AN INTRODUCTION TO THE
SERGE MODULAR MUSIC SYSTEM

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ABOUT THIS MANUAL

This manual is an instruction and users manual for the Serge System Modular Synthesizer. It is intended to serve two purposes:

1. A Self-tutorial instruction manual for those who have never used a synthesizer before and for those who have used a synthesizer, but not a Serge, before.

2. A User's reference manual and guide that provides information on how to use each module, how to interconnect modules to produce certain effects and how to interconnect the Serge with other devices.

This manual is arranged into the following sections:

1. The Introduction.
2. Self-Teaching Patch Number One--Getting a Sound
3. The Theory of Electronic Music
4. Self-Teaching Patch Number Two--The Theory of Sound
5. The Serge System Modules
6. Appendices

If you have never used a synthesizer before it is strongly suggested that you read sections 1 and 2, working out the patches as you go. If you are confused at this point, don't worry, keep going. These exercises will give you the basis for understanding the explanations in subsequent chapters.

If you have used a synthesizer, but not a Serge, the patch worked through in section 4 provides a good working knowledge of the basic modules of the Serge.

If you are already familiar with the Serge system, specific information on the different modules and how to patch them together can be found in sections 4,5 and 6.

For information on interfacing to other equipment, refer to the Appendices.

AN INTRODUCTION TO THE
SERGE MODULAR MUSIC SYSTEM

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INTRODUCTION

A SYNTHESIZER is a musical instrument, which means that it is a tool for making music. With any tool there are two things to learn: what it is that is being made AND how to use the tool to help in its making. In this case you have to learn about the structure and the nature of the music you are making, whether it be electronic, rock, jazz, classical or what-have-you, AND how to use a synthesizer. However, these two things, the music and the instrument, are highly inter-related. Take the piano for instance, which would not have been developed unless there already existed a certain kind of music (chordal, many voiced). And yet, once developed, the piano changed the kind of music written and played. It is unlikely that Rags would have been developed if there had been no piano.

The same is true for the synthesizer. It was developed originally in the Sixties as a response to certain kinds of music that was already being written and played (tape collage, "classical electronic" and certain kinds of jazz), yet it soon changed the music for which it was built. Before long it found its way into other kinds of music, such as pop and rock. So of course each of these musics changed the synthesizer as well. There is now, for instance, a "phaser" available on most synthesizers-- a device that electronically duplicates certain rock and roll recording techniques.

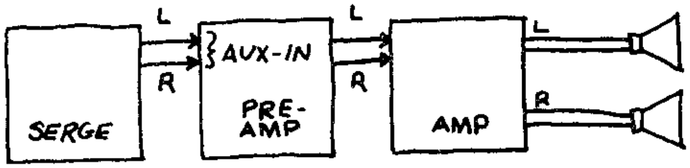
All of this is to say that a synthesizer is best understood within the context of the music in which it is to be played. A good way to learn about synthesizers is to listen to records and tapes of other synthesizer players, to go to concerts of electronic music and to read books available on the subject. In the beginning imitation is a good idea so that one can learn what is considered good standard practice, what is fresh and new, and what is a cliche.

The Voltage Controlled Modular Electronic Music Synthesizer (the instrument's full name) exists in two worlds at the same time: the world of electronics and the world of sound. Although the objective is to produce music, a little of both these worlds must be conquered to be able to use the synthesizer to its full potential. This is because, as will be explained in more detail later, the synthesizer is not really one instrument, but rather, an assembly of smaller ones (modules) which can be hooked together (patched) to create many different larger instruments. The synthesizer player really is, in part, an instrument builder, and just as an electronic organ designer must know something about electronics, acoustics and music, so must a good synthesizer player. However, you do not need to know much about either electronics or acoustics to use the synthesizer to make good music. The synthesizer itself is an excellent teacher of both simple acoustics and elementary electronics.

AMP AND SPEAKER

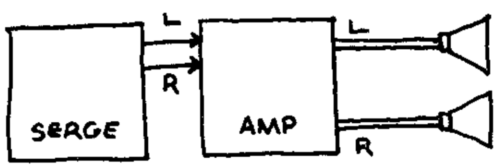
The Serge System does not contain its own amplifier and speaker so it must be connected to such a system to hear any sound. Below are the most common systems and where the Serge connects to them.

COMPONENT STEREO SYSTEM



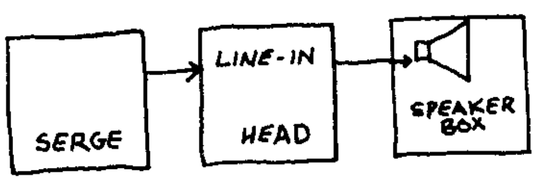
This is the most common home stereo system (though sometimes the pre-amp and amp are integrated into a single larger component.) The Serge should be connected to the AUX-IN inputs in the back of the pre-amp. The Serge can be thought of as another component, comparable in level to that of a reel-to-reel tape recorder. Remember to have both speakers hooked up and to switch the controls on the front of the pre-amp to AUX-IN. Keep the volume down until you have a good sense of the loudness of the Serge so that you don't overdrive your speakers. Most speakers are built for classical music sound spectra. Synthesizers are able to produce sounds that have far more energy in the "high" end and can damage speakers if care is not taken. If there is a headphone output you can use it to listen to the Serge without the speakers. Since it is a "stereo" you can listen to two voices seperately by connecting one voice to the left AUX-IN and one to the right AUX-IN.

POWER AMPLIFIER



Sometimes in a studio situation the Serge will be directly connected to a power amplifier. This is not recommended for home use. When this configuration is used, the only controls on the volume are on the Serge itself so it is important that they be kept at appropriate levels so as not to damage the speakers.

STAGE AMPLIFIER/SPEAKER (ROCK AND ROLL AMPLIFIER)

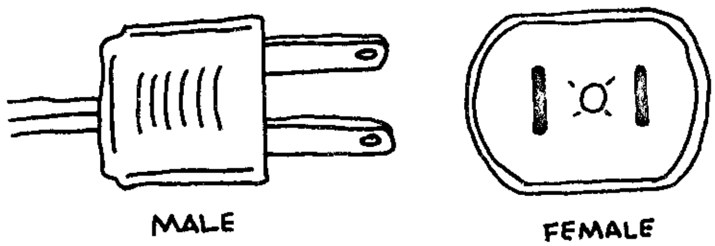


If you are using a stage or rock and roll type amplifier/speaker the Serge should be plugged into the HI-Level or Line level input.

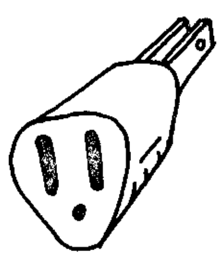
CONNECTING CORDS

Cords are used to connect between the various modules on the synthesizer, and also to connect the Serge to other electronic devices. Every connector on the end of a cord or on a device, has both a Type and a Sex (male or female). Only connectors of the same type and of opposite sex can be connected together. Below are the connectors commonly found in electronic sound equipment.

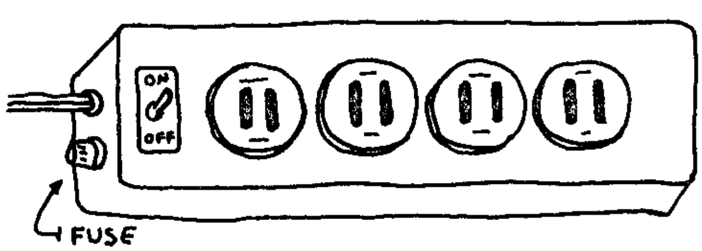
THE AC CONNECTOR



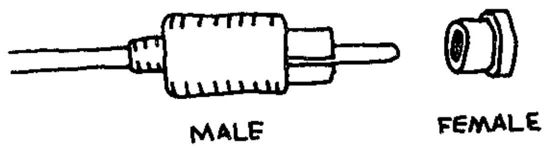
This is the "power" plug and is used, almost exclusively, to get 120 volt AC current out of the wall. If your equipment is not working the first thing to check is whether everything is plugged in. Many AC power plugs have a third, round nib. This is the ground connection. Unfortunately not all AC jacks have a hole to accomodate this nib so an AC adaptor must be used. However in most cases grounding is not required.



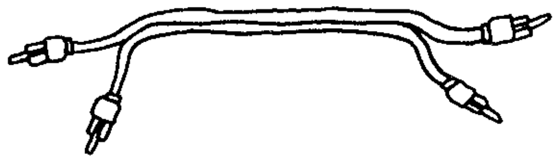
Another problem that arises with AC connectors is that you usually end up with far more males then females. One solution to this is an "octopus" which accepts up to eight plugs and also has the advantage of allowing the user to turn everything off at once. Most octopi also have a fuse which can protect your system from damage. If nothing goes on at all, check to see that the fuse is still good.



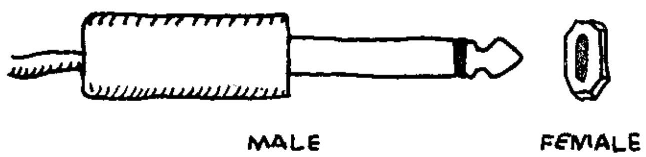
THE RCA or PIN CONNECTOR



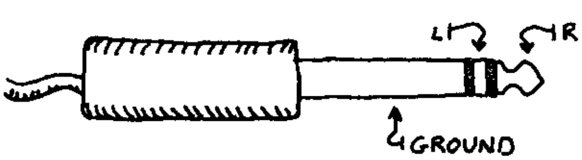
The jacks on the back of most home stereo systems are RCA female. Note that the males of this species have little skirts around them that grip the outside of the females. Make sure that these grip tightly by bending in these skirts just slightly. The skirt is the "ground" while the pin in the center is "hot" and carries the signal. Since they are often used in stereo situations, they often come in a joined pair. When using such a pair make sure that the correct ends are used.



THE PHONE CONNECTOR

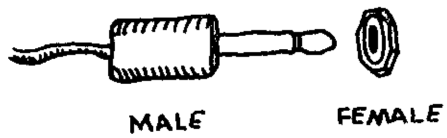


These are the plugs that are found on the end of guitar cords and the jacks that are found on the front of rock amps. (They are also the jacks found in old-fashioned phone systems and hence their name). The shaft of the Phone plug is divided by a narrow black band. The tip is the signal, or "hot", while the upper shaft is ground. A stereo version is also available with two black bands on its shaft. In the stereo version, the lower two segments carry the signal, while the upper section is ground.



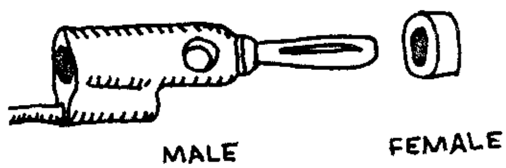
The stereo phone will only work when used with a jack made to handle this type of plug. Typically an instrument input is mono, while a headphone out is stereo.

MINI-PHONES (MINI-JACK)



These are the jacks found on the output modules of the Serge system. (Note: the Serge system uses the American-made version of the mini-phone manufactured by Switchcraft. Mini-phones made in Japan are almost identical, but do not use them since they become intermittent when used with the American jacks. Like the full-size phone plug, the tip carries the signal and the upper sleeve is ground.

BANANA CONNECTORS

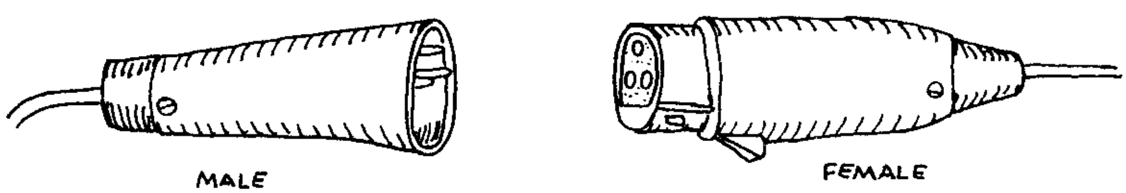


Within the Serge system almost all patching and connecting is done with banana plugs, which are so named because of their curious shape. These connectors are not grounded or "shielded", that is, there is no wrapping of wire mesh around the central wire. The cords are usually color-coded by length.

BARE WIRE

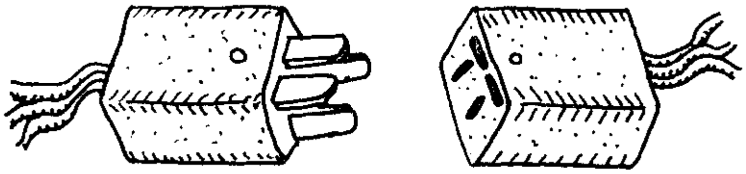
This type of connection is usually only found between amplifier and speaker. Always make sure that the speakers are connected to the amplifier before turning the amplifier on. If the system is a stereo system, make sure that "phase" is maintained by connecting both speakers identically. One of the strands is marked to make this simpler.

CANNON CONNECTORS



The cannon (or XLR) connector is most often found on microphone cords. It is a "balanced" connector with two signal lines and a ground. Note that the male's prongs are within a skirt and that this should not be confused with a female connector. The cannon connector clasps shut and a button or lever must be pressed to disconnect them.

CINCH-JONES



Often found on power supplies, speaker wires and trigger cables because of its inability to be connected backwards. This type of connector is used for wires that need to be connected and disconnected frequently.

MOLEX CONNECTORS

This type of connector is usually used for internal power supply connections since they are not frequently disconnected. Note that the four connections are numbered on the plastic housing.

ADAPTORS

At times it will be necessary to patch between a jack of one type and a jack of another type. There are two ways to solve this problem. The first is with a cord with two different kinds of plugs on the ends, and the second is with an adaptor.

Cords with two different plugs can be purchased or can be assembled without much skill. When making such cords be sure that the grounds of the two plugs are connected together and the signal parts of the plugs are connected together. Perhaps the most common way of dealing with the problem is with adaptors. These are small devices with a jack or plug of one type on one side and another type on the other side. It is also possible to find adaptors that go from a jack to the same kind of jack which are used primarily for making one long cord out of two shorter cords. When purchasing or asking for an adaptor be sure to specify not only the type of jack/plug needed, but also the sex.

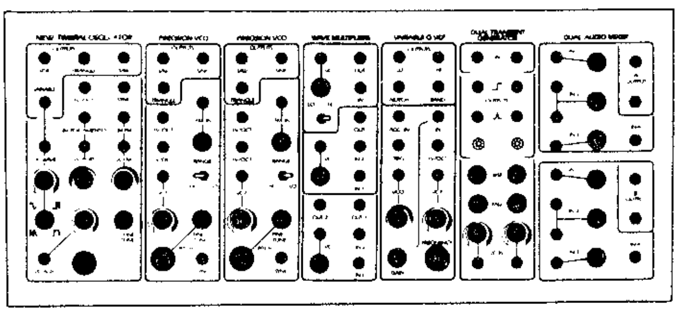
While it is possible to stack one adaptor on another, or to place them in elaborate combinations with various kinds of cords, it is wise to remember that the more connections one has the more likely something will go wrong. In fact, this problem is so common that if no sound is coming out of your system, right after checking to make sure that everything is plugged in, check the adaptors.

