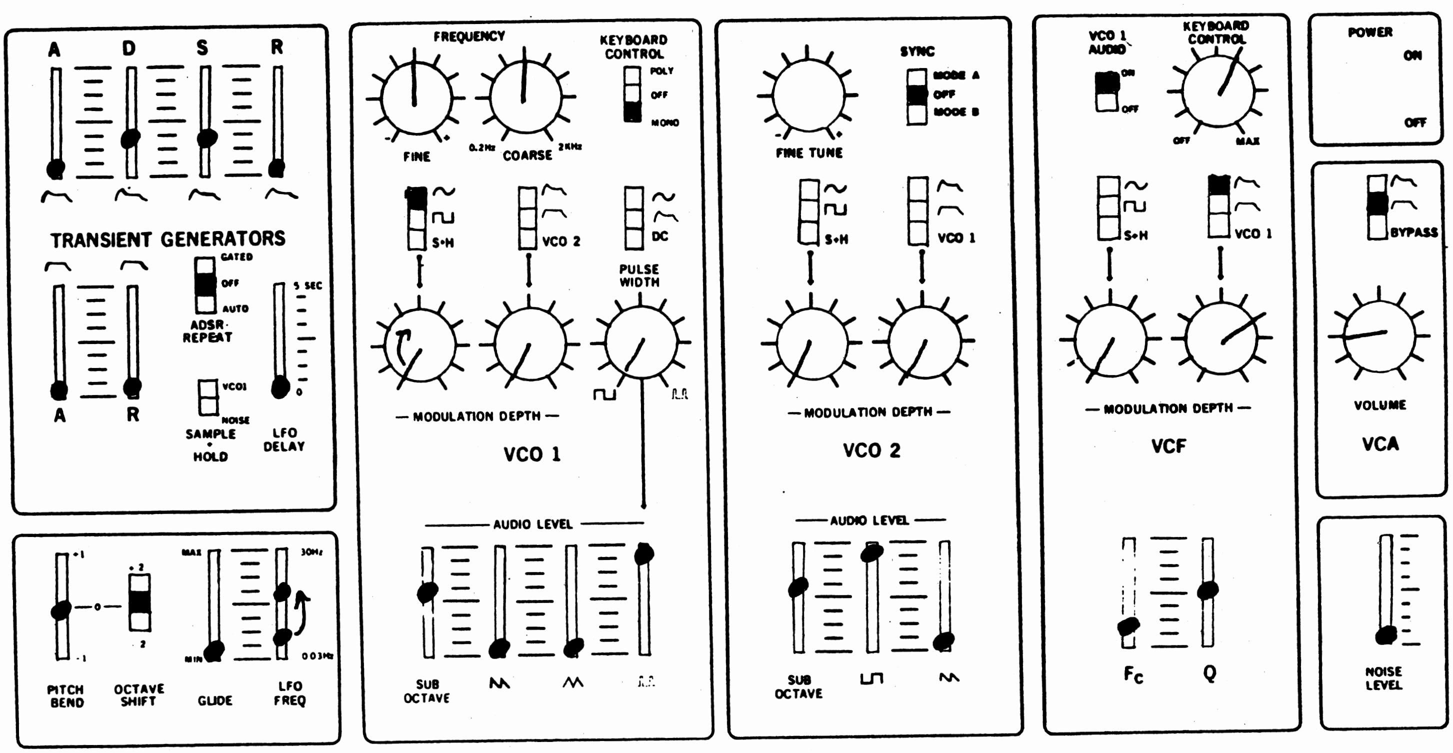


LEAD SYNTH; straight

Here is the typical lead synthesizer sound we're all used to hearing. On the kitten, the sound of chorusing oscillators is obtained by using dynamic pulse width modulation, while on the SRM, we can chorus the two oscillators to fatten up the sound. Try using the Suboctaves to see how they add fullness to the sound. A little glide here and there helps to add dynamics to passages- and don't forget the pitch bend!

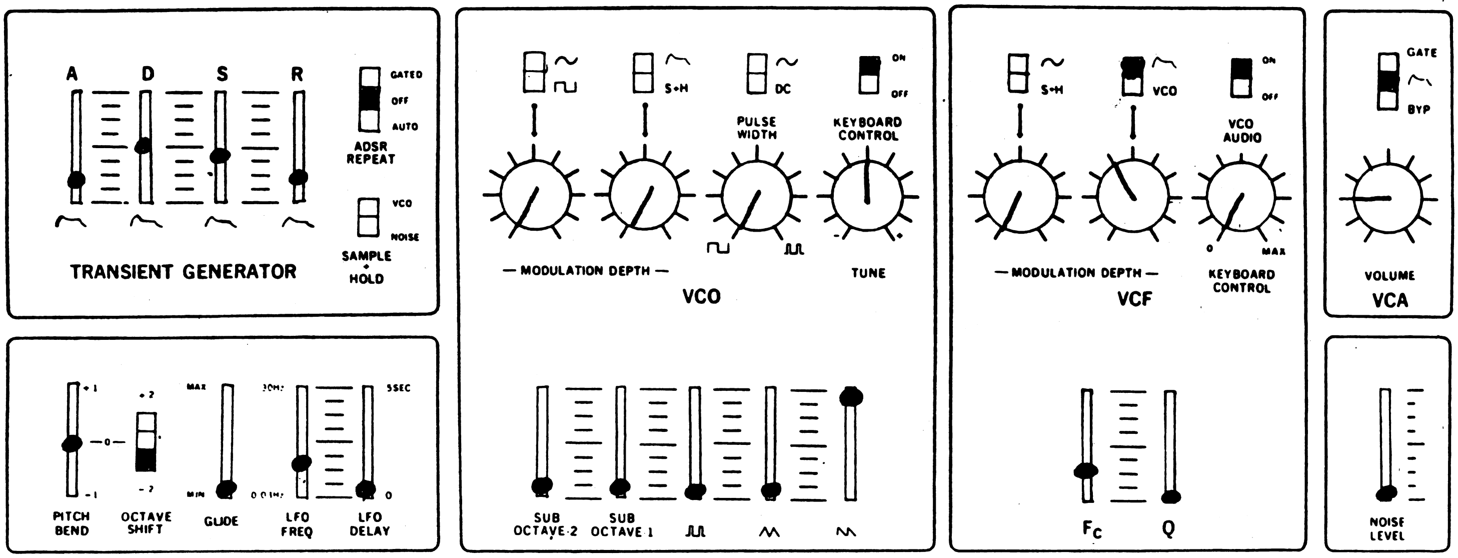
Manual vibrato can be added with the mod knob in VCO 1 or you can automatically add vibrato by leaving the knob on and using the LFO delay.