In the long run it is worthwhile to purchase the correct cords and adaptors for the job. You will find that they are easy to lose and a box for them comes in handy. You may also find that you lose fewer of them if you can mark them in some personal fashion.

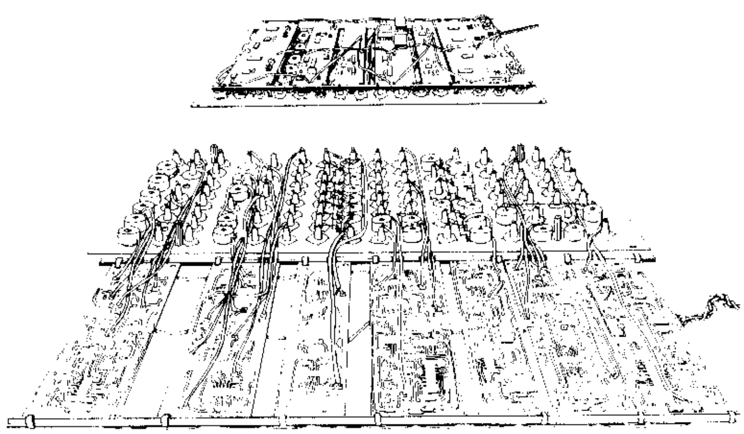
THE SERGE SYSTEM

While the Serge may look like a single machine, it is actually a collection of much smaller devices or, as they shall be called from now on, MODULES. This explains the word Modular in the full title: Voltage controlled modular electronic music synthesizer. Except for the internal power supply wires which supply them with the power necessary to operate, each module is COMPLETELY independent. They are no more interdependent than your TV and your toaster are when plugged into the same AC wall socket. Unless hooked together with PATCH cords these modules remain independent and can be used separately. But the real interest of the synthesizer lies in hooking these modules together.

The Serge is divided into panels that are 17 inches across and 7 inches high. Except for the sequencer/keyboard module, which takes up a whole panel by itself, each panel contains a number of different modules. Each module is surrounded by a line and has the name of the module at the top and the logo "SERGE" at the bottom. Modules with the same name are identical and interchangeable. Learning about a synthesizer is learning about what each module does separately and what they do when all hooked together.



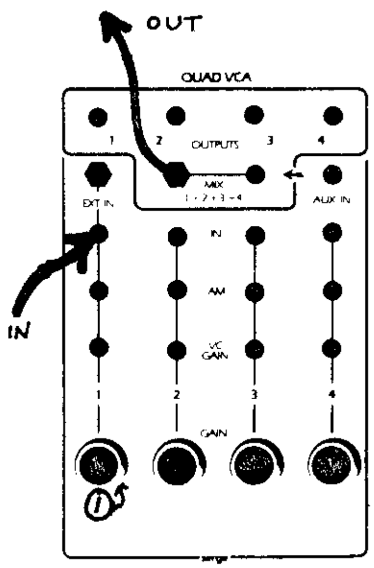
ONE SERGE PANEL



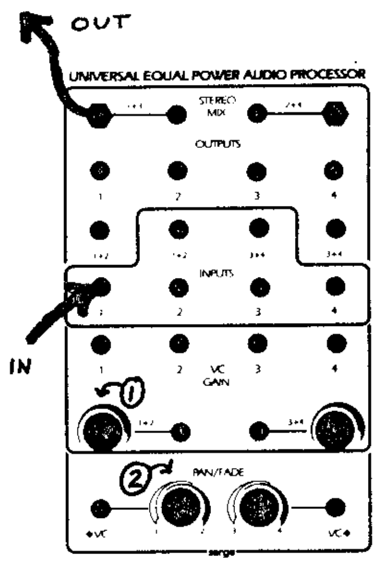
INTERIOR OF SERGE PANEL/RACK

CONNECTING SERGE TO SPEAKERS/AMP

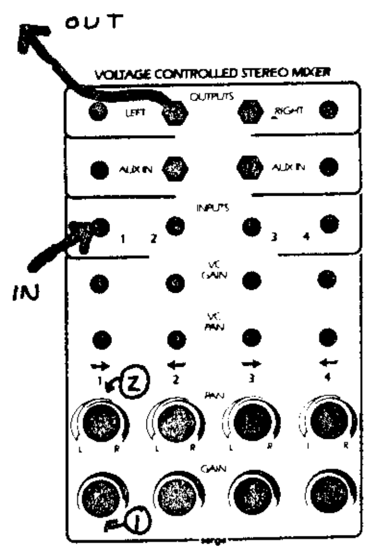
In all Serge systems there is at least one module whose function is to connect the Serge to the amp/speaker system. These are called "output modules" because they take the signals from the Serge and OUTPUT them to the speaker and amps. Output modules have inputs that accept banana plugs from the other modules on the Serge and have either mini-phone, phone or RCA outputs. These non-banana connectors can go directly to the amp/speaker system. Below are diagrams of the various modules that serve this function. Each diagram is labelled to show where to "input" into the module and where to take the "output" from. The output cord goes to the amp/speaker system. Under each diagram are instructions for setting the dials or POTS (short for potentiometer) on the module. It is important that these instructions be followed at this point. In particular, no cords other than those listed should be patched to the output module. In the following section this module is referred to as the Output Module.



① FULL RIGHT



① FULL RIGHT
② FULL LEFT



① FULL RIGHT
② FULL LEFT

If the Serge is not already on, flick the power switch ON now. The amp should be turned all the way down.

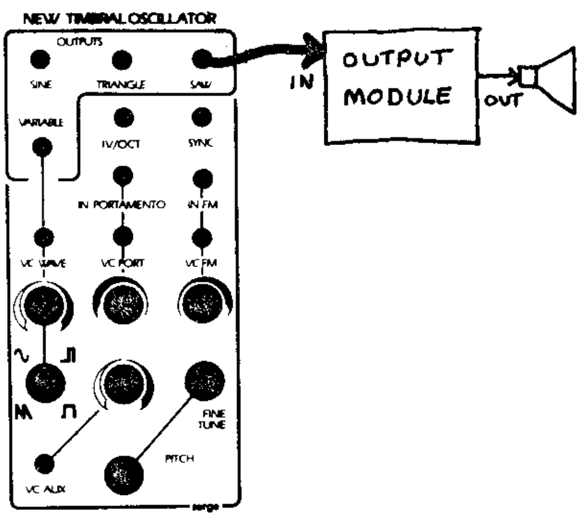
You are now ready to make your first electronic sound, which will also test whether all the inter-connections so far are good. After the sound the theory.

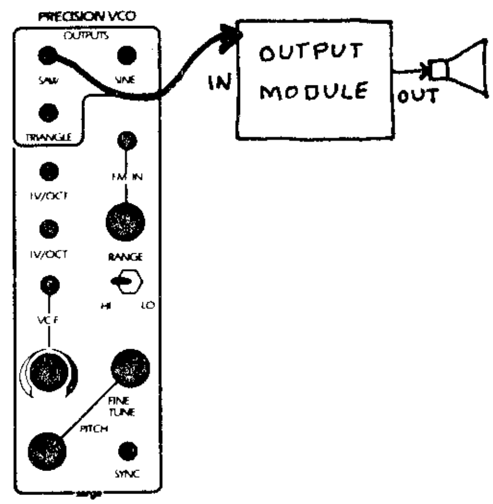
LEARNING PATCH NUMBER ONE

Or, Making Sounds Knowing Nothing.

STEP ONE

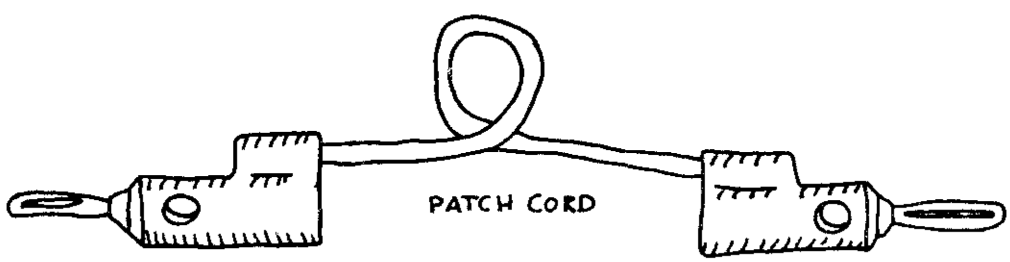
1.1 Locate a Precision VCO or a New Timbral Oscillator on the Serge. If you choose to use a Precision VCO, set its Range switch to HI. Both of these modules are Oscillators and differ from each other only by some special features. All identically labelled modules are identical and interchangeable, so that if the system contains more than one of these modules any of them will do. For brevity these two modules will be referred to as "OSC".





In the following discussion the Precision VCO will be used as the example Oscillator module, though the New Timbral Oscillator can be substituted as in the above diagram.

1.2 It should be clear that all of the jacks on the OSC, though of different colors, are Banana jacks, as are almost all the connectors on the Serge system. This means that only one kind of plug is needed to patch from anything to anything else: a banana to banana cord. This kind of cord, since it is the primary cord for patching various modules together, we will simply call a PATCH CORD.



1.3 At the top of the OSC there is an OUTPUT section containing three jacks labelled SAW, SINE, and TRIANGLE. Like most modules on the Serge the output section will be enclosed within a border. Take one end of a Patch cord and insert its plug into the jack labelled SAW. The other end of the cord should be inserted into "Input 1" of the output module (see previous section). Make sure that the volume control on your amp is all the way down.

In the above diagram, and in all subsequent diagrams in this section, a patch cord will be shown by a thick line drawn between the appropriate jacks on the modules. The output module will be shown as a square with Input 1 labelled. The amplifier/speaker is shown by a small speaker.

1.4 On the OSC you will find a POT that is labelled PITCH. (POT is the name given to all knobs on synthesizers because beneath the faceplate the knob turns a potentiometer.) Turn this POT so that its pointer is set to 11 o'clock.

1.5 There should be no sound from your speakers until, slowly, you turn up the volume on your amp. You should hear a buzz. Set the volume on the amp so the sound is at a comfortable level. Try moving the PITCH POT back and forth. You should hear a whining sound that moves up and down not unlike the sound of a police siren. The further the Pitch POT is turned clockwise the higher the sound becomes; the further counterclockwise, the lower the sound.

TROUBLESHOOTING

IF YOU HEAR NO SOUND, then it is time to "troubleshoot"--that is, to look for the problem and fix it. This should be done as methodically as possible and with as cool a head as you are able to muster. Things go wrong for the best engineers. Learning how to troubleshoot is just another, and rather important, thing to learn about synthesizer playing. Below are some elementary trouble-shooting steps if no sound is heard:
(Be sure to do these things one at a time.)

A. Check to see that amp, pre-amp and Serge system are all plugged into an AC outlet and that each device is turned on. All the modules on the Serge are turned on by the single master switch. If there is a Touch-Activated Keyboard Sequencer in the system, one of the red lights on one of the touch pads should be lit.

B. Visually check that everything is patched correctly together. The SAW output of the OSC to Input-1 of the output module. The output of the output module to the amp. The amp to the speakers.

C. Check that OSC's pitch POT is set to the 11 o'clock position and that the small switch on the PCO is set to HI (if you are using the PCO as the OSC). Try moving the pitch pot right and left to see if anything happens.

D. Check the settings on your Output module against the settings listed.

E. Check to see that all the settings on your pre-amp are correct. It should be set to AUX if the synthesizer is patched into AUX. The tape monitor switch should be off. Try unplugging the cord that goes to the amp from the Output module and touching the disconnected end with your finger. If the amp and speaker are set up correctly you should hear a buzz or hum.

F. If all else fails try switching the patch cords. It would not be the first time that a patch cord or adaptor was broken. Try a different OSC.

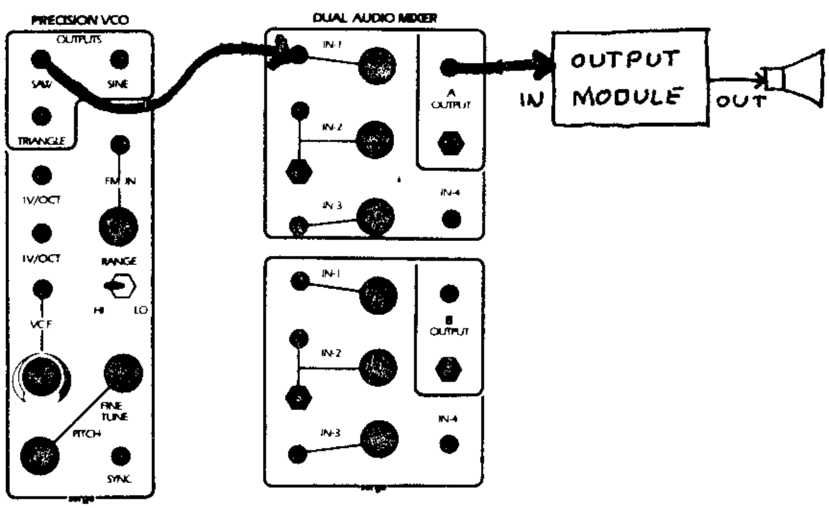
Now that you have your first PATCH working there are a few things you can try:

1.6 Try turning the Pitch pot as far to the left as it will go. At some point it will start to become a series of clicks. Then try turning it as far to the right as it will go. The pitch will get higher and higher until it is barely within the human range of hearing. Try turning the POT labelled "fine tuning". Like the Pitch POT it alters the Pitch, but over a much smaller range.

1.7 Unpatch the plug inserted into the SAW output. The sound will suddenly stop. Re-patch into the SINE output and then the TRIANGLE output. You should notice a difference in sound quality between these outputs. This quality is called "timbre" and is one reason why different instruments playing the same pitch can be distinguished. For instance, a piano playing a middle C and a violin playing a middle C are distinguished by their timbres.

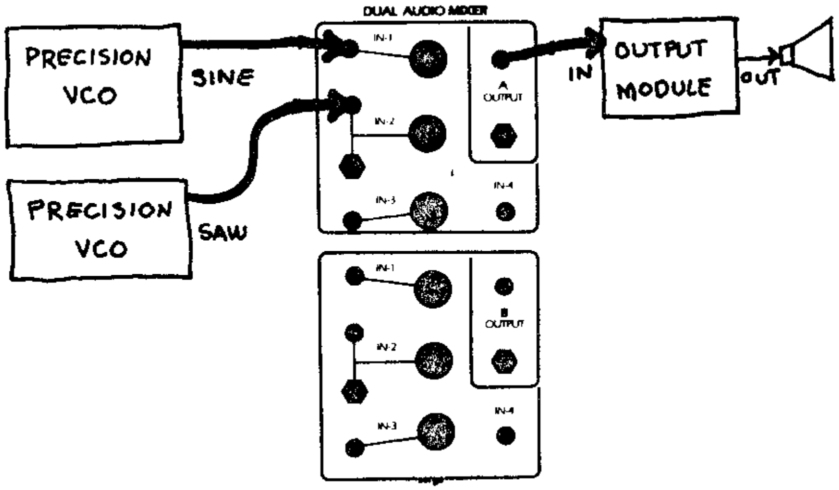
STEP TWO

2.1 On the Serge system find the module labelled Audio Mixer. Like a few modules on the Serge this is a double module, that is, it is actually two identical modules, one above the other. For this patch use the top one.



2.2 Patch the SAW out of the OSC to "IN-1" of the Mixer. Patch the output of the Mixer to Input-1 of the Output Module. The POT that is associated with IN-1 by a line should be turned full left. Make sure that the Pitch POT on the OSC is set as in STEP ONE.

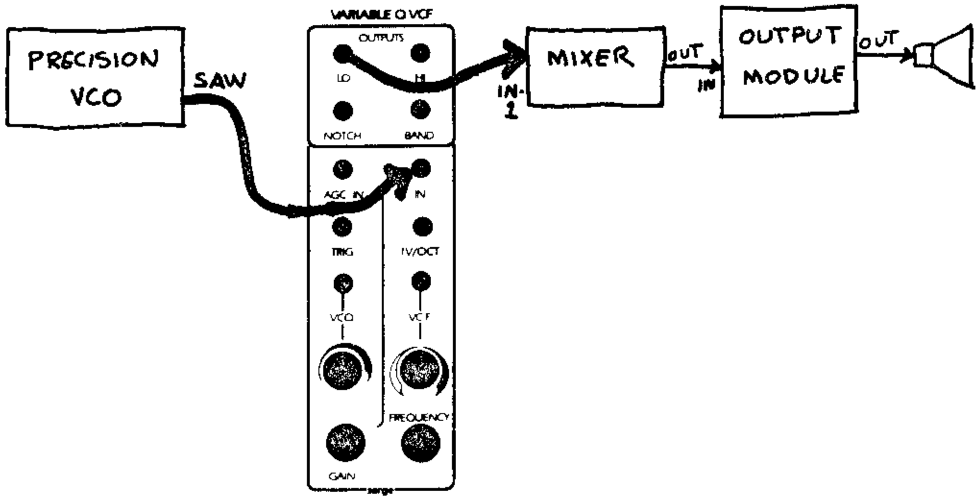
2.3 Slowly turn the POT associated with IN-1 to the right. The buzzing sound heard in STEP ONE should now once again be heard from the speaker, but at a softer volume. By turning the Mixer POT further and further to the right, the sound can be made louder and louder.



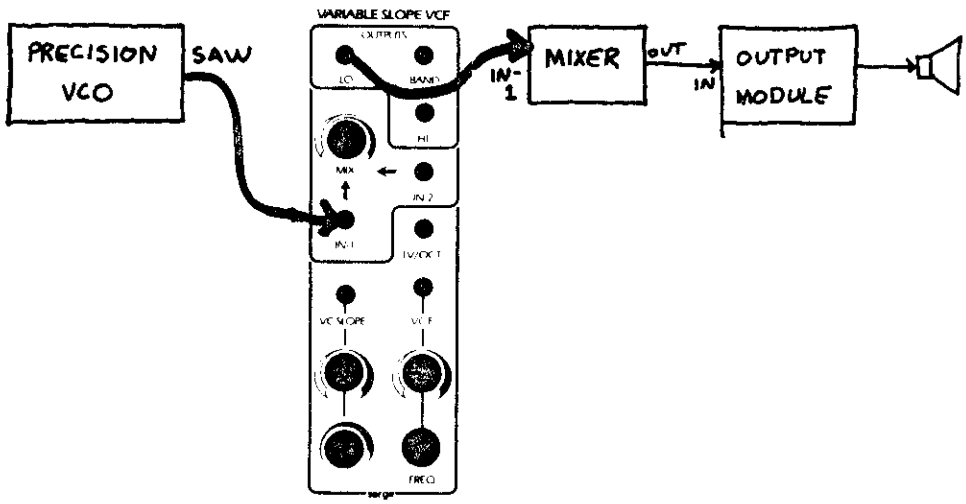
2.4 Using a second OSC on the Serge set up the above Patch. Use the second OSC's SINE output to IN-2 of the mixer. Slowly turn its associated POT up. A second sound being MIXED with the first should now be heard. The loudness of each of the two sounds can be determined by their associated POTS on the Mixer. Note that you can adjust each OSC's Pitch separately. If you have a third OSC you can patch its TRIANGLE output (or any of its outputs) to IN-3 of the MIXER.

2.5 When a single piano note is struck the sound quickly gets loud and then slowly dies out (if the pedal is down). Try getting this effect by turning the Mixer's pot quickly to the right and then slowly back to the left.

STEP THREE



OR



3.1 The Variable Q VCF and the Variable Slope VCF are both filters and differ only in some of the functions available on them (VCF stands for Voltage Controlled Filter). Either one can be used for this Patch, although in this text we will be referencing the Variable Q VCF. If you are using the Variable Q VCF, set the VCO POT full left and the GAIN POT Full Right. If you are using the Variable Slope VCF, input should be to IN-1 and the MIX POT should be full left. VC SLOPE POTS should be set full left. Connect the output from the VCF's LO jack to the Output module.

3.2 Turn the PITCH POT of the OSC to about 9 o'clock, which will produce a fairly low sound. The GAIN POT on the Mixer should be at the level that it had been adjusted in the previous step.

3.3 Turn the FREQ knob on the Filter all the way right. You should hear a buzz simiiliar to the one in the previous steps.

3.4 Slowly turn the FREQ POT on the Filter to the left. The sound should get softer and softer and finally disappear. However, it will get softer in a different manner than when the Mixer's POTs were turned down. This time, the sound seems to get more and more muffled. The high buzz in the sound disappears first, and then the rest of the sound. Try OSCs set to different FREQs, Different outputs of OSCs (SAW, TRI, SINE), and a Mixer output. When listening to these sounds start with the Filter's FREQ POT set full right in order to hear the full sound before filtering it.

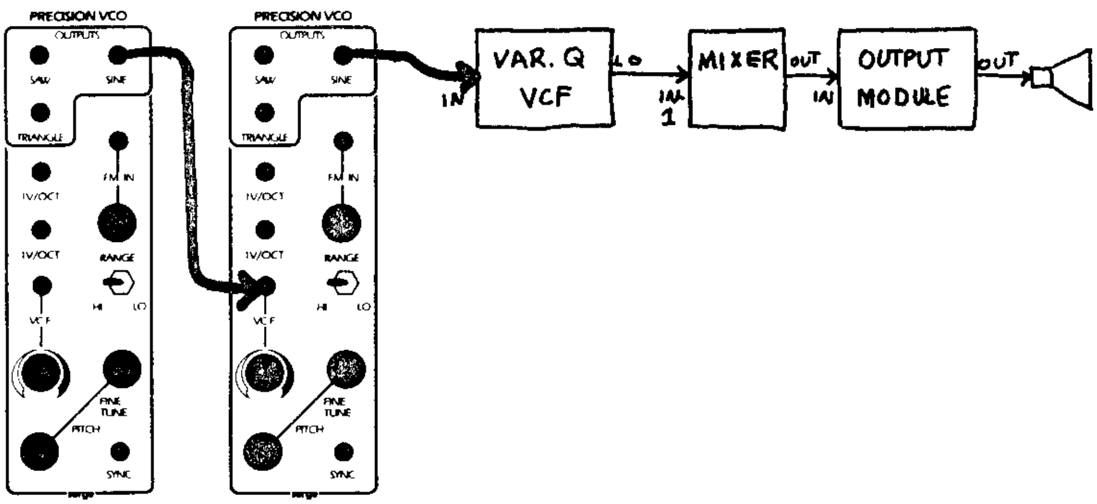
3.5 Try the same Patch but using the BAND output of the Filter. Try the HI output of the Filter, but start with the FREQ POT full left.

3.6 It should be clear that there are a lot more POTS to turn and adjust than there are hands to turn them. It should also be clear that the sounds so far produced are still very simple and hardly music. These two problems are attacked in STEP FOUR.

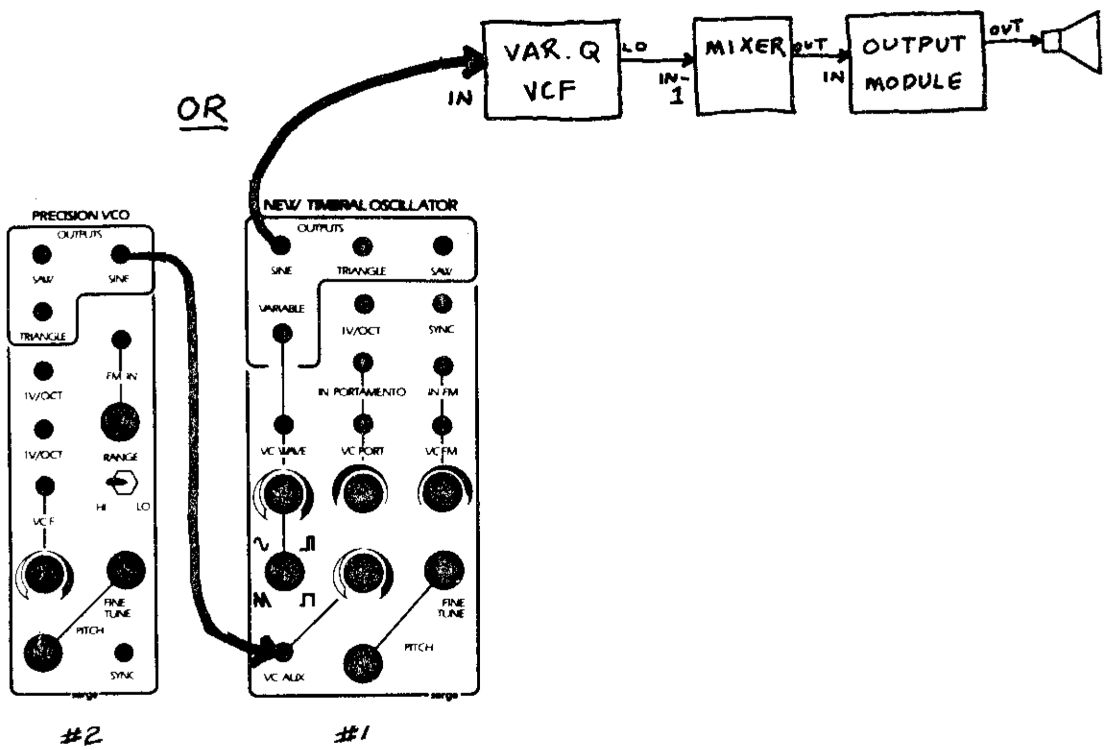
STEP FOUR

Certain things about STEP THREE can be easily said. The PITCH POT on the OSC controlled how high or low the frequency of the sound would be, while the GAIN POT on the Mixer controlled how loud or soft the sound would be. The Filter controlled certain aspects of the timbre or quality of the sound.

Loudness, Pitch and Timbre are all components of a single sound. Even in traditional scores these three "parameters" are notated.



#2 #1



4.1 Set up the above Patch using a Precision Controlled VCO in combination with the OSC already being used. If the OSC that you had been using is a Precision Controlled Oscillator, then the new OSC should be patched to the VCF jack. If the OSC in use was a New Timbral Oscillator, then the Precision Controlled VCO should be Patched to the VC AUX jack. The new OSC will be referred to as OSC #2.

4.2 On OSC #2 switch the HI/LO to LO and turn the PITCH POT full left. On OSC #1 either the VC AUX POT or the VCF POT (depending on which OSC. is being used) should be turned right to about 3 o'clock.

4.3 Set the PITCH POT of OSC #1 to about 9 o'clock; set the GAIN POT of the Mixer to a comfortable volume. Set the FREQ POT on the VCF full right (use LO output). The Pitch of the output will start to rise, getting higher and higher. It might get so high that you will **not be** able to hear it. Then suddenly the sound will drop to a very low sound and start rising again. It is JUST AS IF you were turning the PITCH POT of the OSC slowly to the right and then suddenly, fast as light, turning it full left.