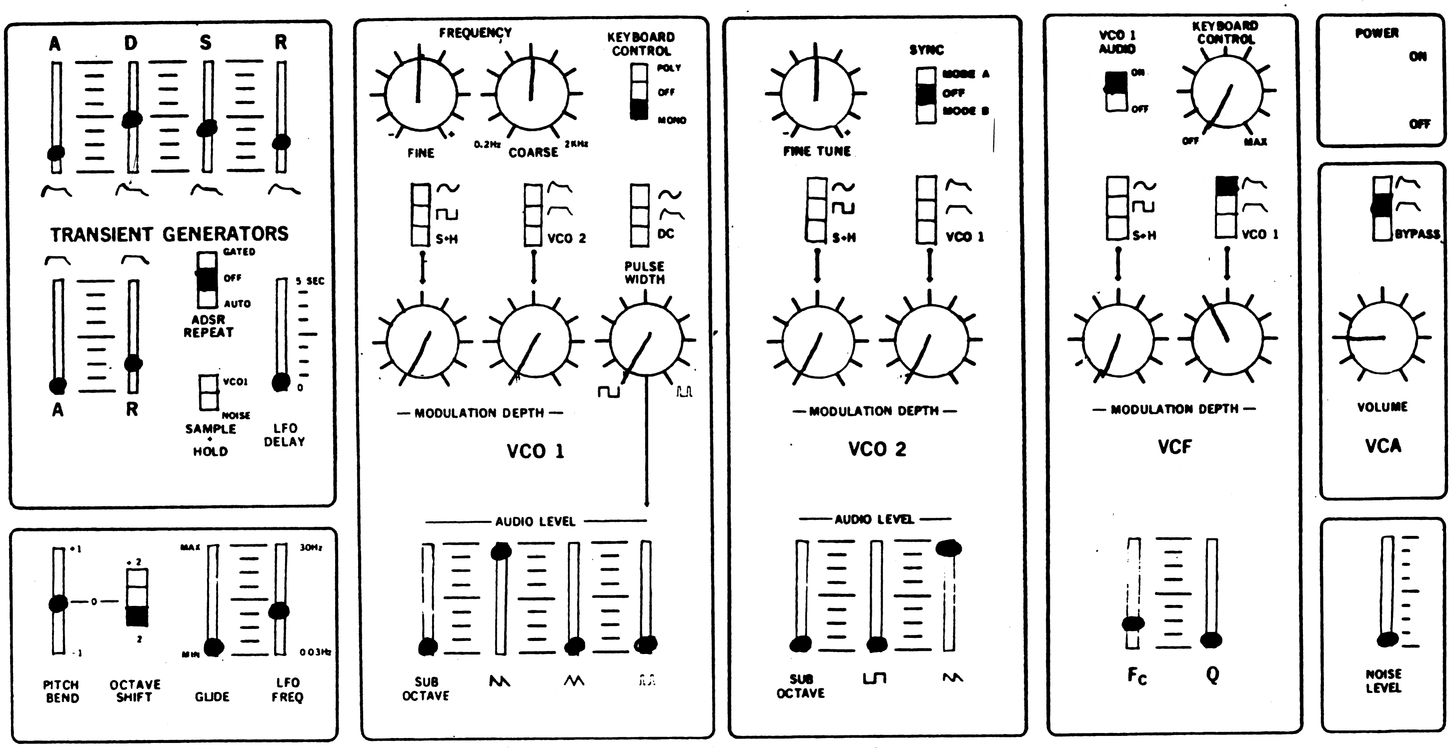




LEAD SYNTH; filtered

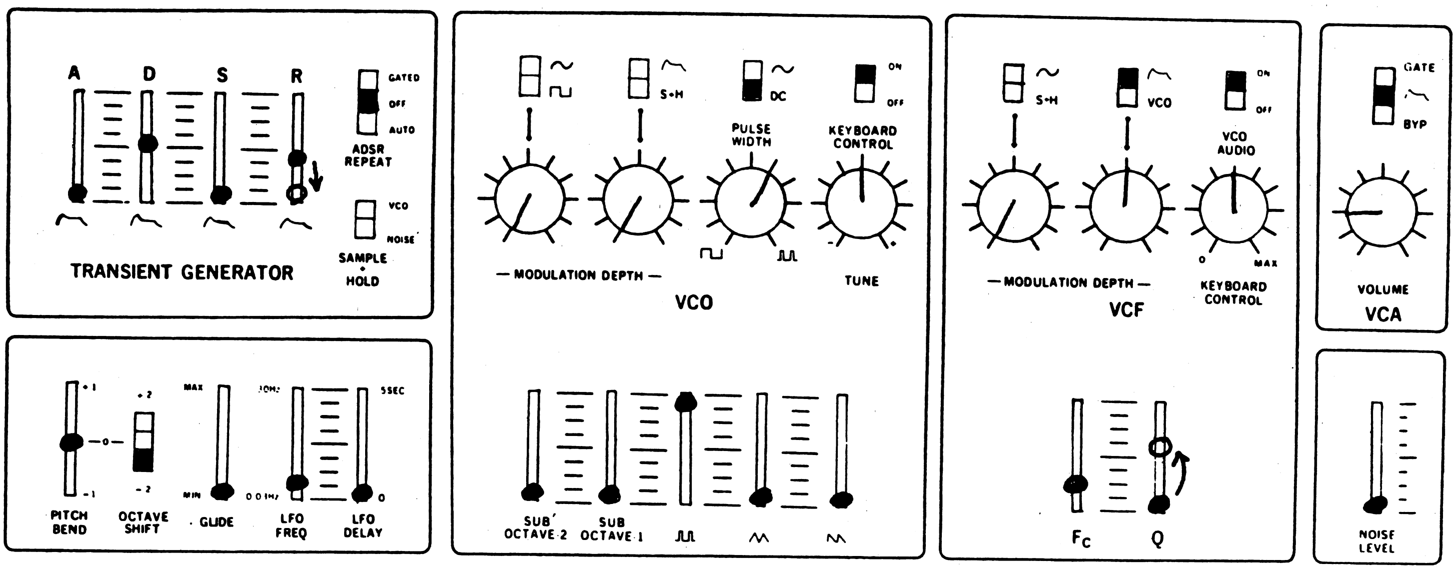
This is another typical lead synth sound except it has that characteristic "waow waow waow" tone. Follow the same instructions as in the previous patch. Try it in the -2 octave range for some fat bass sounds.

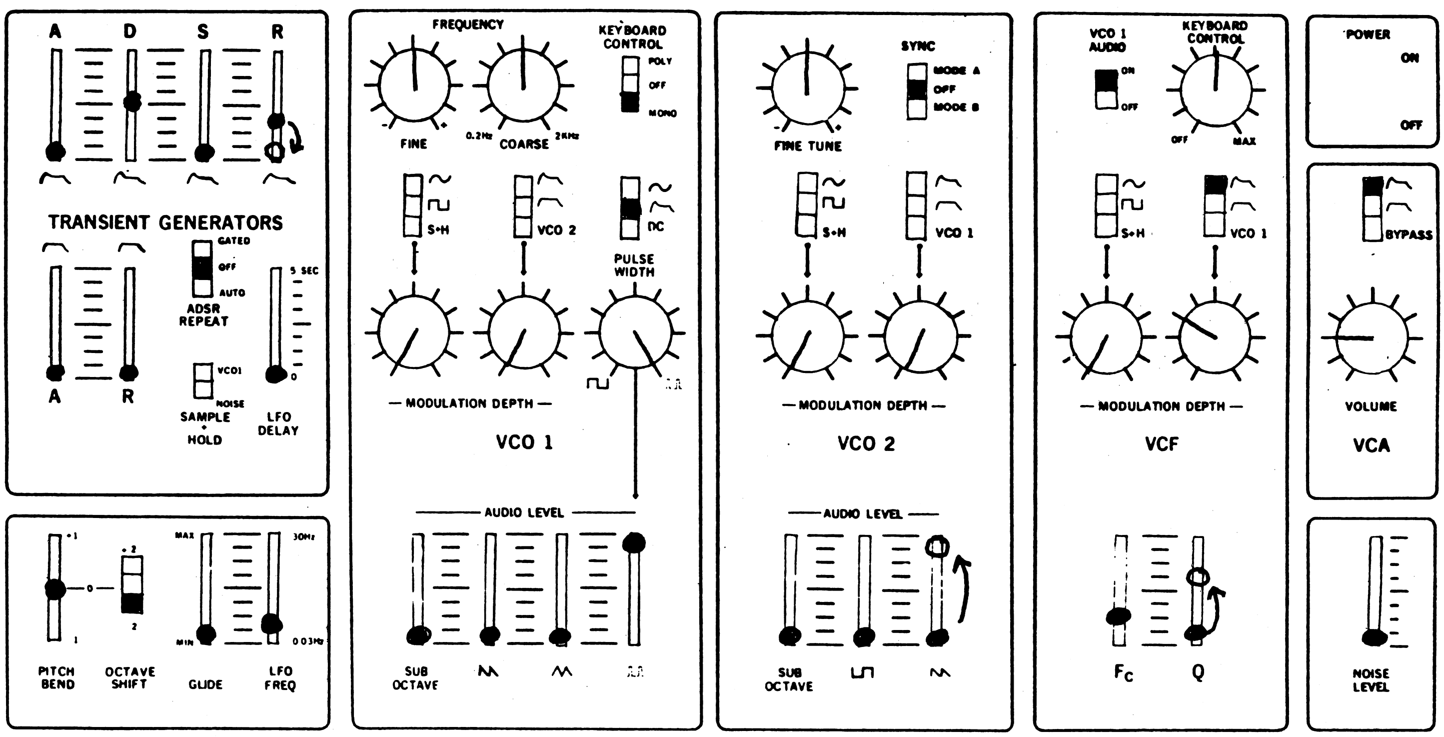




TUBA

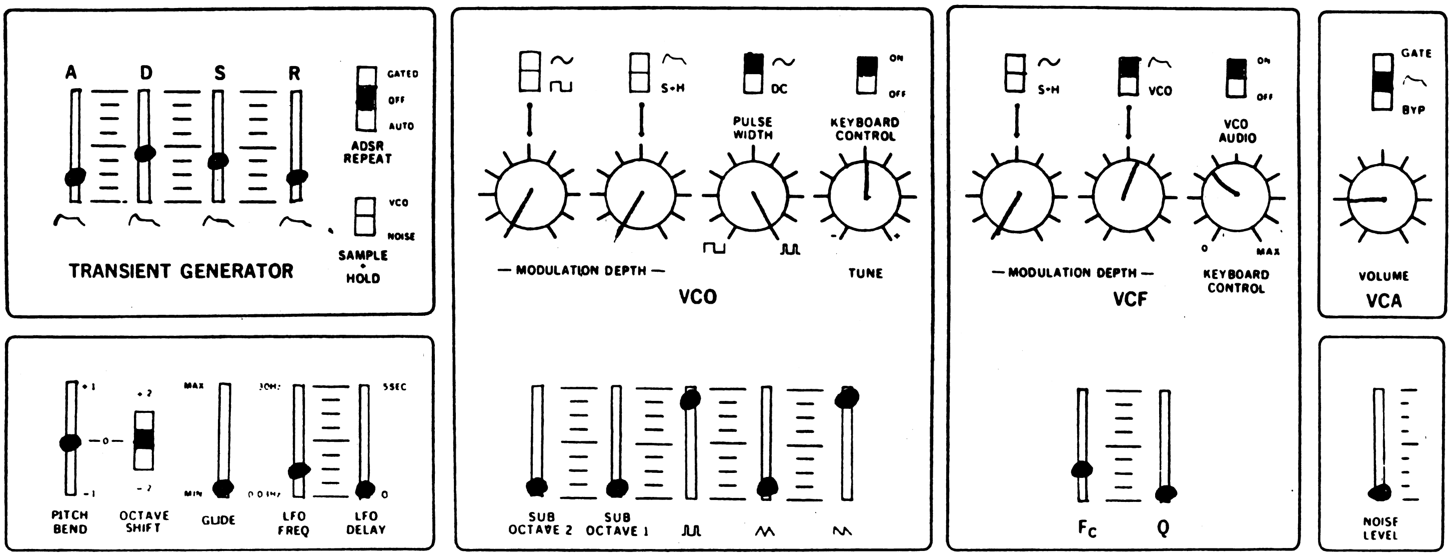
Here is a tuba sound you can use for polkas. The brassiness of the sound can be adjusted with the VCF modulation depth control. Add some bottom with the suboctave. Try it in the higher range to get a trumpet type sound.

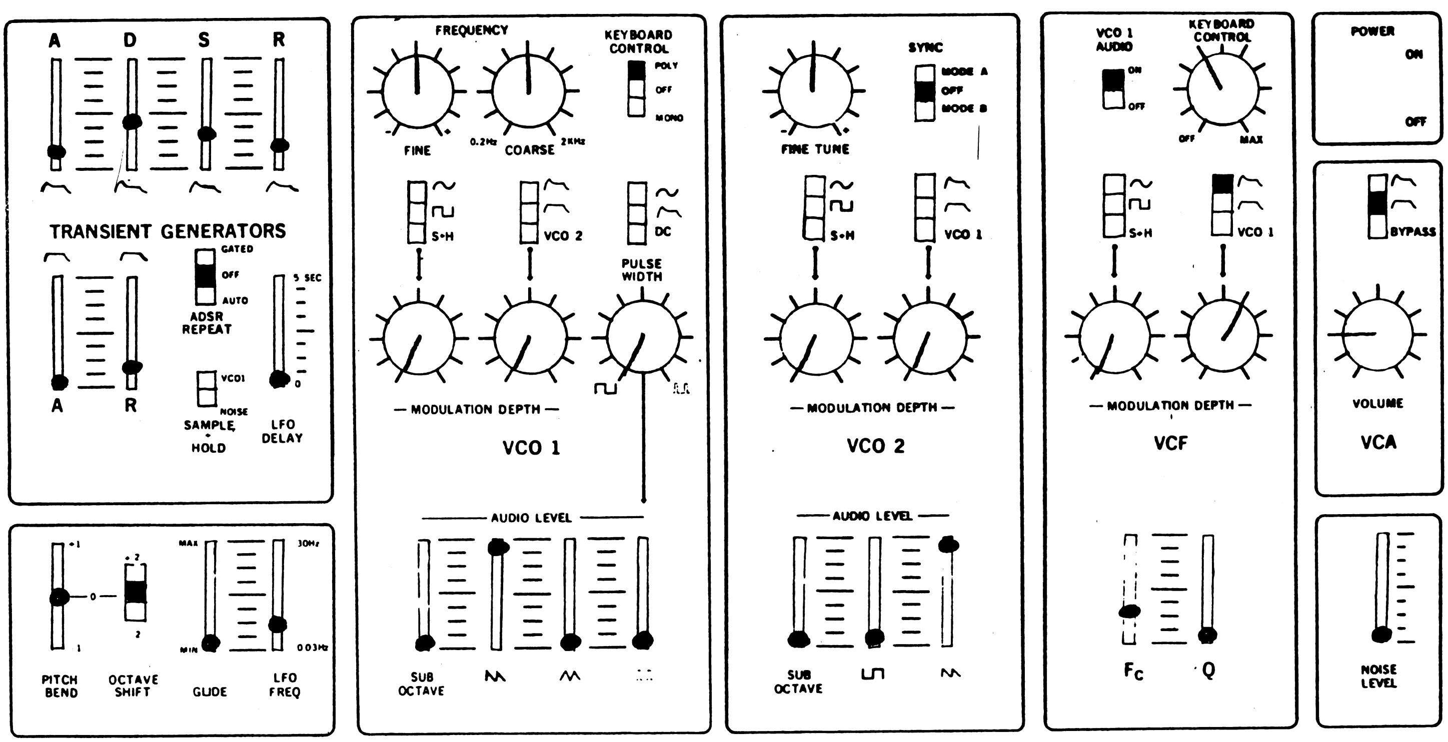




BASS GUITAR

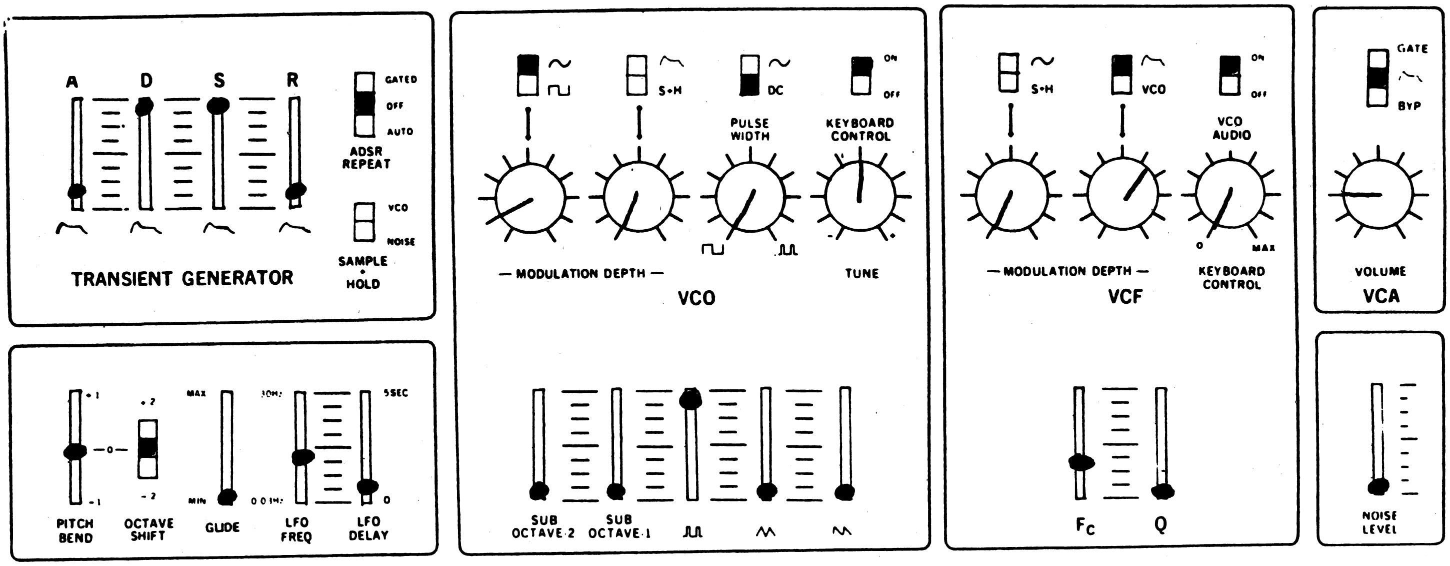
Adjust the tone with the pulse width control to get a twangier sound. Try the SRM pulse width in the DC position. Turn up the Q to get a funky bass sound. The R slider lets the note sustain after you release the key. Adjust the punchiness with the D slider.

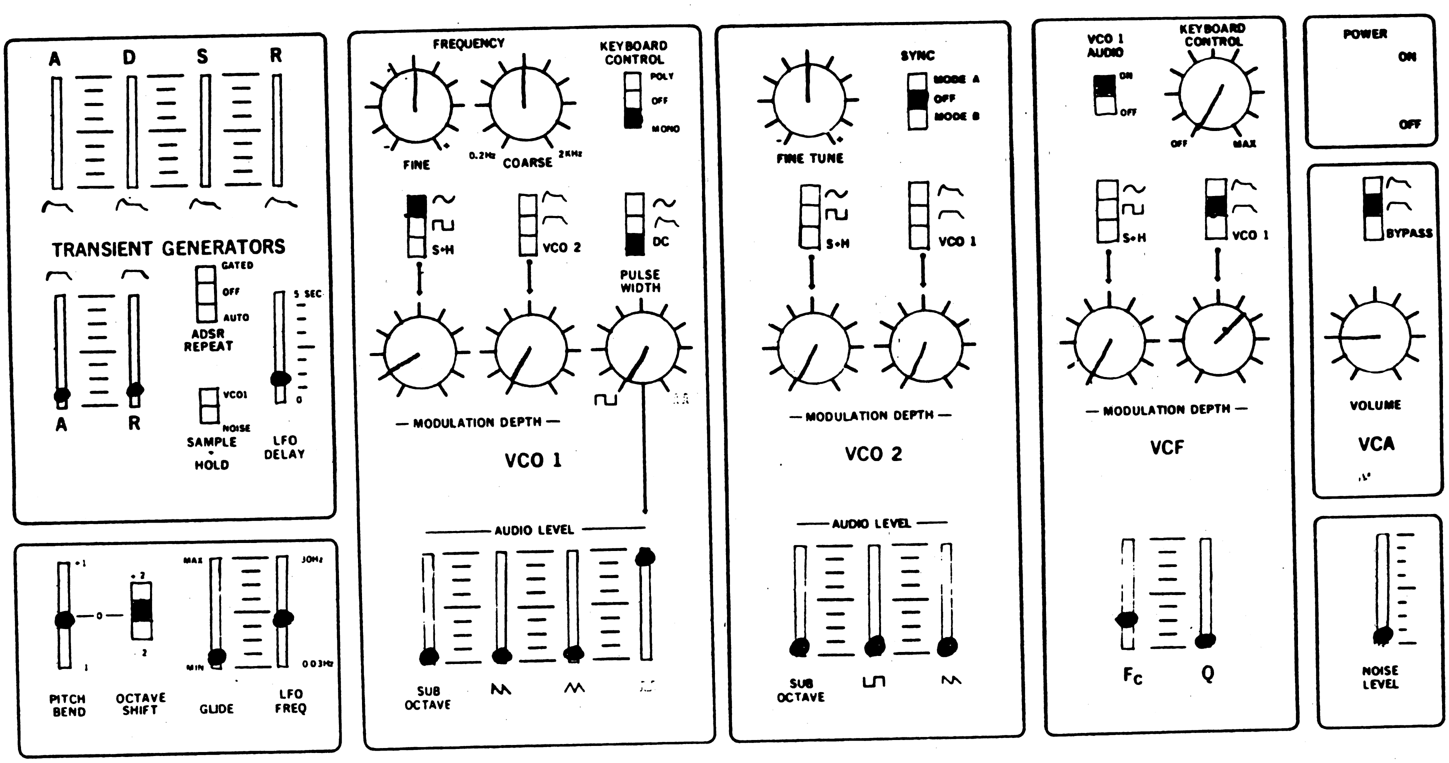




BRASS

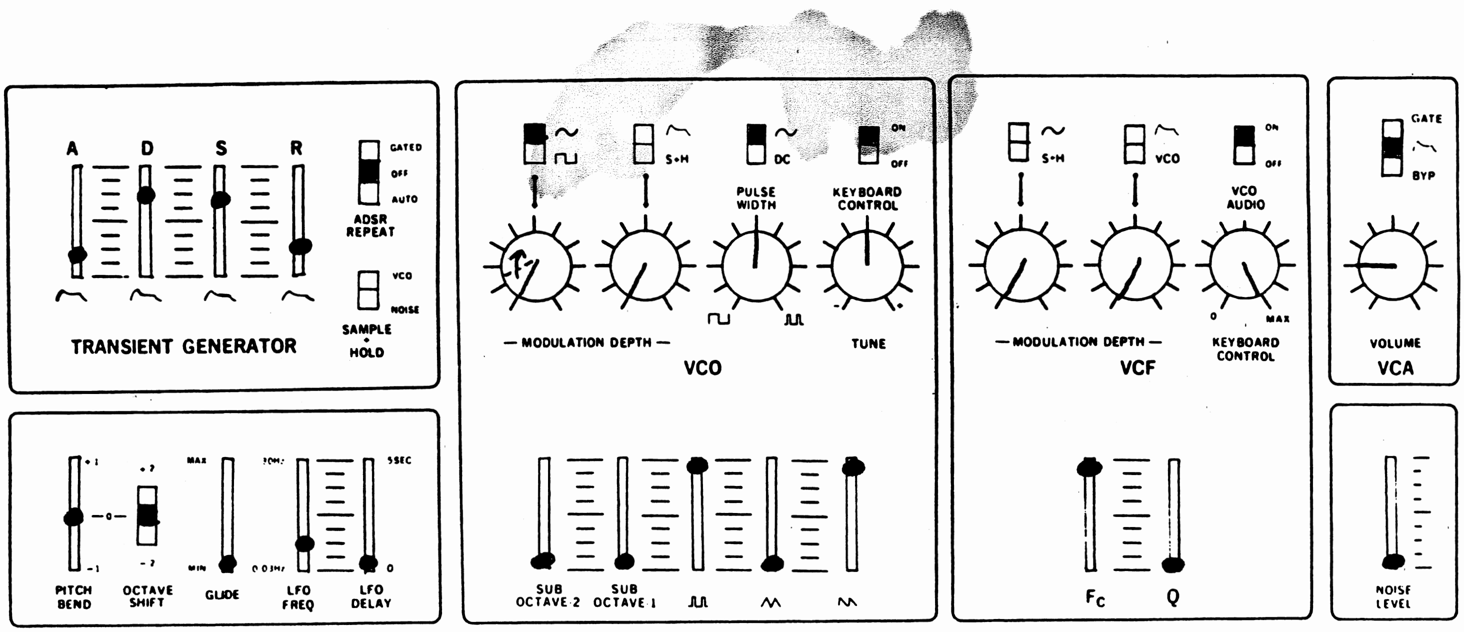
Here is a brass patch you can fool around with. The critical settings are the VCF modulation depth, Fc and the ADSR sliders. Make some adjustments so it suits your taste.