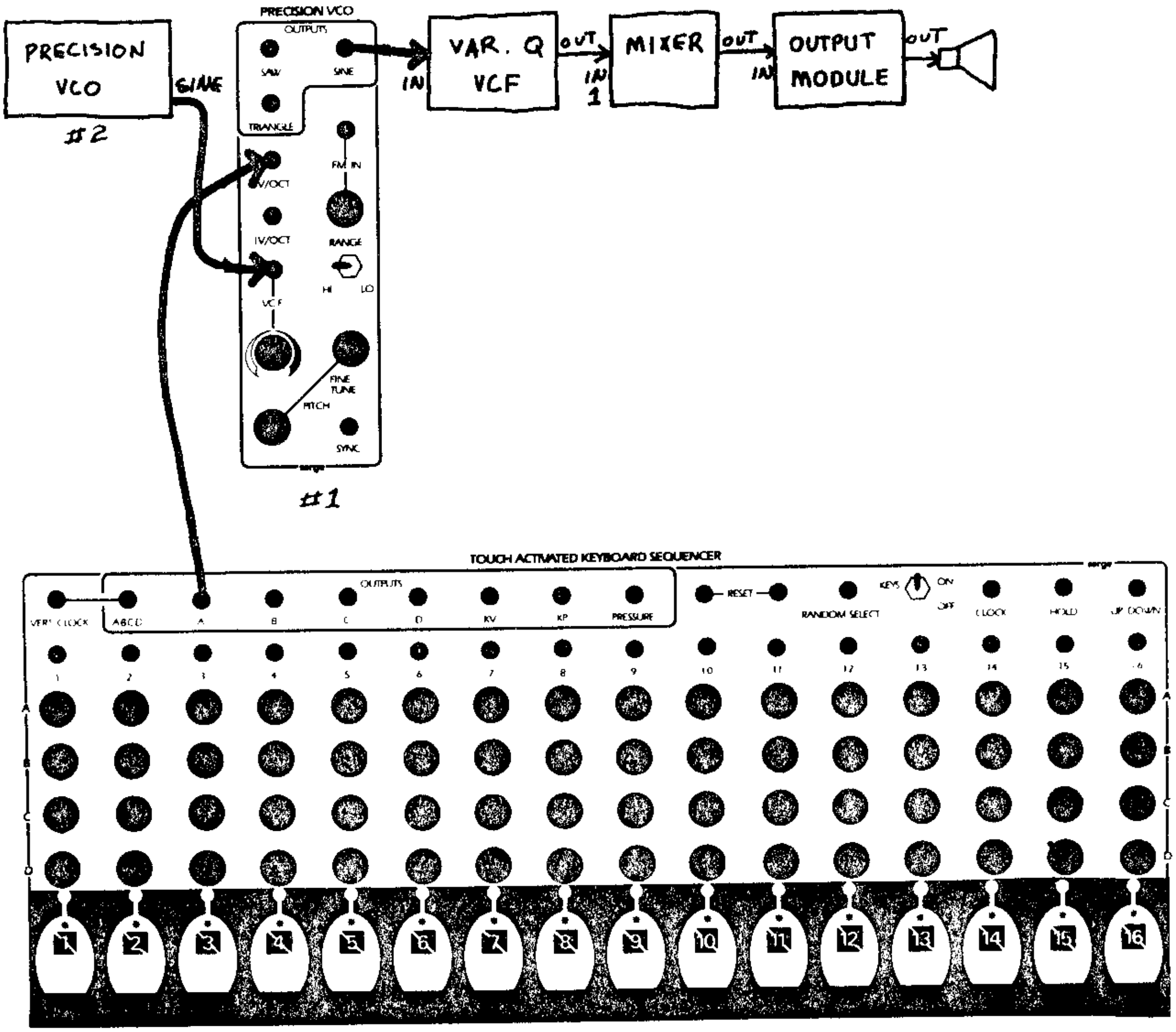
4.4 Slowly turn the PITCH POT on OSC #2 to the right. As you do this the speed at which the pitch rises should increase. As this rising and falling accelerates to more than once a second, the ear starts to hear it as a constant wavering sound. When it approaches 20 times per second, the ear no longer distinguishes individual sweeps (though they are still happening), but rather hears a new, very low sound as well as some strange higher sounds. Turn the PITCH POT on OSC #2 higher and higher listening to results as you do so. Try switching OSC #2's HI/LO switch to HI and sweeping the PITCH POT to the left. Experiment with changing the PITCH POT of OSC #1.

4.5 Return OSC #1 and OSC #2 to the settings at the beginning of this STEP. Try using the SINE out and then the TRIANGLE output of OSC #2. When the SINE out is used, instead of sweeping up and then suddenly falling back down, the sweep is a gentle one both up and back down. With the TRIANGLE out the sweep should be similiar to the SINE out except a little sharper at the top and the bottom. The TRIANGLE wave will also go higher than the SINE, and the SINE will make the output sound go below the initial setting of the PCO. Try these two other outputs with different POT settings on both PCO #1 & #2.

4.6 Re-set OSC #1 and #2 to the settings at the beginning of this STEP. Slowly turn the VCF POT or the VC AUX POT from its 3 o'clock setting to its 12 o'clock setting. The sweep will take as long, but won't get as high or drop as low. At 12 o'clock there should hardly be any sweep at all. As the POT is turned past 12 o'clock to 9 o'clock the sweep should start heading DOWNWARD, jumping UPWARD just the opposite of what it had been doing.

STEP FIVE

5.1 Set the Patch in the previous step. Set the VC AUX or the VCF POT of OSC #1 to 12 o'clock so that there is almost no sweeping and so that the sound is neither muffled nor too soft. The FREQ should be set around 11 o'clock.



5.2 Locate the TOUCH ACTIVATED KEYBOARD SEQUENCER (or TKB as it will be called from now on) which is normally at the bottom of your synthesizer and fills an entire panel.

5.3 At the top of the TKB find a switch labelled "KEYS" and turn it "ON".

5.4 Beneath this upper section there are 4 ROWS and 16 COLUMNS of POTS. The Rows are labelled A-D and the columns are numbered 1 to 16. Set the POTS in Row A to 16 different positions--it doesn't matter what positions these are so long as they are all different.

5.5 At the top of the TKB you will find the output section. It contains a number of different outputs including ABCD, A, B, C, & D. Patch from A to 1V/OCT in OSC #1 as in the above diagram.

5.6 Touch the different keypads at the bottom of the module with your finger. As you do, notice that a little light that goes on on the pad you have touched. The light remains on until you have touched another key, so only one light will be on at a time. This light indicates that the column above it is ACTIVATED. The pitch of the sound should change as each pad is touched.

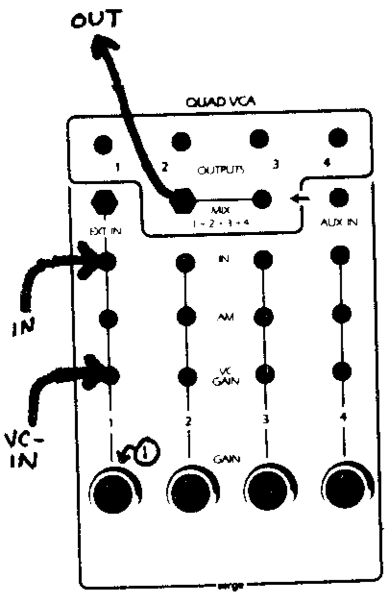
5.7 Turn the pot in the A Row in the column that is now activated (i.e. has its light lit). It should be just like turning the PITCH POT of the OSC #1, but BY REMOTE CONTROL.

5.8 Try setting the POTS in Row A so that a tune can be played on the keypads.

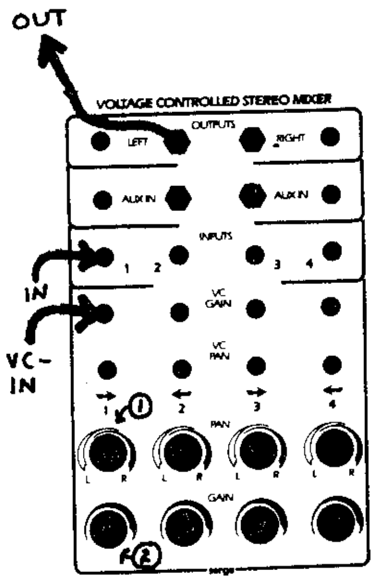
STEP SIX

6.1 Just as an oscillator's frequency can be made to rise or fall by remote or Voltage Control (that is without turning that oscillator's Pot) there are a number of mixers and other modules on the Serge which can control the GAIN of a sound by voltage control. The volume can thus be increased or decreased without turning a pot. In general these modules are called Voltage Controlled Amplifiers (VCAs -- or sometimes: GATES) when they have single inputs, or Voltage Controlled Mixers when they have more than one input.

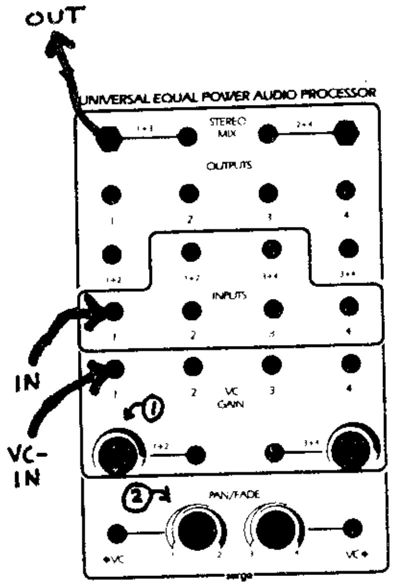
6.2 Almost all the Output Mixers in the Serge System are voltage controllable, meaning that the Gain of their inputs can be controlled by a voltage coming from another module. Each input to the mixer has one or more VC inputs. Below are noted the VC inputs for the various Output Mixers. These will be noted as "VC-IN" on the Output Mixer module.



① 11-12 O'CLOCK

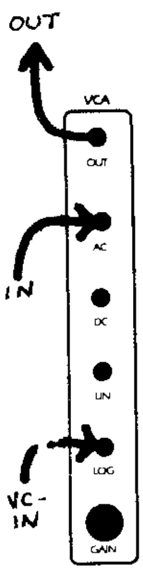


① FULL LEFT
② 11-12 O'CLOCK

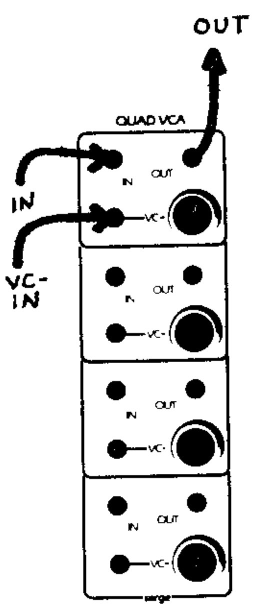


① 11-12 O'CLOCK
② FULL LEFT

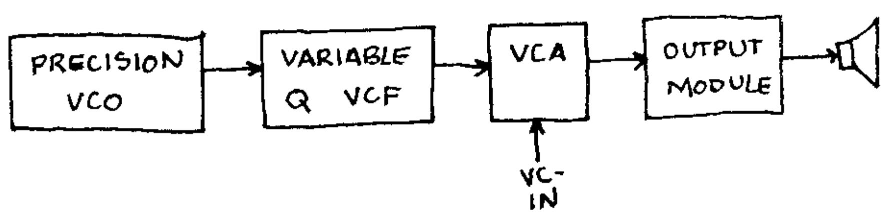
6.3 Some Serge Systems have independent Voltage Controlled Amplifier modules which often contain extra features. Patched as another link in the synthesis chain they can be used instead of the VC-output mixer to provide voltage controlled gain. In the diagrams below the VC-IN is noted on these modules. In the patches that follow, the VC-IN will be shown to the Output Mixer, but it can refer to these independent VCAs as well.



SET GAIN TO 9 O'CLOCK

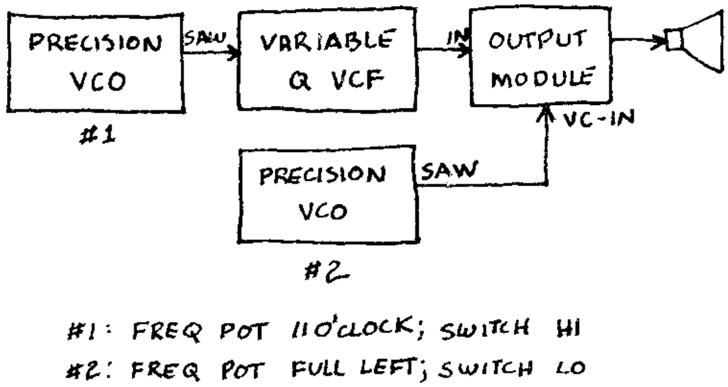


SET VC-POT TO 11-12 O'CLOCK



TYPICAL VCA PATCH

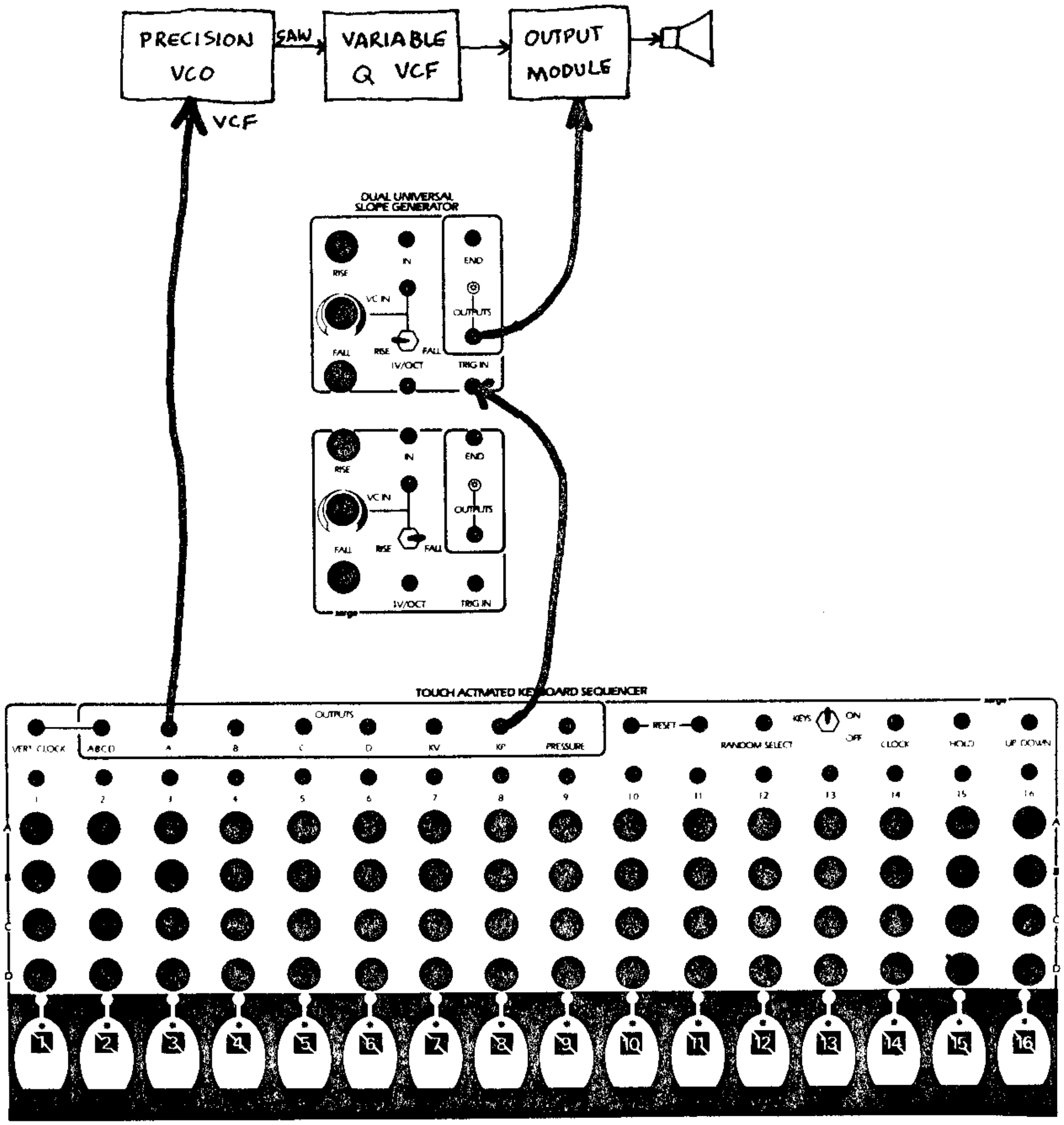
6.4 In the following patch the PCO which is connected to the VC-IN of the mixer should be set "LO" and its FREQ turned to the left. It will increase and decrease the GAIN (loudness) of the sound in the same fashion that it controlled the frequency of the other oscillator in the previous patches:



6.5 Try increasing and decreasing the frequency of the controlling PCO. The bursts of sound should get faster and slower. Also try using different outputs of the PCO (sine, triangle, saw) and listen to the different "shapes" of the bursts. This shape is called the "Envelope" of the sound.