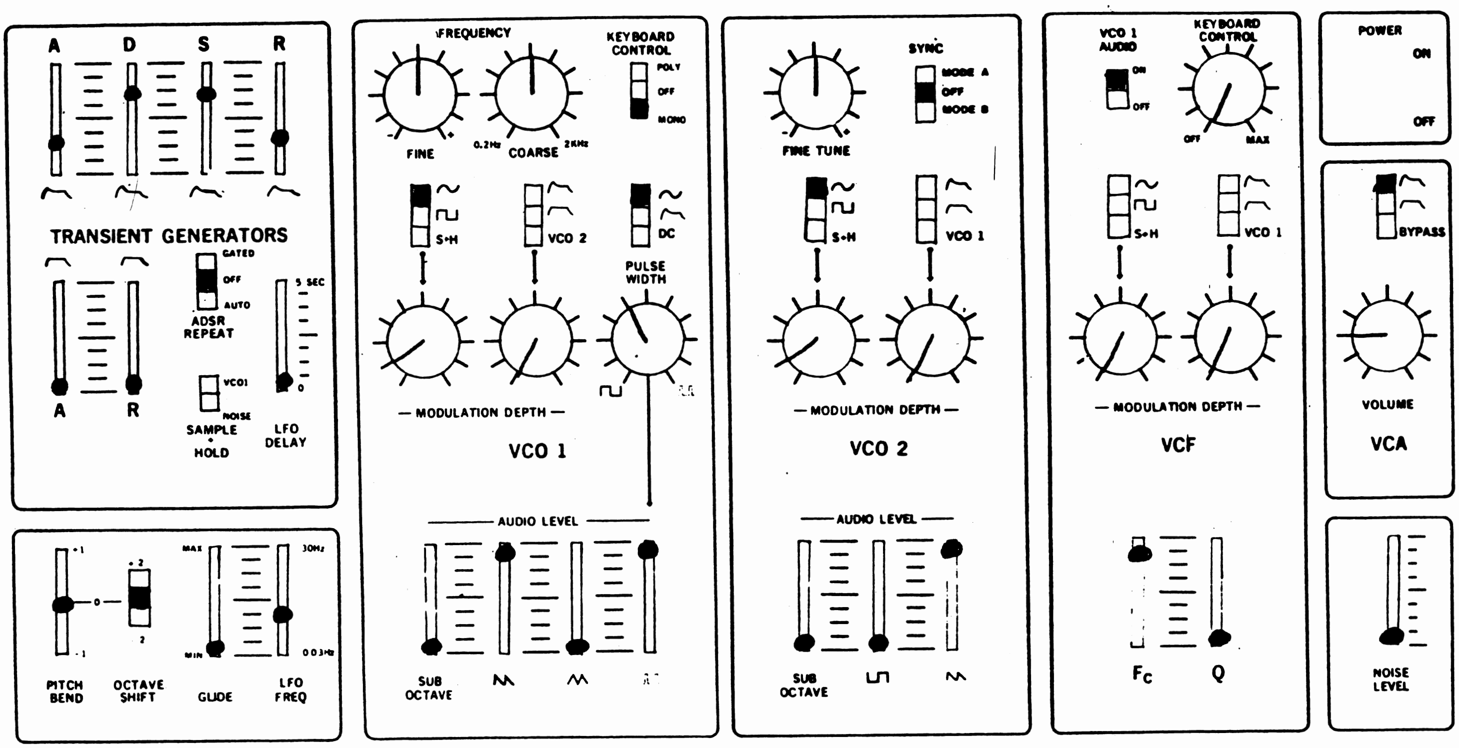




CLARINET

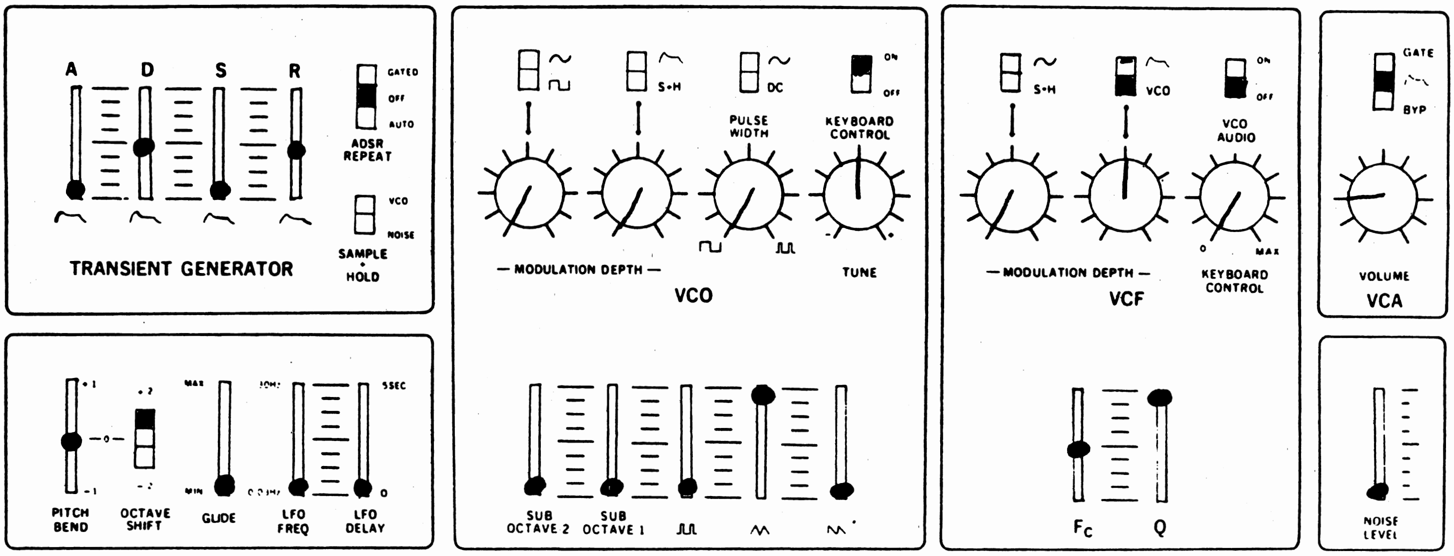
Clarinets are easy to get because a reed instrument puts out a nice hollow, square wave sound. This patch is pretty straight forward and you shouldn't need much adjusting.

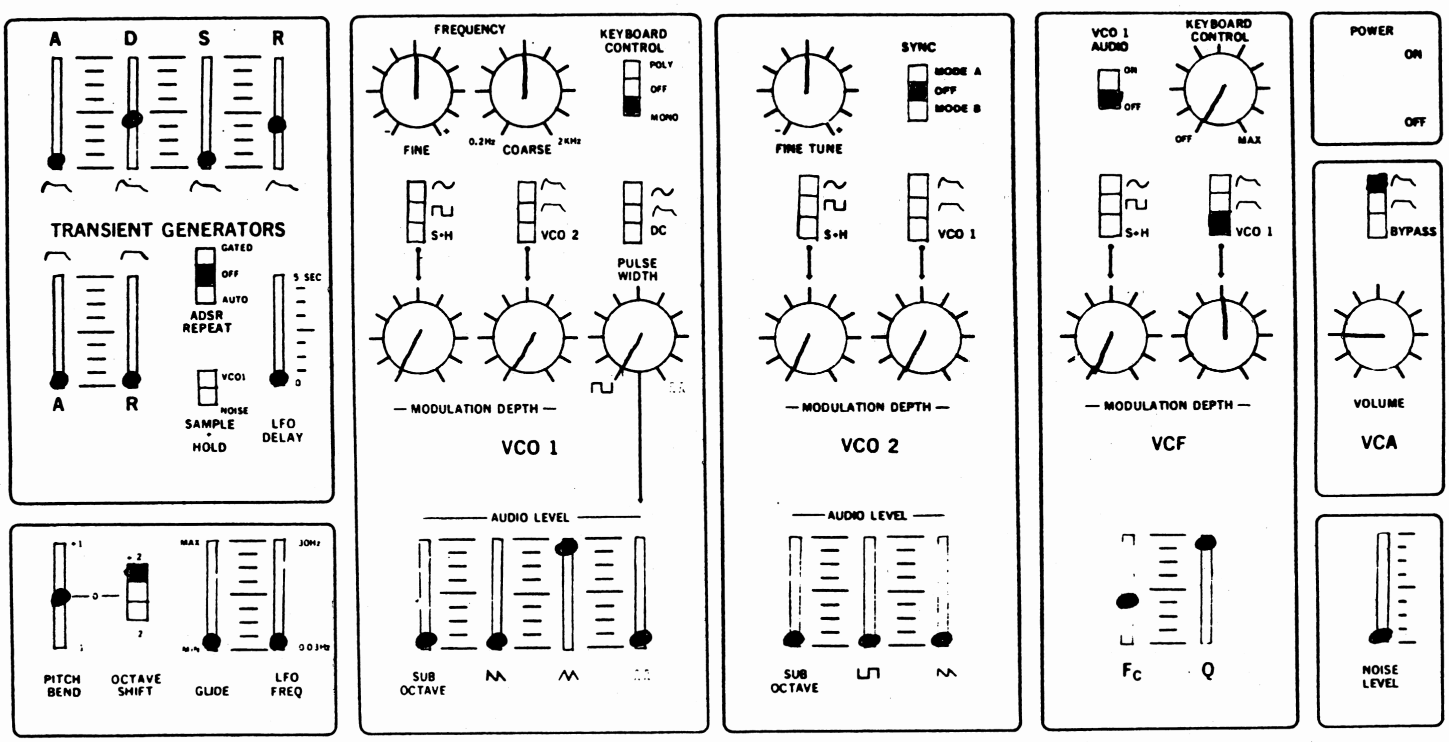




STRINGS

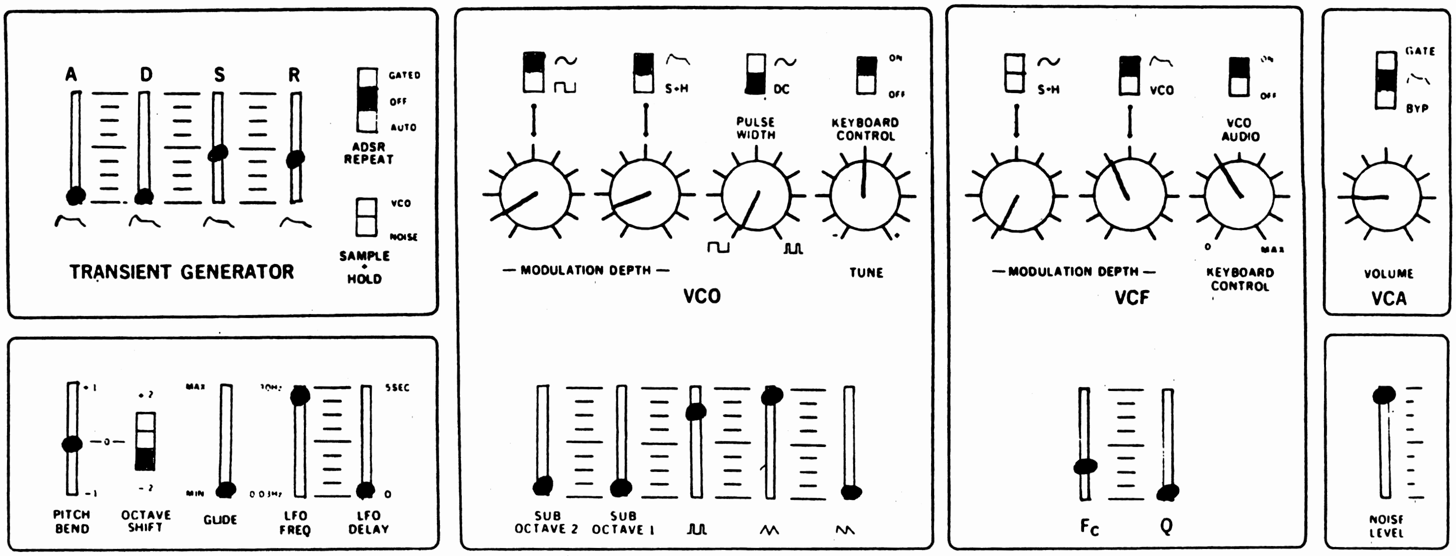
Strings are really hard to get exactly- they need a lot of playing around with. This patch sounds much better if you play it through some reverberation, analog delay or a chorus generator. The VCO modulation depths are very critical and on the SRM you should slightly chorus the oscillators. Play it slow and sweet like strings