6.6 The Dual Universal Slope Generator (DSG) is a dual module with two identical modules in it, one above the other. For this patch the upper one will be used.

6.7 On the output section of the TKB find the jack labelled KP (Key Pulse) and patch it to the TRIG IN of the DSG. The A output of the TKB will control the frequency of the PCO as in previous patches. The output of the DSG should be patched to the VC-IN of the output module.

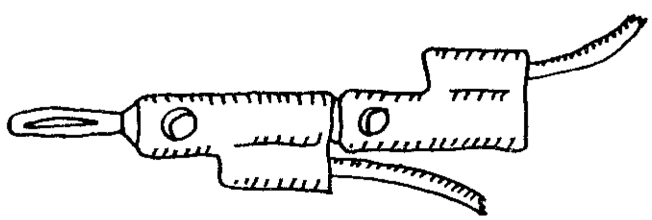


6.8 In this patch the DSG produces a single envelope, unlike the PCO, which produced a continuous stream of envelopes. Furthermore it will produce this envelope on demand, in this case whenever a key is touched on the TKB. In this sense the TKB is controlling the DSG. The length of the RISE and FALL times of the envelope are controlled by the RISE and FALL pots on the DSG. To produce the approximate envelope of a piano set the RISE pot to 1 o'clock and the FALL to 9 o'clock.

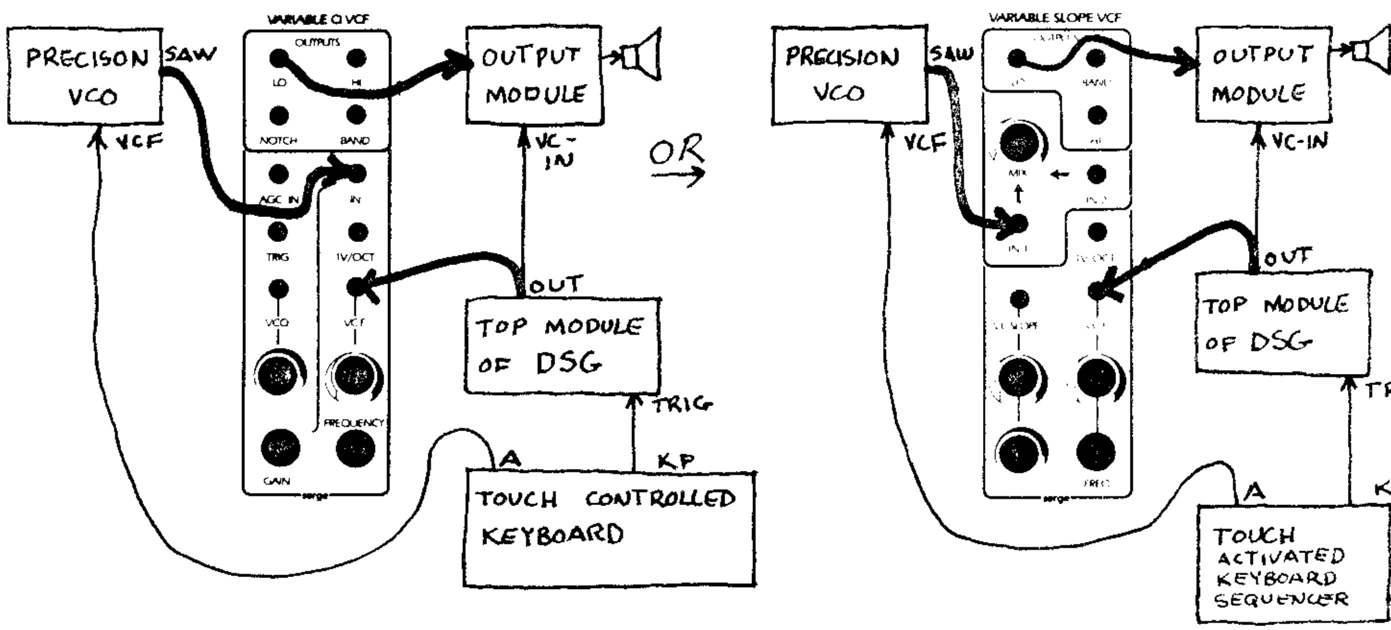
6.9 It is possible to get a somewhat "backward" feel to the sound by reversing the settings of the two pots. It is also possible to get very long envelopes or envelopes so short that they sound like nothing more than clicks.

6.10 At the back of each plug on the patchcords is another Banana Jack. This allows you to "stack" the plugs so that you can patch more than one cord from a single jack on the Serge. You should NOT STACK MODULE INPUTS, however. In order to combine voltages, a Mixer or Processor is required. If you try to stack at the inputs, you will be connecting the outputs of modules directly together. This is the one connection that should always be avoided in the synthesizer, and if it is not adhered to, it is possible that you will damage the modules. This message will be repeated: Do not connect outputs together.

OK, now that we've said this, let's moderate it a little. You should not live in total fear of using the system. In every case, modules are protected against inadvertent output shorts, but if a lot of outputs are connected together over a long period of time, it is possible to put an excessive strain on the power supply. If you find outputs shorted together accidentally, simply remove them, but get in the habit of thinking of "multing", or stacking, outputs whenever you are stacking banana cords.

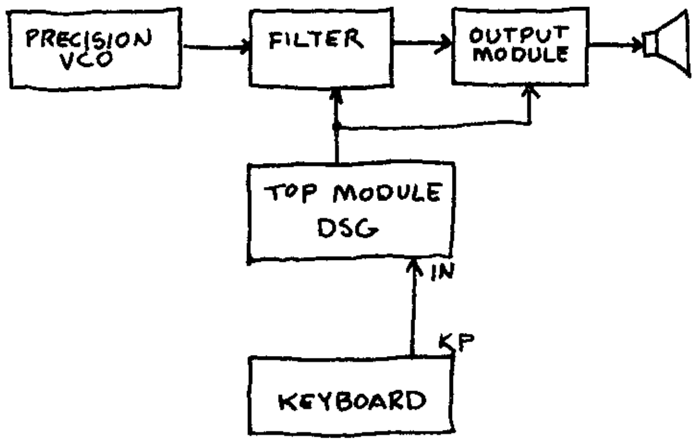


6.11 Stacking cords in this way, patch the output of the DSG to the VC-IN of the mixer AND to the VCF input of the filter. Set the associated VCF Pot full right. Set the FREQ Pot of the Filter to 9 o'clock.



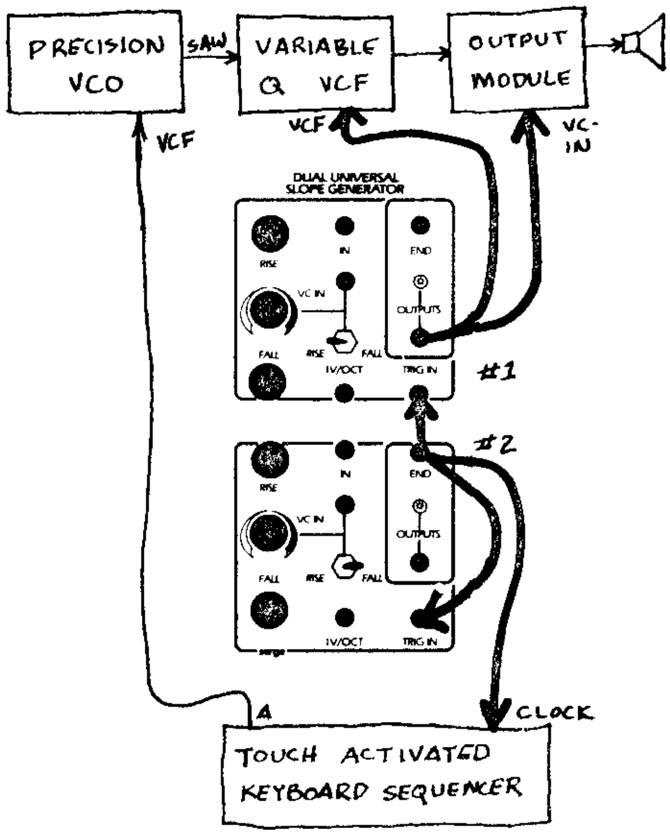
6.12 Tap a key. The sound should now approach very closely the sound of an acoustic instrument. Like the oscillator and the MIXER, the Filter is now being controlled. Not only does the sound get softer and softer as it dies away (from the action of the Mixer), but it should sound as if it is being damped, or losing its "highs" as time progresses. This is what happens when a piano note is struck.

6.13 In the following patch all remains the same as the above patch except that the KP is now connected to the black IN jack of the DSG.



What has been added here is the ability to hold or sustain a sound by keeping one's finger down on the keyboard. The RISE will begin as soon as the key is touched, but the FALL won't begin until the finger is released.

STEP SEVEN



7.1 Using the lower module of the DSG, which will be referred to as DSG #2, patch from the END jack to the TRIG IN jack of the SAME module. This will be the first time that you patch between two jacks of the same module.

7.2 Using the LED light on DSG #2 as a guide, set the RISE and FALL of DSG #2 so that the light flashes about once per second or even slightly slower.

7.3 Remove the cord from KP on the TKB to TRIG IN on DSG #1 and Patch it from END on DSG #2 (stacking the plugs) to TRIG IN on DSG #1. Also stack a patch cord from the END on DSG #2 to the CLOCK jack on the TKB. (Again, note that we are stacking at the output. No input has more than one output connected to it, but the output of DSG #2 is multed to two places, the TRIG Input of DSG #1 and the CLOCK Input of the TKB.)

7.4 The TKB should now begin to "sequence" all by itself, stepping through its columns as if you were touching one keypad after the other.

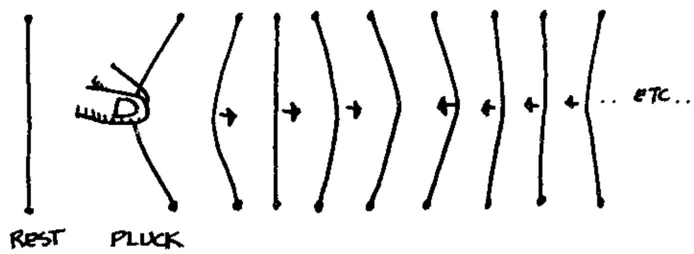
7.5 Try turning the RISE and FALL pots of DSG #2 further and further to the right. The TKB should step even faster. You may have to adjust the RISE and FALL POTS on DSG #1 to the left to keep the sounds from "blurring" together. Try turning the POTS on DSG #2 farther and farther to the left to make the TKB step slower.

7.6 The PATCH that you have now set up is one of the most common PATCHES on a synthesizer, yet it is but one of an infinite variety. To this PATCH other modules may be added that change the timbre, that create complex rhythms, that add second, third and fourth voices. From this first section it is hoped that you have gained some sense of the sounds that a synthesizer can create and to some degree how it creates them. In the following section the theory of synthesis will be explored.

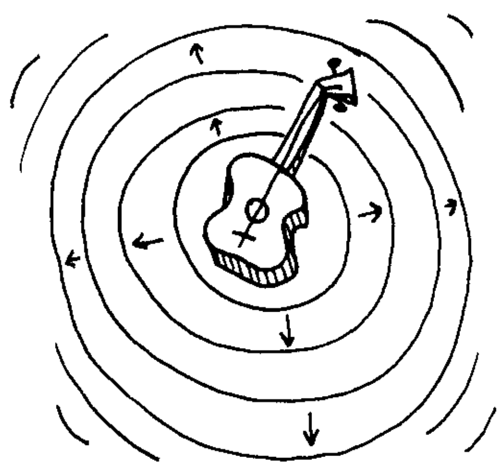
ELECTRONIC MUSIC THEORY

SOUND

Sounds are vibrations of the air caused by vibrating objects. Take a simple musical example--the string on a guitar. When it is plucked, it is pulled in one direction and released. Because it was under tension from the pulling, it snaps back to its original position and because of its momentum, it keeps going through its at-rest position to an opposite state of tension.



It proceeds to move back and forth, each time with a little less power, until it comes to rest in its original position. Almost all struck or plucked instruments vibrate in some variation of this action. When the string is released it pushes the air in front of it causing a slight extra compression of the air molecules or, put another way, a slightly higher pressure. This is called "compression". When the string flicks back, it causes a slight vacuum, or low pressure area. This is called "rarefaction". As the string vibrates back and forth more and more of these compression and rarefaction areas are created. They act like ripples in a pond, spreading out quickly and always at the same speed, the speed of sound.



If you are standing some distance away from the vibrating string when these ripples reach you, if there were some way of counting how many waves occur per second, many things could be told about the string itself! For one thing, because the speed of sound is constant, you would know how many times the string vibrates in a second. This number is called the FREQUENCY of a sound. The second thing you would want to determine is how strong the ripples are, that is, how compressed the compression wave is and how vacuous the rarefaction wave is. This strength is called the AMPLITUDE of the wave. When working with sound it can also be called the VOLUME or LOUDNESS of the sound. The Amplitude can tell you one or both of two things: how strong the source of vibrations was (i.e. how powerfully it could push air around) and/or how far away the source of the vibration is, because the amplitude of the ripples decreases with distance.

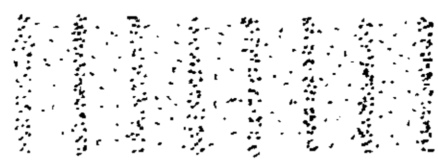
There are a number of other things we wish to detect about the sound waves that reach us. No object vibrates simply. Each has a characteristic "waveform" that, when perceived, can identify that object. This is called the TIMBRE or quality of the sound and is how we can distinquish a piano from a violin. We would want to detect these variations and have a sense of where the sound is coming from.

We perceive these complex waves with our ears. We hear different Frequencies as different PITCHES and we can hear them over the range of about 20 to 20,000 cycles (vibrations) per second.