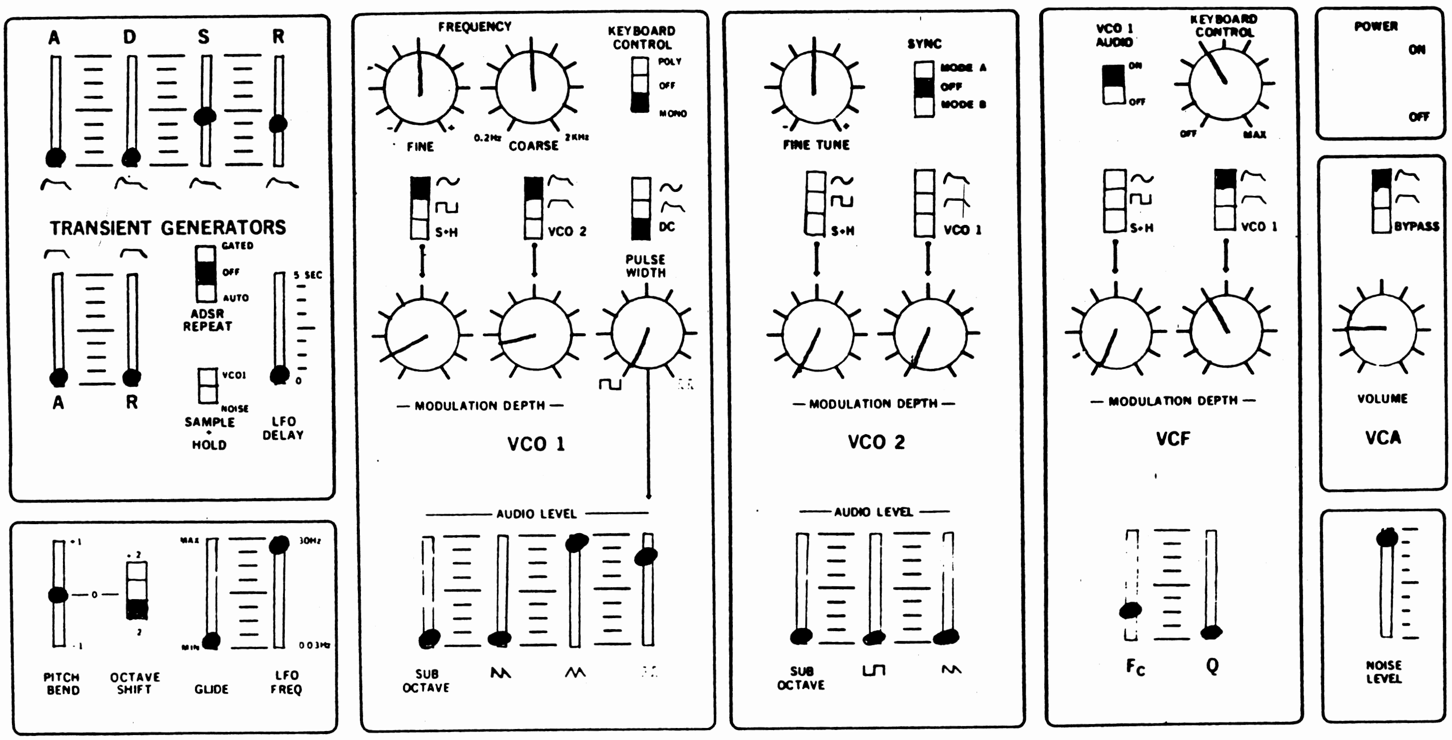




METALLIC BELLS; ring modulation effects

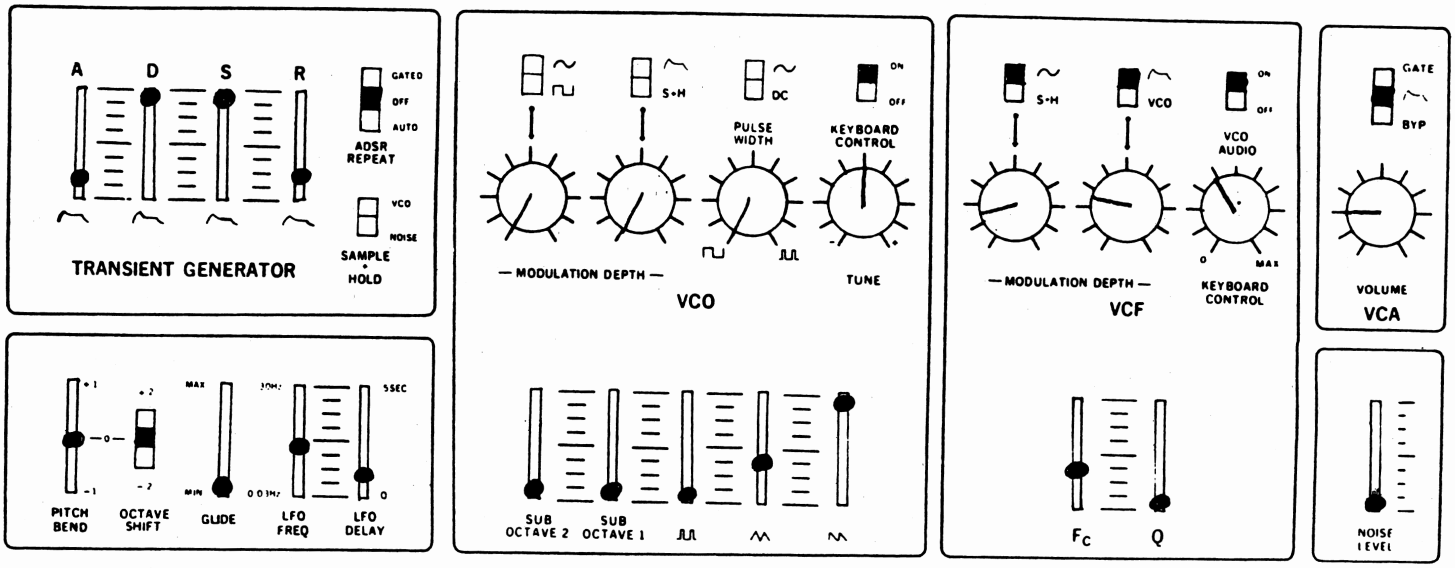
A ring modulator gives a complex frequency spectrum and this is a pretty close simulation of it. What's happening is that the VCF is being used as an oscillator whose pitch is varying very fast- at the rate of the VCO1 frequency, in fact. The result is a complex bell-like tone. Try different Fc settings and different VCF mod depths to see their effect.

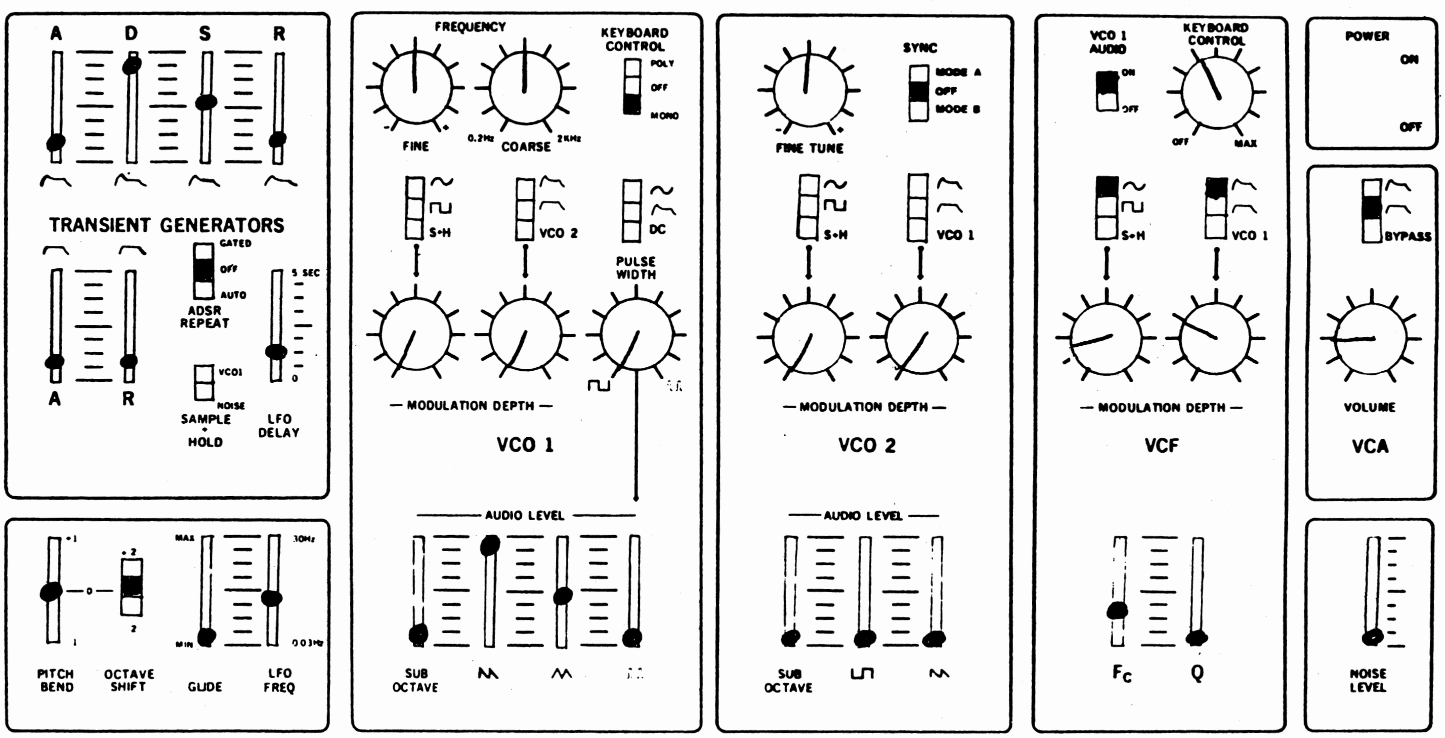




TUNED DRUM

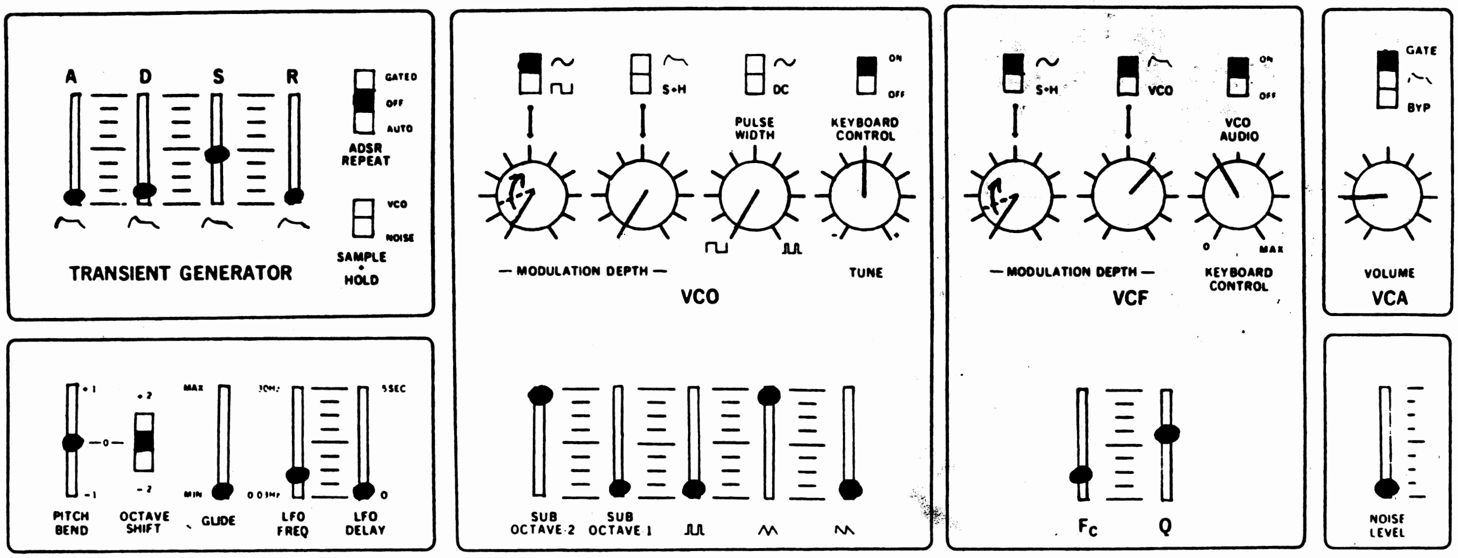
This is a percussion patch so the keyboard should be played very percussively (like drums) and don't sustain any keys.

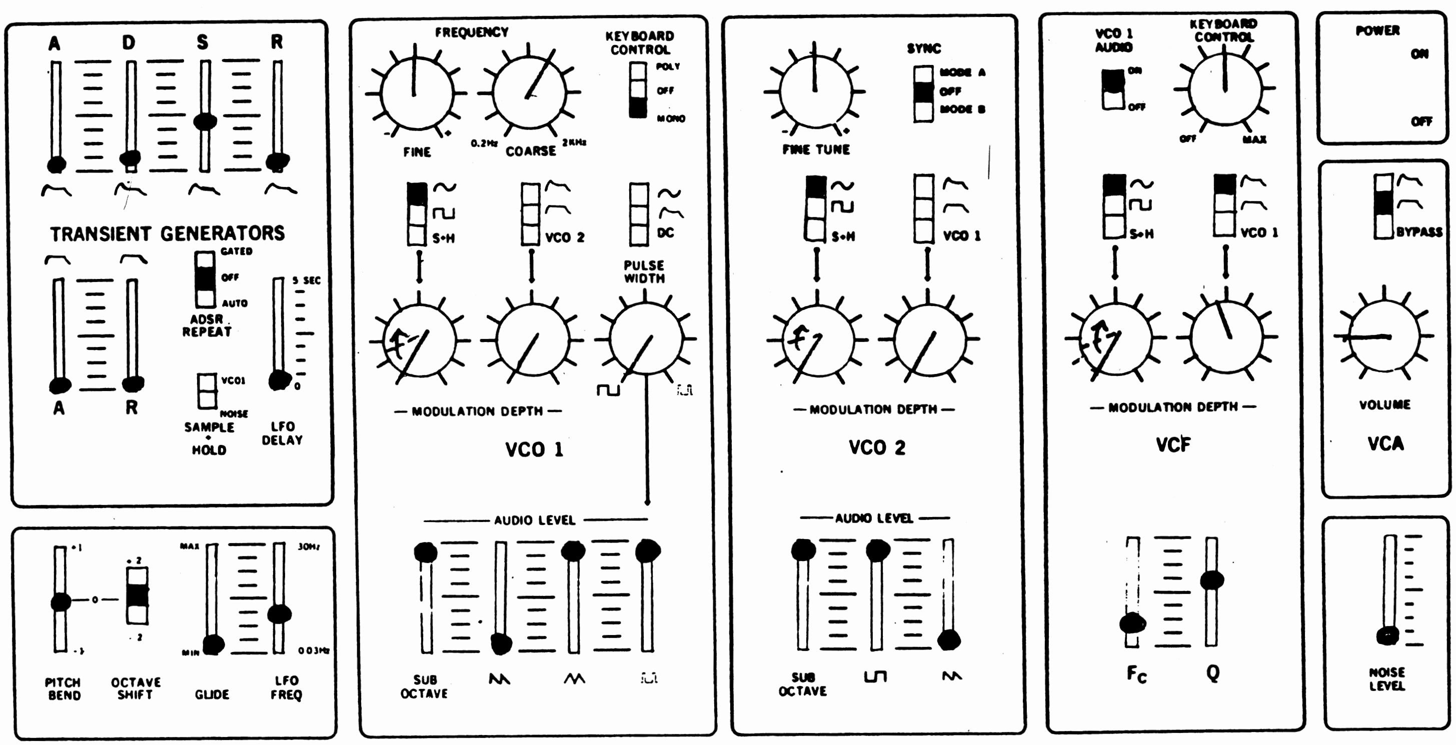




FLUTE

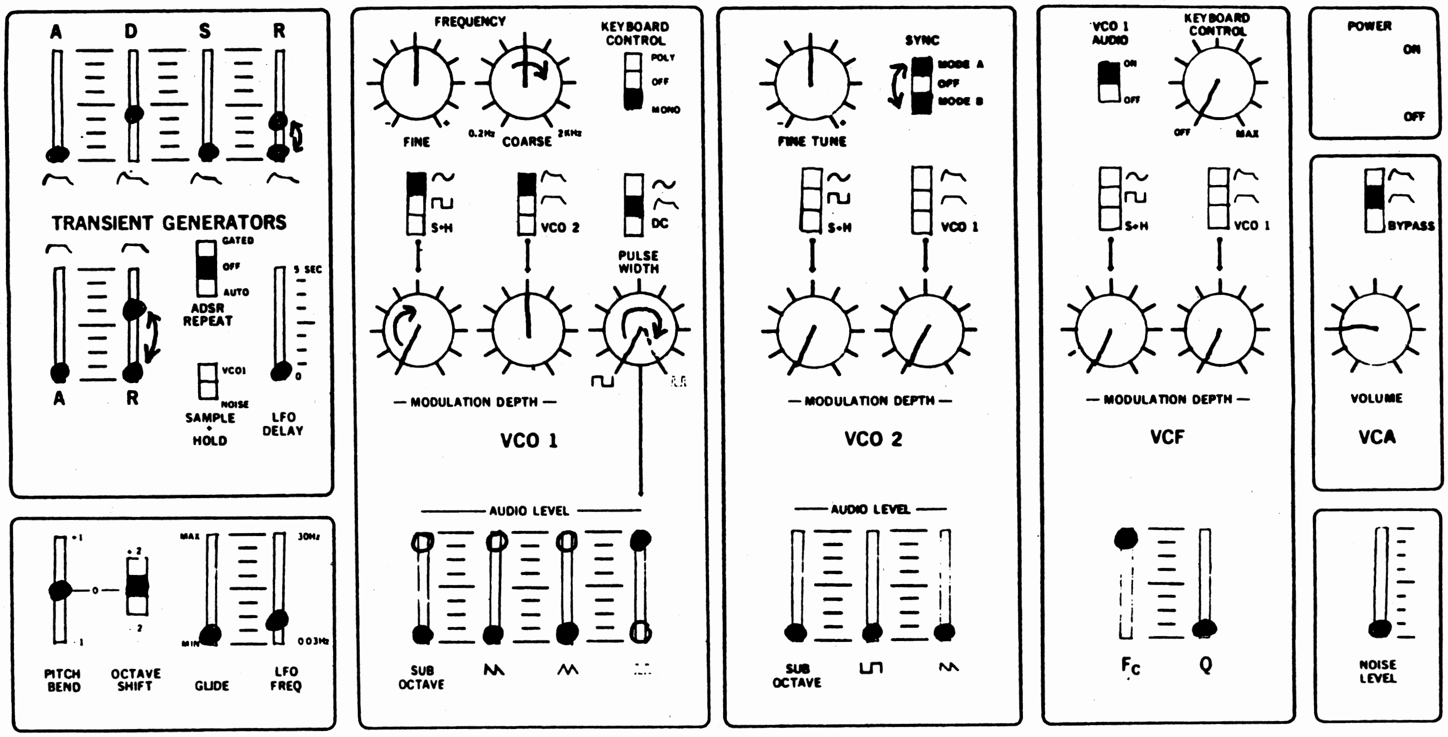
This is a straight flute patch. The mellowness of the flute is set by the VCF ADSR mod depth and the tremolo is set by the VCF LFO mod depth.