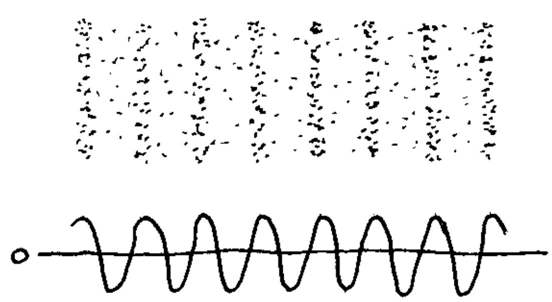
We perceive Amplitude as loudness, and remarkably, we can sense the amplitudes of rustling leaves or those of a jet plane. The jet produces compressions and rarefactions nearly one million times greater than the leaves!

Without going into much detail, this is the way the ear works: The pressure inside the human head remains constant (though adjusted to the normal pressure of the atmosphere of the air.) When there are no sound waves in the air, the eardrum is at rest between two areas of equal pressure. However, when a sound wave ripples past, with its fluctuating bands of high and low pressure, the eardrum is pulled slightly outward during a rarefaction wave and pushed slightly in by the high pressure part of the wave. This means that the eardrum is going in and out at the same rate (with the same frequency) as the original sound source. The eardrum's vibrations are transmitted by means of small bones to the cochlea, a spiral organ in the inner ear filled with a liquid and coated on the inside with millions of small hairs. Each of these hairs is connected to a nerve ending through which these signals are sent to the brain.

If we could take a picture of a small section of air through which a sound wave is moving, it might look like this:

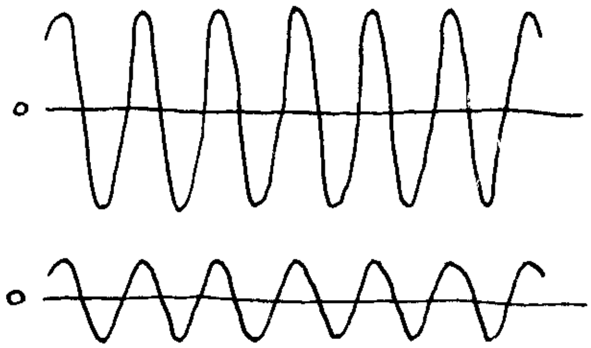


In this drawing each dot represents a few million air molecules, but even with this simplification it is a rather clumsy way of describing how a wave "looks". Here is a better way is to describe the "pressure" at each point of such a wave:

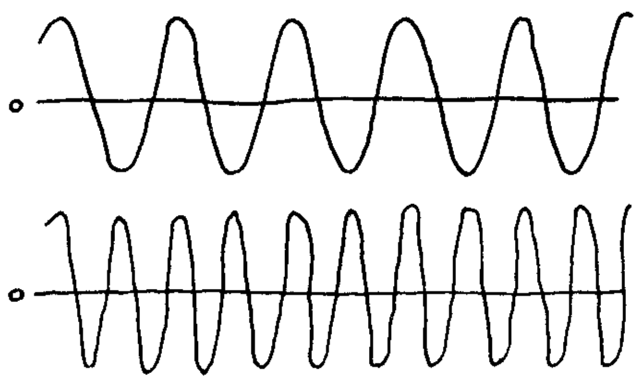


The line labelled "0" is normal pressure and the wavy line is a graph of the pressure of the wave. When the wavy line is above the 0 line, the pressure is greater than normal air pressure, when below the 0 line, it is less than air pressure.

Below are two sound waves drawn using pressure graphs:

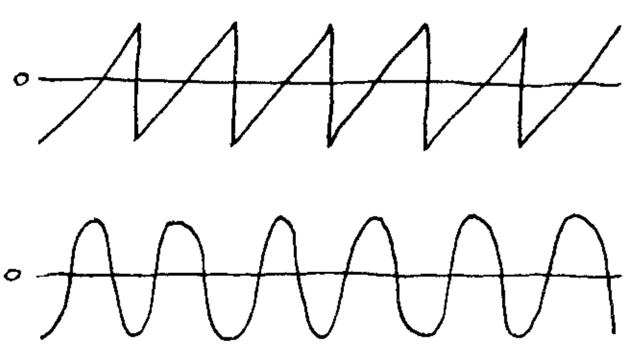


The difference between these two waves is that the top one goes further above and below the 0 line than the bottom wave. This indicates that its Amplitude or loudness is greater and is measured from "peak to peak", from the top of the highest peak to the bottom of the lowest trough. Below are two more waves.



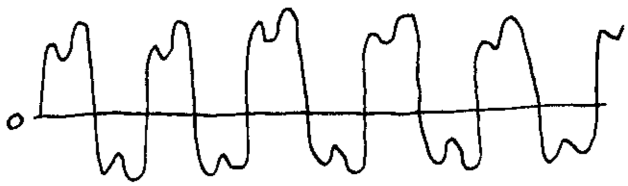
Notice that in this case the amplitude of the two waves is the same, but that in the same length of time there are twice as many excursions up and down in the bottom wave as in the top -- that is the bottom wave has twice the frequency of the top wave. The bottom wave will sound ONE OCTAVE HIGHER than the upper wave. If a wave has twice the frequency of another wave, we hear it as one octave higher. Notice that if the first octave starts out at 80 cycles per second (or 80 Hertz which means the same thing), then the next octave starts at 160 Hertz (twice the first), the third will start at 320 Hertz, the next at 640 Hertz, then 1280, 2560, and 5120 Hertz. Whereas the first octave had a range of only 160 cycles per second, the top octave had a range of 2560 cycles per second! But to our ear/brain both sound like a single octave.

Below are two waves:



These are two wave types that you will find on most synthesizers; the top one being a SAWTOOTH wave and the bottom being a SINE wave. The two waves in this drawing both have the same frequency and the same amplitude but a different SHAPE. The shape of a wave affects its TIMBRE or sound quality. Picture your eardrum being pulled in and out by the two waves shown above to see the difference in the kind of motion the liquid in the cochlea would have. In the real world, of course, nothing can vibrate in quite these shapes and if it could, the air cannot ripple in quite this fashion and if it could the eardrum cannot be moved in precisely this way. But it can all come remarkably close.

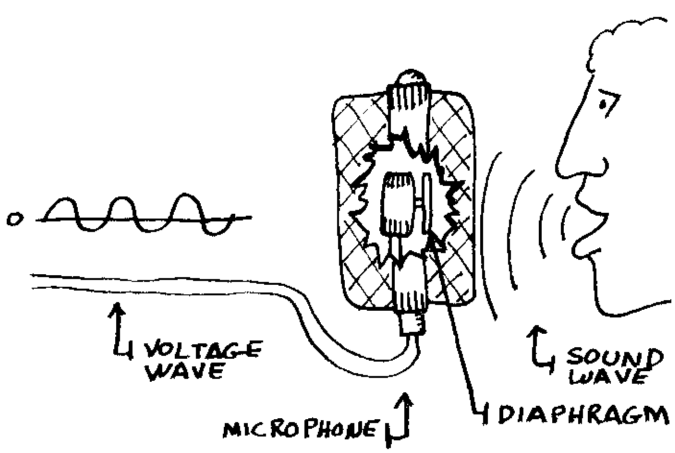
Below is what a guitar sound wave might look like:



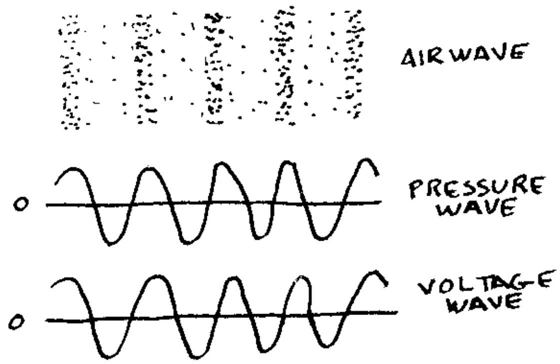
VOLTAGE

Voltage can be considered to be electric pressure. By the middle of the 19th Century, many of the advantages of converting sound waves (rapidly changing atmosperic pressure) into voltage were discerned. Primary among them was that while sound waves died out relatively rapidly, voltage waves could be sent thousands of miles over wires, around corners and through walls,. The main problem was how to convert sound waves into voltage waves and then, after a journey of perhaps a hundred miles, convert the voltage waves more or less accurately, back into sound waves. In other words, the problem was the invention of the telephone.

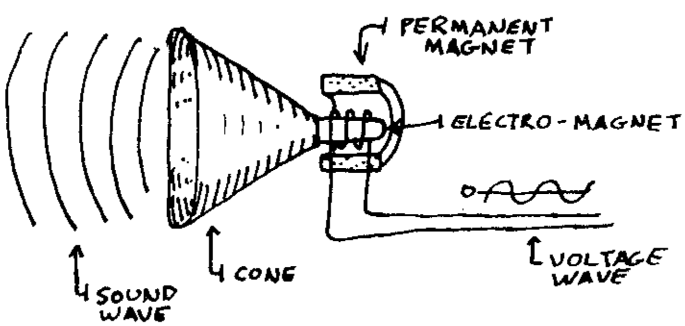
A Microphone is a device for converting sound waves into voltage waves, or atmospheric pressure into electric pressure. The simplest microphones have a diaphragm which acts much like the eardrum in its response to sound waves. It is pushed inwards by a compression wave and pulled outward by a rarefaction wave. This diaphragm is attached to a device which, when it is pushed inward creates a Positive Voltage and when it is pulled outward creates a Negative voltage. When the diaphragm is at rest, its output is Ground-- or 0 volts.



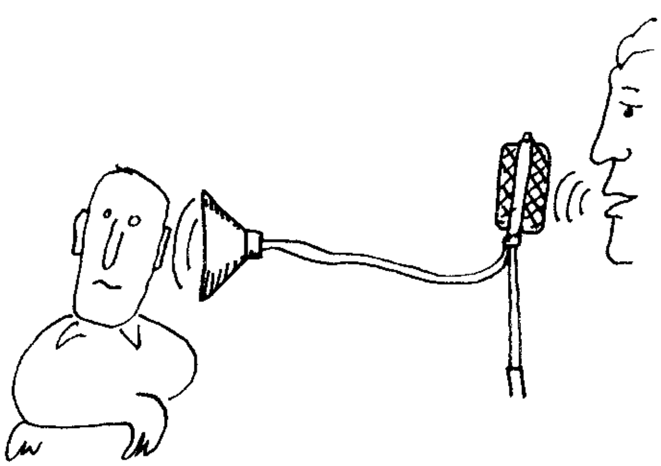
Because of this one-to-one correspondence the voltage output of a microphone is said to be isomorphic with the sound wave input.



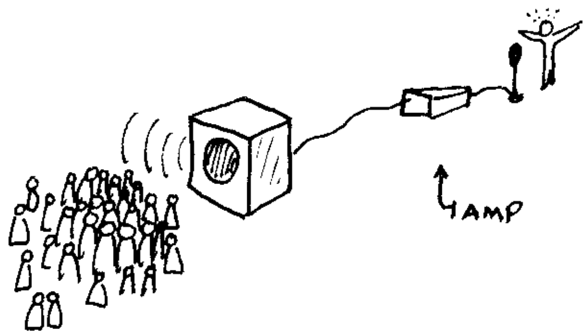
A Speaker is a device that takes a voltage wave and converts it into a sound wave. Though there are many kinds of speakers the most common ones work by moving a cardboard speaker "cone" with an electro-magnet.



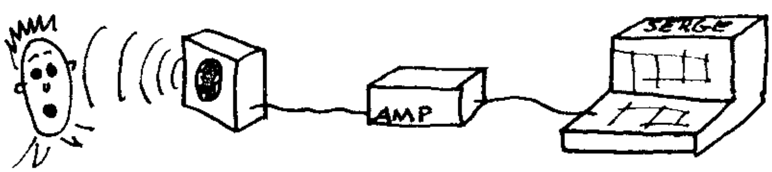
In this kind of speaker the coil of wire attached to the speaker cone sets up magnetic fields which push and pull itself in and out from the permanent magnet as the voltage changes, thereby pushing and pulling the cone in and out. This creates rarefaction and compression waves in front of the cone. The speaker cone, therefore, reproduces the movement of the diaphragm of the microphone and in so doing reproduces the original sound wave.



It was the ability of such a system to transmit sound over long distances that first attracted attention. It soon became clear that there were other advantages. Once the sound wave was converted into a voltage wave, it was far more malleable. It could be amplified, for instance, so that when the speaker re-created the sound it could be louder than the sound originally picked up by the microphone.

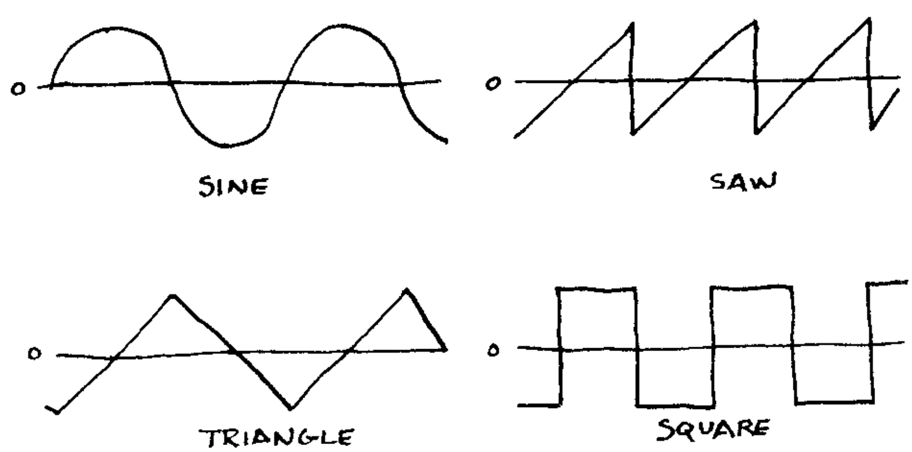


A speaker doesn't know where the voltages it is receiving are coming from. Its cone will move in response to any varying voltage. A SYNTHESIZER is a device which creates and sculpts voltages of various shapes that, when directed to a speaker, create sound that can be used in musical settings.



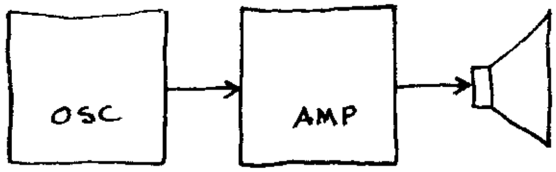
THE DEVELOPMENT OF THE SYNTHESIZER

From the earliest days of electronics, there have been various devices to create and alter voltages of audio frequency. We've already discussed the amplifier which takes an input of a varying voltage and puts out that same varying voltage magnified in amplitude. Another device is the oscillator, which simply puts out a varying voltage in a number of simple shapes:

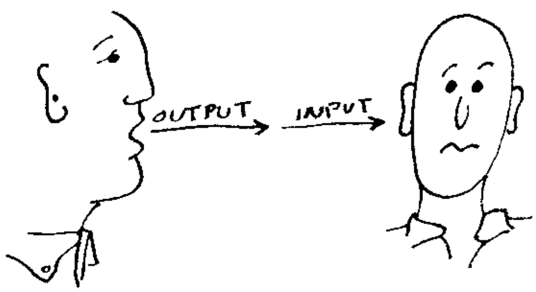


A knob, or POT (short for POTentiometer, which is the device the knob turns) on the front of the oscillator would determine the frequency of these waves, that is, how often in one second the wave would rise and fall.

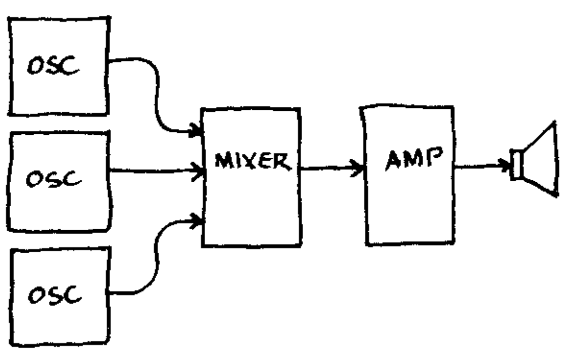
The first step towards electronic music was taken when the OUTPUT of the oscillator was connected, or PATCHED to the INPUT of the amplifier. The OUTPUT of the amplifier was sent to the speaker.



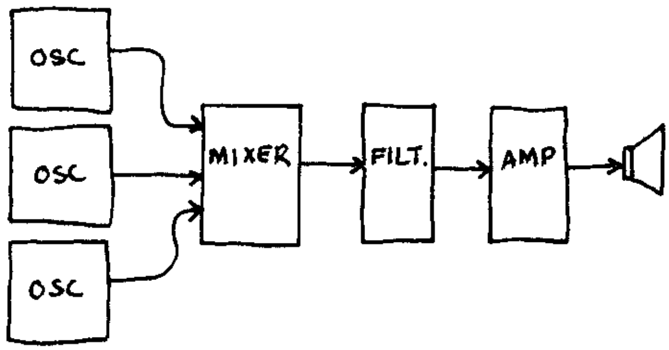
Note the "block diagram" used above. In this form of notation a block indicates an electronic device. The arrow coming out of a device is its output, while an arrow going into a device is its input. An output of one device is always the input to another device. The output of a speaker goes to the input of your ear. What outputs from your mouth inputs into someone else's ear.



Another device was the Mixer, which takes inputs and adds them together to produce a single output. Unlike the amplifier the mixer has more than one input.



Still another important device was the Filter. A filter is a device that can eliminate or accentuate various frequency components of a complex sound. For instance it can be used to eliminate all the very high components (the hiss) in a sound, by only allowing those frequencies in the range of the human voice to pass. A pot on the front of the filter controls which frequencies will be attenuated or eliminated.



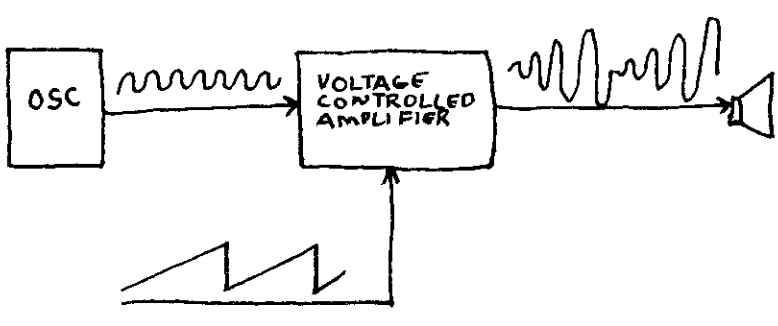
There were two problems with this procedure of adding device after device. The first was that very quickly there were just physically too many knobs to twiddle. The second problem was that the knobs couldn't be turned quickly or precisely enough. The amplifier could not be turned up and then quickly down again fast enough to make the "sound envelope" of a single whack of a drum.

The invention of the tape recorder, just after World War II, solved some of these problems. A single sound could be produced electronically, recorded on to a short piece of tape, and spliced onto another previously made sound and so on until a string of sounds had been made. Two of these tapes could be mixed together through a mixer and recorded on a third tape. The speed of the tape machines could be varied, and the segments could be reversed or even cut to form spliced "envelopes". This was (and still is) a very tedious process, but it is a very rich and flexible one. A studio built to be able to produce electronic tapes in this way is called a Classical Electronic music studio.

The first major improvement in the classical studio came from Columbia University where they devised a controller which could set all the dials instantaneously from the instructions given on a punched paper tape.

It wasn't until the Sixties that the synthesizer as we now know it was designed by Don Buchla and Robert Moog by adding Voltage Control to the classical studio.

The Voltage Controlled Electronic Music Synthesizer solved both of the two major problems of the classical studio by employing Voltage Control which works in the following manner: Each device is given a special input called a Voltage Control Input. This input accepts a voltage such that as this voltage INCREASES it is JUST LIKE TURNING UP THE KNOB ON THE FRONT OF THE DEVICE. And when the voltage goes down it is like turning down the pot on the front of the device. That is, a voltage can be used to CONTROL the device. For instance, in a voltage controlled amplifier, if the voltage at the voltage control input increases, it turns the amplifier up and makes its output louder.

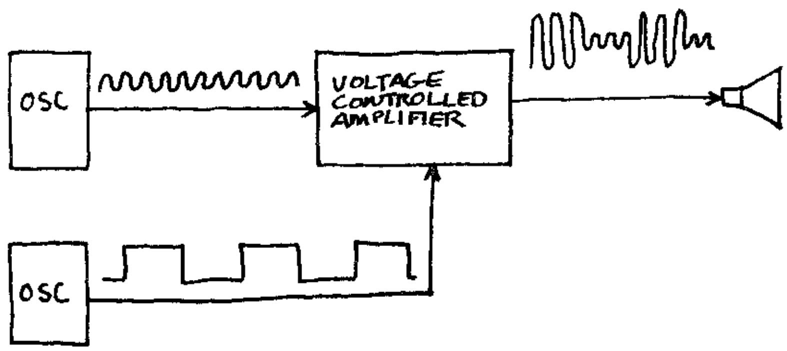


(Note that in these block diagrams, as a matter of convention, the control voltage input is on the bottom of the device, the "signal" input is on the left side and the output is on the right side.)

In a voltage controlled oscillator, a rising voltage at its voltage control input would make its frequency rise. For each device the control voltage affects only the function of that device.

Control voltages solved both problems of the classical music studio: With enough control voltages you could change all the settings of all your devices. And secondly you could change these settings so rapidly as to seem instantaneous. You could change the settings very, very slowly, or you could change them at audio frequencies, for instance 500 times per second. When a devices's settings are changed at those rates, some very strange things begin to happen, many of which can be musical.

The only problem left, of course, is where to get all these control voltages. This problem is not as great as it seems for a control voltage is identical to any other kind of voltage. For instance, we could use an oscillator to control an amplifier since the output of an oscillator is a voltage!



In the above example Oscillator #2 is controlling the amplifier, making the signal from Oscillator #1 louder and softer.

Most of the early synthesizers have two different sets of patch cords, one for the control voltages and one for the signals, even though the voltages themselves are indistinguishable. The Serge System does not make this distinction.

THE SERGE SYSTEM

The SERGE SYNTHESIZER is a Voltage Controlled Modular Music Synthesizer. By MODULAR it is meant that it is composed of separate devices or modules which must be patched together to produce a complex sound. By Voltage Controlled is meant that almost all of these devices can be controlled by a voltage as well as by their own pots. By music is meant that the Serge can be used to create complex, ordered sound, and by Synthesizer is meant that it needs no other input (though it is able to accept one) and that it can create, or synthesize, sound.

There are Four basic kinds of Modules on the Serge. Many modules can serve more than one of these functions:

SOUND SOURCES. The basic sound source is the oscillator though there are others such as white noise. Sounds from the external world, so long as they have been converted into appropriate voltages (by the use of microphones or pickups) can also be used as sound sources. Oscillators are completely voltage controllable.

SOUND PROCESSORS. Processors are devices that input one or more signals, operate on these signals, and then output a different but related signal. Mixers, filters, wave shapers, amplifiers are all processors. Almost all of these devices are voltage controllable.

CONTROL VOLTAGE SOURCES. Control voltage sources are devices that are used to create the voltages which are used to control other devices. The keyboard, for instance, puts out a voltage which can be used to control the setting of an oscillator. Other devices are envelope generators, sequencers, sample/hold devices and envelope followers. These devices are voltage controllable themselves, making possible complex levels of control.

CONTROL VOLTAGE PROCESSORS. These devices input a control voltage, operate on it, and output a related but different voltage. Processors and portamentoes are examples of these modules.

Each module on the Serge is surrounded by a border with the name of the device at the top and the Serge logo at the bottom. In some cases there is more than one device in a module and these are referred to as "dual" or "triple" modules. These dual or triple modules are two or three completely separate, though functionally identical modules.

Every module has at least one output. Outputs are usually enclosed within a border of their own.

All processor type modules have at least one input as well as an output.

Most modules have control voltage inputs which control the function of the module. These inputs are of two basic types:

PROCESSED INPUTS which have a pot associated with the input jack that can attenuate, amplify and/or invert the control voltage.

UNPROCESSED CONTROL VOLTAGE INPUTS affect the given module in a predetermined way.

Most modules have one or more pots that can control the function of the module without a control voltage. In most modules these pots control the basic setting of the module on which the control voltages operate. These pots are labelled with their function, for instance. GAIN, FREQ, etc.

The jacks on the Serge are color coded.

BLACK JACKS indicate audio or signal voltages.

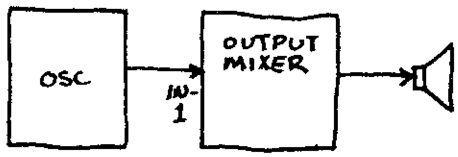
BLUE JACKS indicate control voltages.

RED JACKS are pulse input/output jacks and are used to turn modules on and off, have them step through stages and control the timing of various functions.

OTHER COLOR JACKS are special jacks whose function will be described individually.

LEARNING PATCH NUMBER TWO

STEP ONE

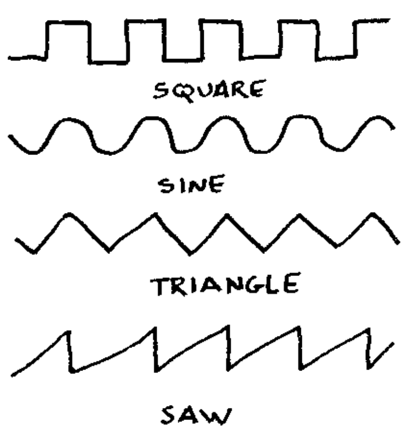


(Note: The output mixer should be set the same as in the First Learning Patch. In this section, block diagrams will be used to represent the patches. Each module is represented as a block. Its signal output is from the right side of the block. Signal inputs are shown going in to the left side of the block. Control voltage inputs go in to the bottom of the block, and control voltage outputs are shown coming off the top of the block. Each of these inputs/outputs will be labelled on the diagram. Any special pot settings necessary to make the patch work will be listed below the diagram. On some of the diagrams drawings of the waveforms will be drawn next to the appropriate patchcord.)

There are two basic oscillators in the Serge system: the New Timbral Oscillator and the Precision VCO. They are identical oscillators except for some control and output functions unique to each. This discussion will concentrate on the New Timbral Oscillator but you can try it with both the oscillators.

1.1 Set up the above Patch on your Serge. The SINE out of the OSC (the abbreviation "OSC" will be used from now on to refer to any oscillator, either a New Timbral Oscillator or Precision Controlled VCO) should be patched to Input #1 of the Output Mixer.

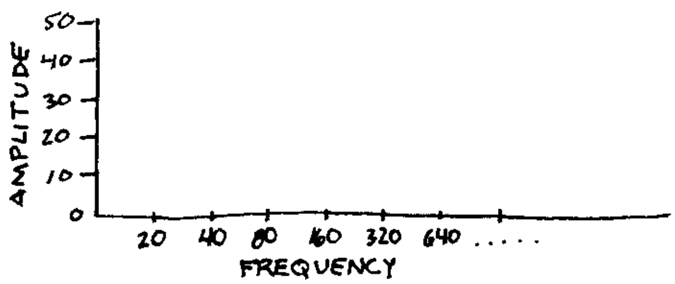
1.2 OSCs produce repetitive varying voltages referred to as "waves". These waves are produced in different "waveshapes" of which SINE, SAW, TRIANGLE and RECTANGULAR are the most common. An OSC can produce these waveshapes at different frequencies. The frequency of a wave determines its pitch. The higher the frequency of a wave the higher its pitch. The shape of a wave determines its Timbre or sound quality. Each OSC on the Serge provides a number of simultaneous outputs, all at the same frequency but with different wave shapes.



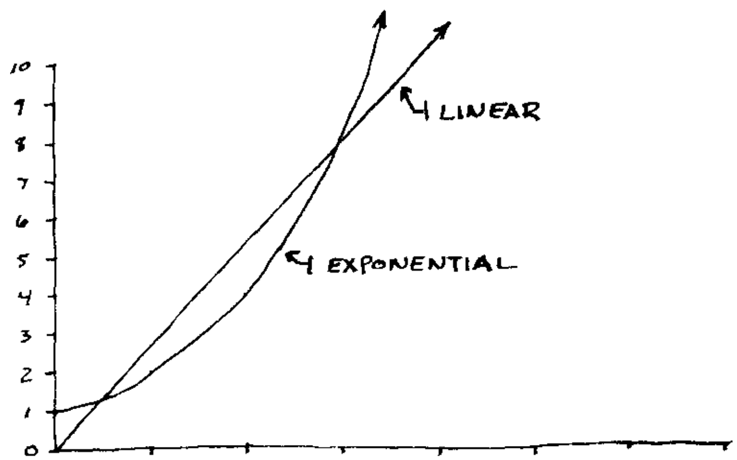
While most OSCs on most synthesizers can produce waves across the entire spectrum of human hearing -- about 20 cycles per second to 20,000 cycles per second-- (cycles per second will be referred to as Hertz), some OSCs on the Serge synthesizer, can go below this threshold. Waves of these low frequencies are useful as control voltages.