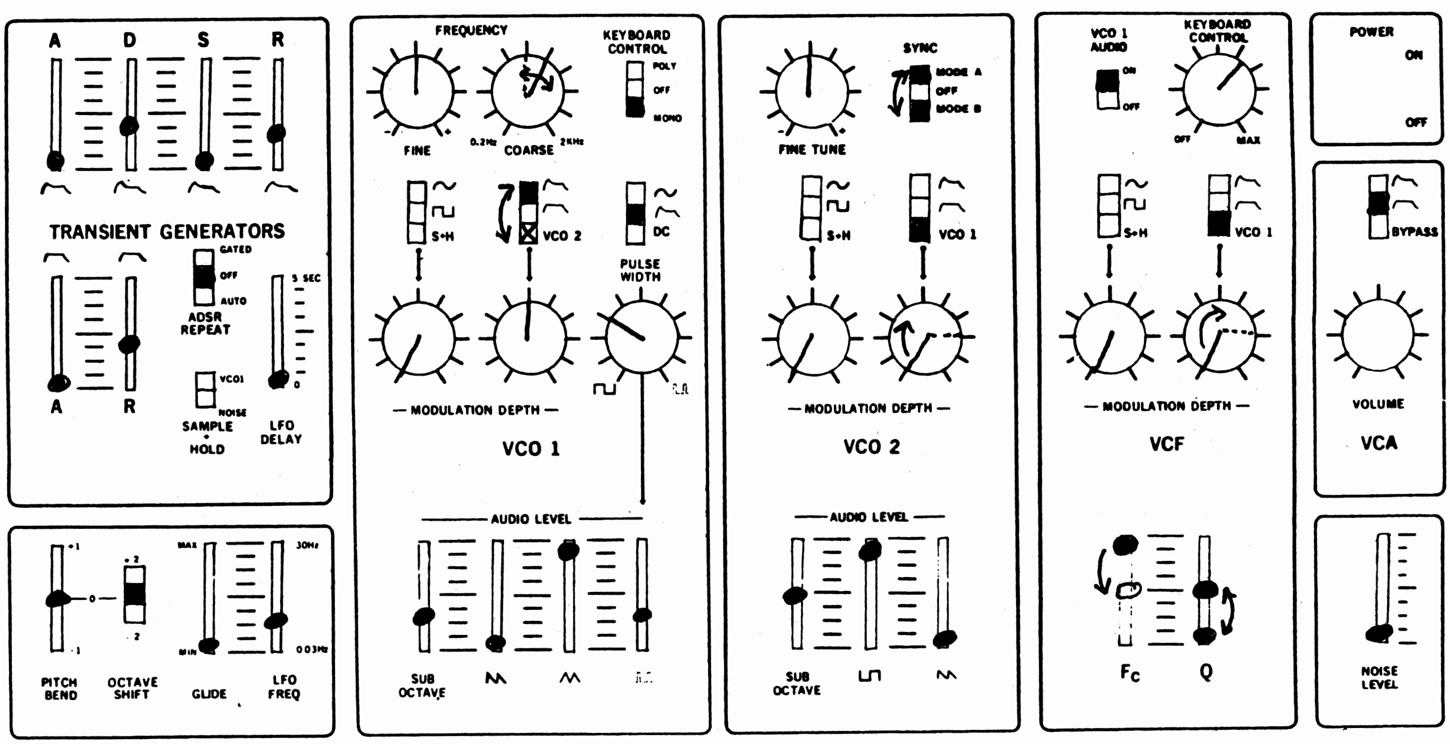




JAZZ ORGAN

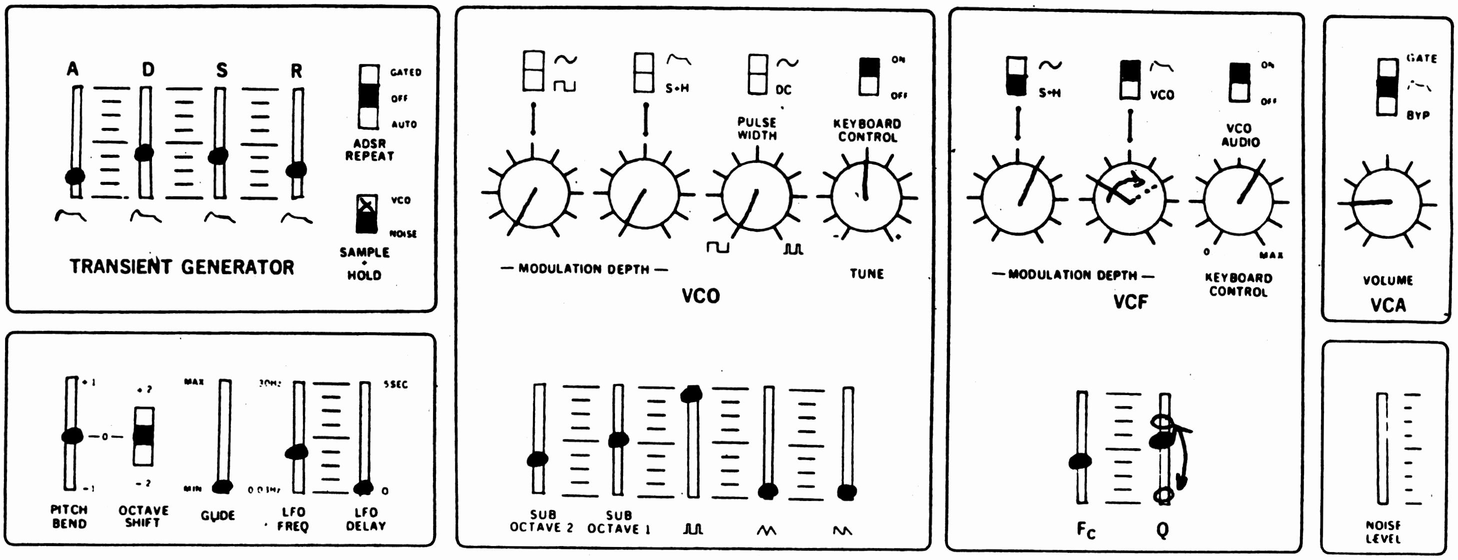
This one's not easy. in the kitten, you have to tune the VCF Fc so that it is an octave and a fifth above the VCO pitch. This makes it sound like another pitch is there. In the SRM You can try the same thing plus tune the VCO's a fifth apart. Use the suboctaves to add some bottom. with a little effort you can get this to sound like the real thing.

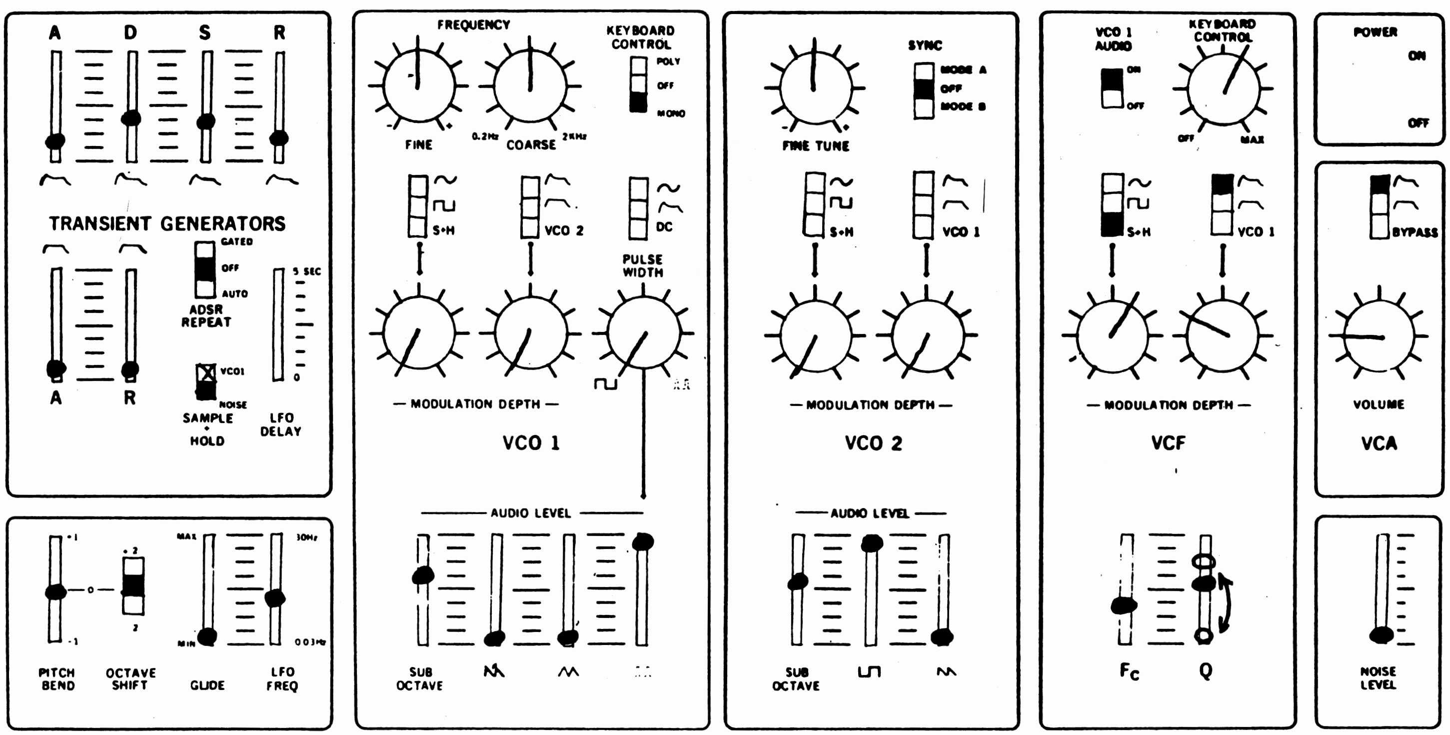




SRM SYNC

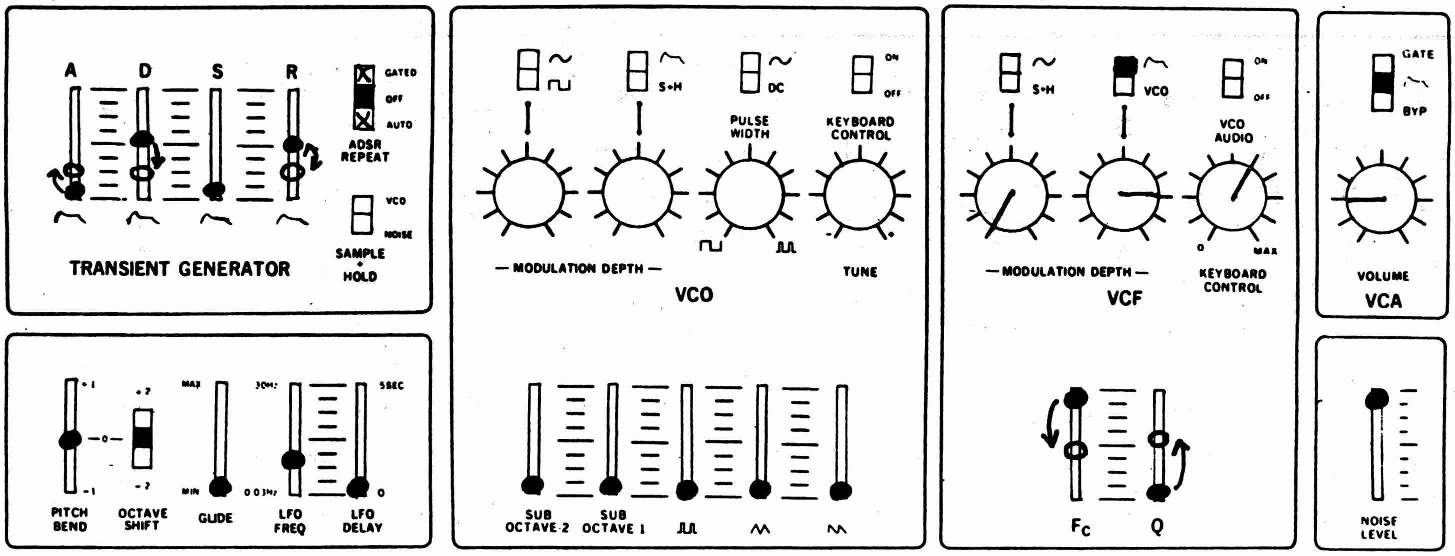
This is a set up for some sync effects on the SRM. Try both set ups and fool around with anything you want, just to see what the effect is. The principle behind this patch is that the VCO1 sync mode gives you complex filtered effects without using the VCF. So by changing the VCO1 frequency in different ways you can come up with a bunch of different sounds. Remember that each waveform on the VCO mixer has a different sync effect, so try them individually and in different proportions.

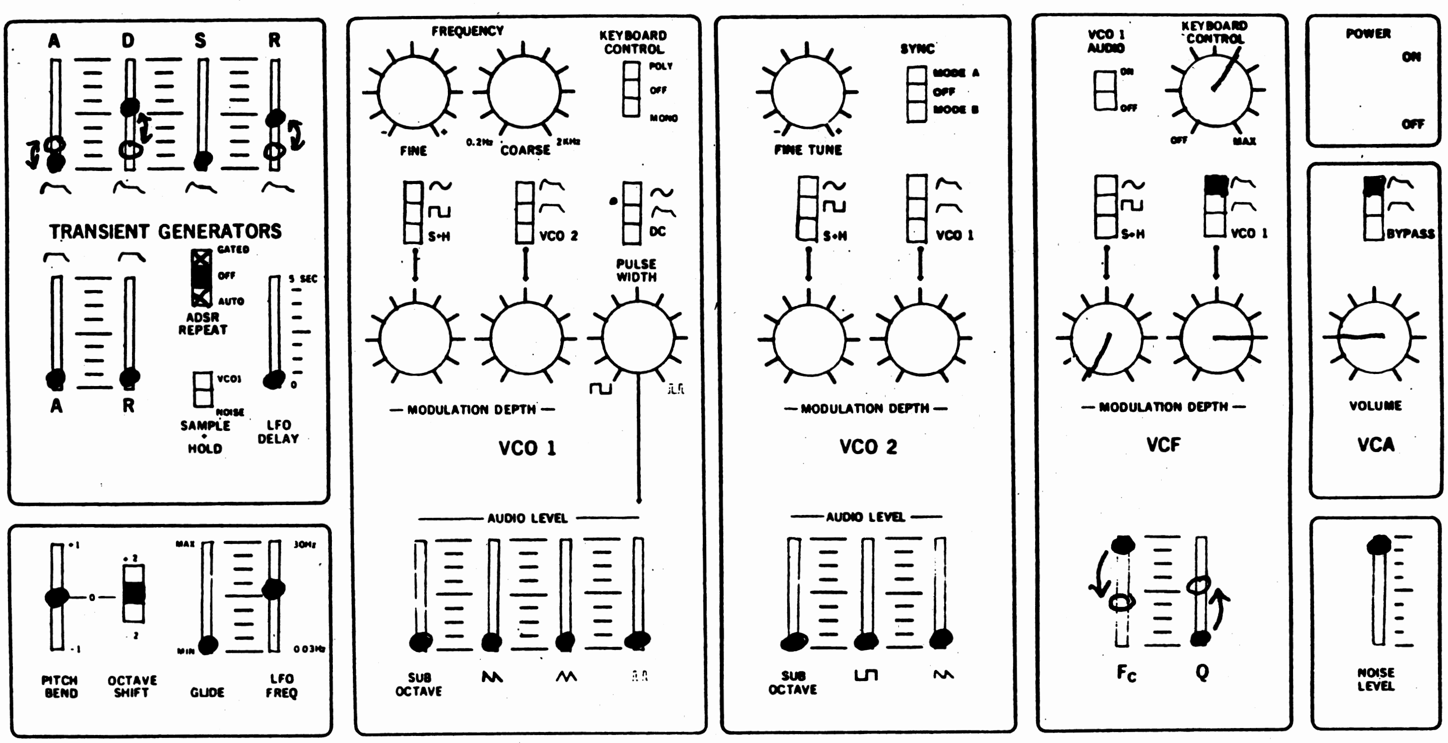




S&H FILTER

Here we're using the S&H to vary the VCF Fc to get a sampled filter effect. Try both the VCO and Noise position on the S&H input switch. Vary the Q to see what it does in this patch- but don't bring it up too much or the VCF will oscillate.

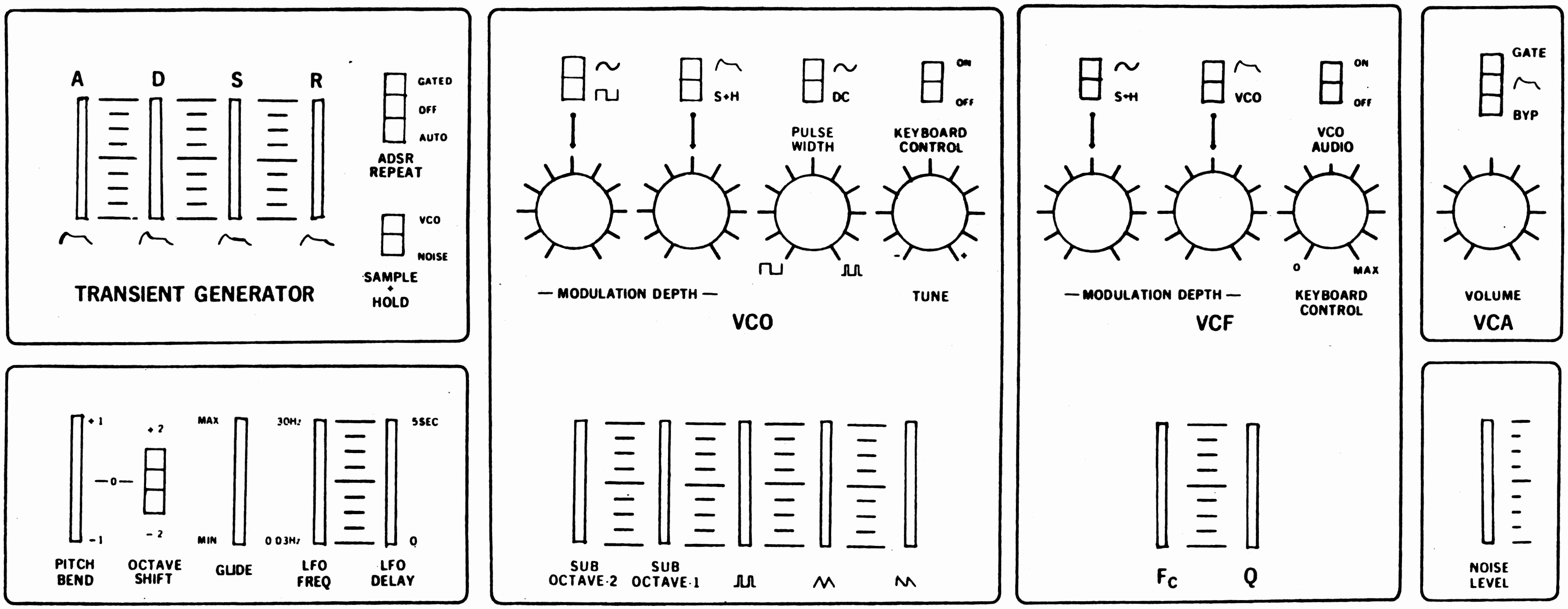




NOISE EFFECTS

Here is a basic noise patch that will let you get bombs, wind, surf, etc. just by changing the ADSR setting and filter Fc setting. Try it and see what you can get. Also try the ADSR repeat mode to get helicopters and machine gun sounds.

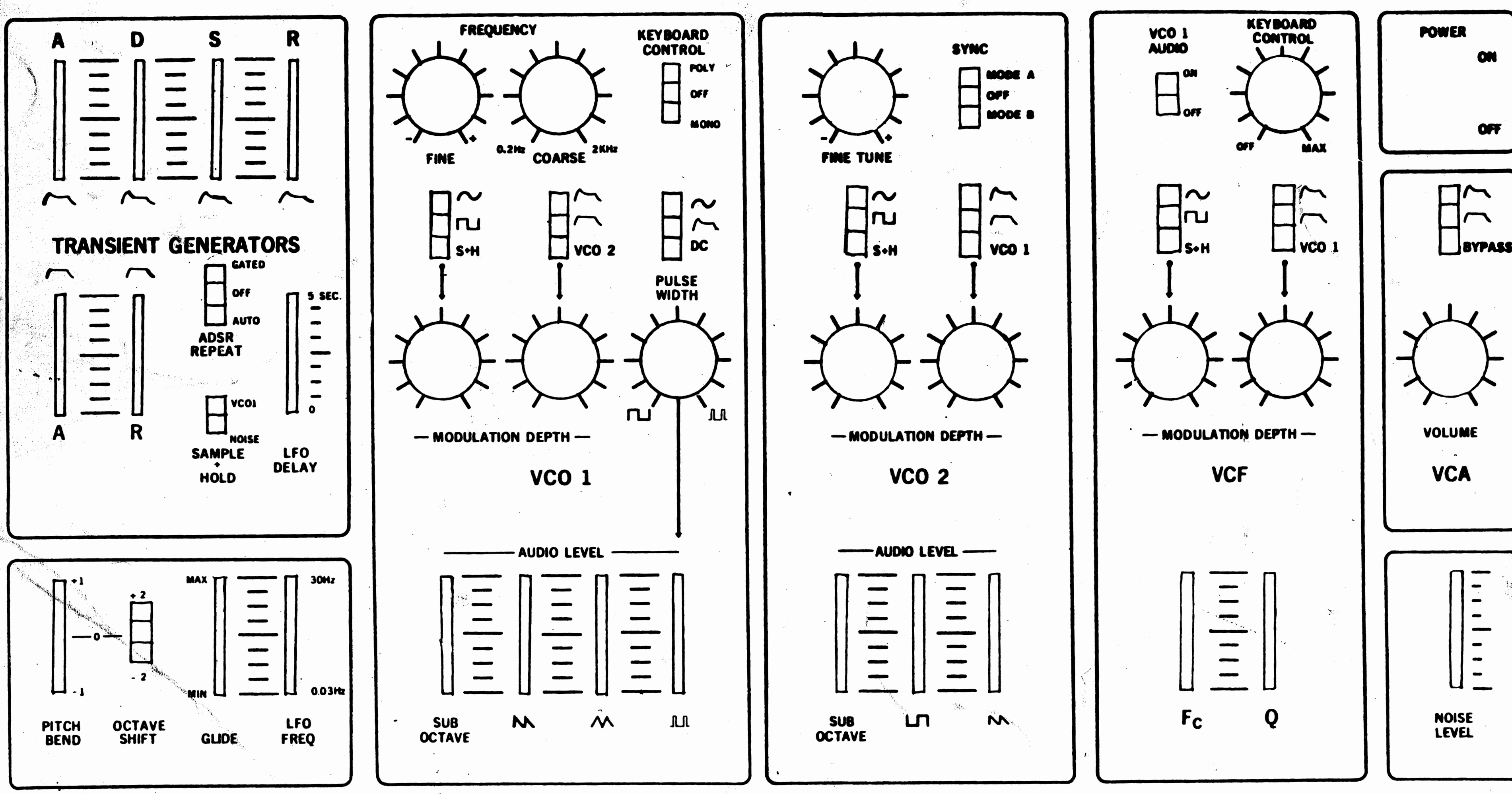
kitten II PATCH CHART



SOUND______________________________

CAT SRM II

PATCH CHART



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