1.3 A Sine wave is the simplest form. Any waveform except a perfect SINE wave can be treated as a combination or mix of simpler waveforms. That is, ANY wave can be analysed as a mix of Sine waves of specific frequencies and amplitudes.

1.4 One way of visualizing this is with a chart that has the audible frequencies across the bottom and amplitude on the verticle axis:



Note that the scale across the bottom is EXPONENTIAL; that is, each interval marked off is TWICE the frequency of the previous interval even though the intervals are of equal lengths. This is the way we hear, with each octave having twice the frequency spread of the previous octave (e.g. 20,40,80,160,320,640...) and yet these intervals sound identical to our ears/brains. The Exponential scale contrasts with a LINEAR scale where each interval is a set distance from the previous interval. For instance a linear scale would procede 20,40,60,80,100,120,140.



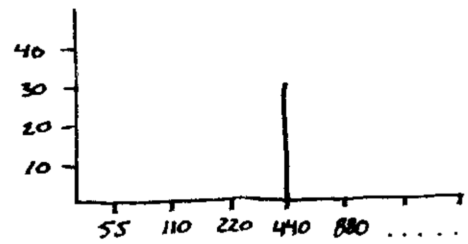
$X; 2X; 3X; 4X, 5X; 6X; 7X;$ etc.

LINEAR TERMS

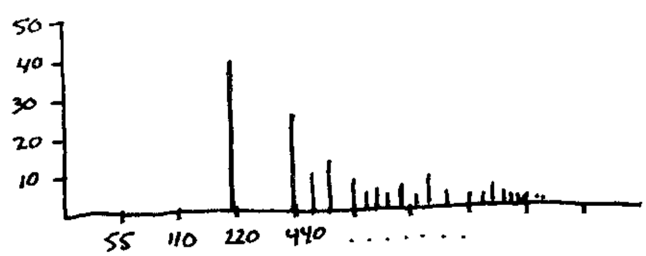
$X; X^2; X^3; X^4; X^5; X^6; X^7;$ etc.

EXPONENTIAL TERMS

To notate a sound on this chart, place a vertical line at the point where each component Sine wave occurs. The height of the line will indicate the relative amplitude of the Sine wave. This vertical scale is also exponential and is measured in Decibels. Though our actual perception of loudness is not quite this simple (we are less sensitive, for instance, to frequencies at the top and bottom of the scale), generally speaking, the higher the Decibels the louder the sound. For instance a pure Sine wave with the frequency of 440 would be shown like this:

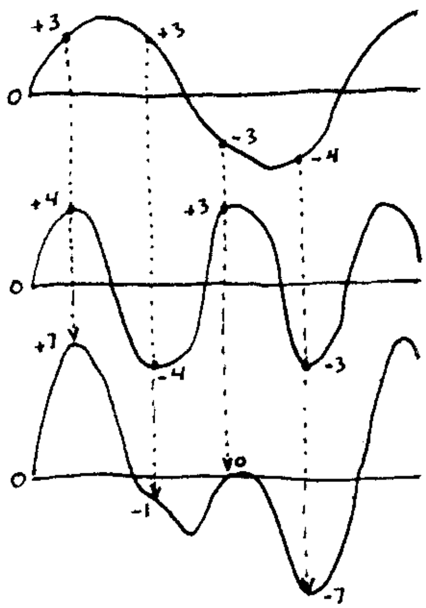


1.5 Most sounds, including electronic sounds, are composed of more than one sine wave. We now have two ways of picturing a sound,: its pressure or voltage wave and its sine-wave spectrum. The "shape" of a wave refers to its voltage as can be seen on an oscilliscope. This is called a time-domain dsiplay. The spectrum graph is called a frequency-domain graph and is an analysis of the voltage waveform. Below is a wave and its hypothetical frequency-domain spectrum.

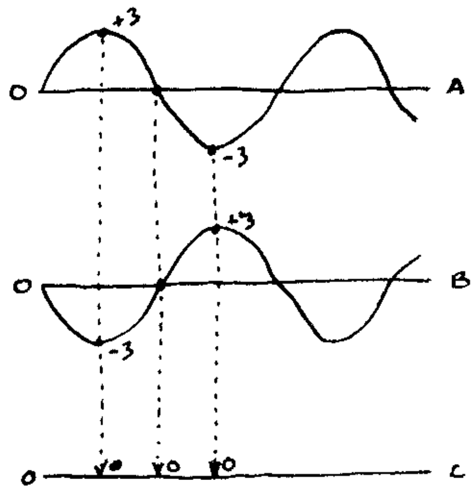


1.6 To determine the overall shape of a wave from its component sine waves, the values of the component waves AT EACH INSTANT are added together.

This also means that if two waves of identical frequency but of opposite "phase" (one goes up while the other goes down) are mixed together, silence will result.

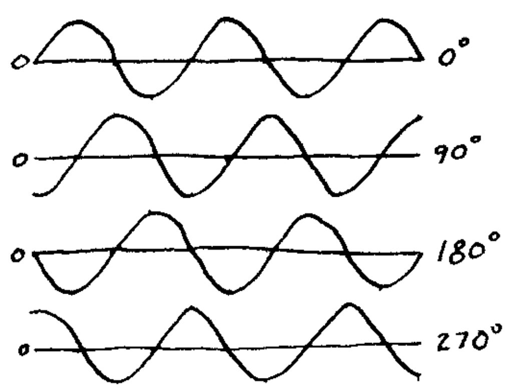


A+B=C



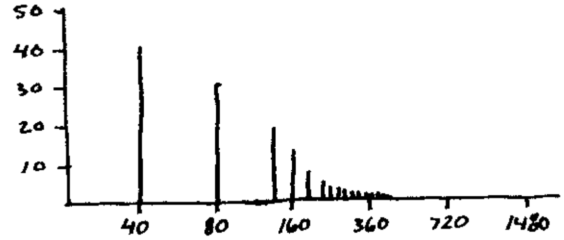
A+B=C

Phase is noted in degrees where 360 degrees brings a wave right back to where it started.



1.7 In theory, any sound can be created by adding together sine waves of the correct amplitude, frequency and phase. This is called "additive" synthesis or "Fourier" synthesis. This technique is of limited use in the synthesizer because the number of sine waves would have to be tremendous.

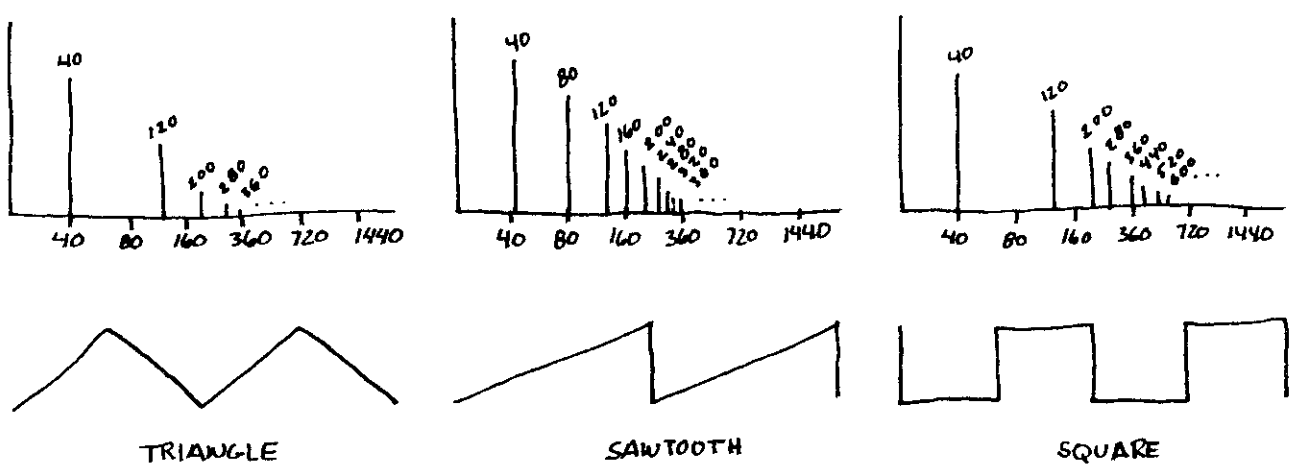
1.8 Another reason that this technique is not often used is that most sounds, and almost all musical sounds, are composed of sine waves in "harmonic" relationship to a "fundamental". The fundamental usually corresponds to the apparent pitch of a complex sound and is usually the lowest strong sine wave of the sound. If "X" is the fundamental and the other sine waves are in a harmonic relationship to it, then there is a sine wave at 2X,3X,4X,5X...etc. These sine waves are called "overtones" and they generally decrease in amplitude as they get higher in pitch. Below is the spectrum of a typical acoustic musical instrument such as a guitar:



Note that the overtones seem to be getting closer and closer together on the spectrum chart the further they get from the fundamental. We hear them in this fashion. Remember that the audio spectrum as we perceive it is exponential, but the overtone, or harmonic series, is linear!

To calculate the positions of the harmonics add the fundamental frequency to itself to get the first overtone; add it in again to the total to get the second harmonic, again to get the third and so on. An example would be: First: 100; Second: 100+100; Third 100+100+100; Fourth:100+100+100+100; etc. Thus it can be seen that the frequencies are spaced at equal, absolute spacings, that is the harmonics fall on a Linear graph.

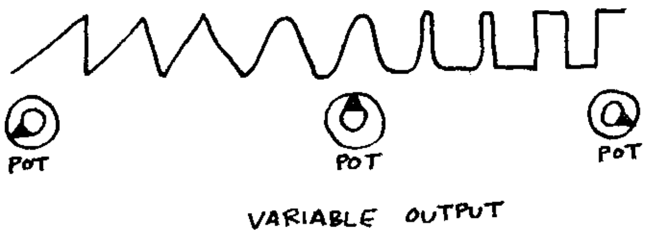
1.9 Besides the Sine wave the New Timbral Oscillator also has a Sawtooth output, a Triangle output and a Variable waveform output that can put out a Square wave, or other waveforms. (The Precision VCO has all these outputs except for the Variable waveform.) Below are the voltage or pressure diagrams of these waves and the spectrum charts of these waves.



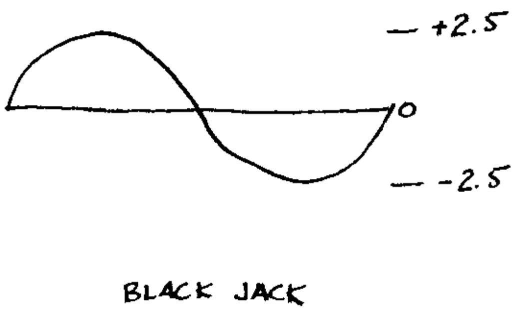
Additive synthesis can be greatly simplified by using these more complex sounds since these waves will often contain the desired harmonics.

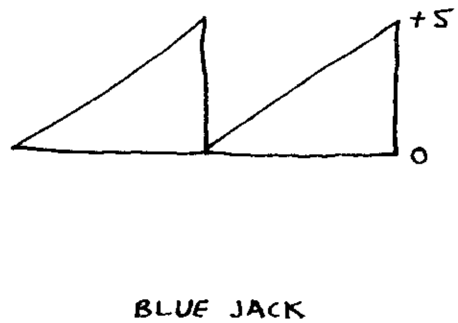
1.10 The Triangle and Square wave contain only the odd harmonics (harmonic #1,#3,#5,#7, etc), although the amplitude of these harmonics decreases more rapidly in the Triangle than in the Square. The Sawtooth wave contains both even and odd harmonics that decrease at about the same rate as in the Square wave.

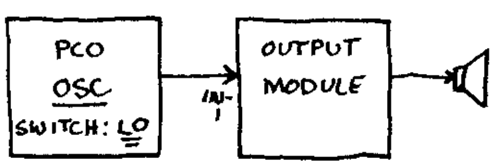
1.11 Try the different outputs of the OSC including the Variable output. The pot directly below the Variable output adjusts the shape and therefore the timbre.



1.12 Though all the outputs of the OSC are of the same amplitude, the saw and the square wave may seem louder because our ears tend to hear complex sounds as louder than pure ones. All waveforms from the oscillators have a 4 to 5 volt peak-to-peak voltage. The Sine output is from +2.5 to -2.5 volts. Black jack outputs typically have this voltage range. The other outputs of the osc. have a voltage range of 0 to 5 volts (still a 5 volt over-all amplitude).

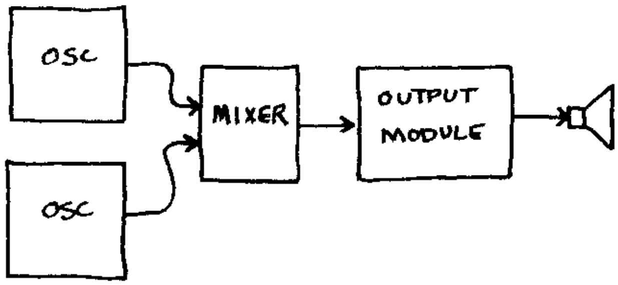






1.13 This patch uses the Precision Controlled VCO with the HI/LO switch at its LO setting to change the range of the oscillator from a range of 20 to 20,000 Hertz (audio range) to a range of .01 to 500 Hertz. Start with the Pitch pot at its furthest right position and begin slowly moving it to the left. The pitch should get lower and lower until a series of clicks appears simultaneous with the sound. The further left you go the more the pitch drops away until you are left with only clicks. Because hearing only goes down to about 20 Hertz you can no longer hear the frequency as a pitch, but the sharp edge of the sawtooth wave pulls back the speaker cone each time producing the "click". If you try the same thing with a sine wave, you will hear nothing, for there are no sharp "edges" on a sine wave. But if you can see your speaker, you will note that the cone is still moving in and out silently and slowly, responding to the changing voltage. Most audio amplifiers can only go down to a certain frequency after which there will be no motion in the speakers at all.

STEP TWO



2.1 Patch the Sine output of two OSCs to two of the inputs of the upper mixer of a Dual 3-input Audio Mixer. Note that this module is a dual module and that the top mixer is totally separate from the bottom mixer. Patch the output of this mixer to the Output Mixer (QCA, UPAP, etc.). Each of the inputs of the mixer has a Pot associated with it that can limit, or "attenuate" the gain of its input. The output of this module is the summation of all its inputs at their assigned gain.

2.2 Tune the two OSCs so that they are very close in pitch and set their gains so that they are at the same level. When they are exactly the same pitch, they should sound like a single sound. If they are a few Hertz apart, you will be able to hear a "beating" between them. The frequency of this beating is the difference in frequency between the two sine waves.

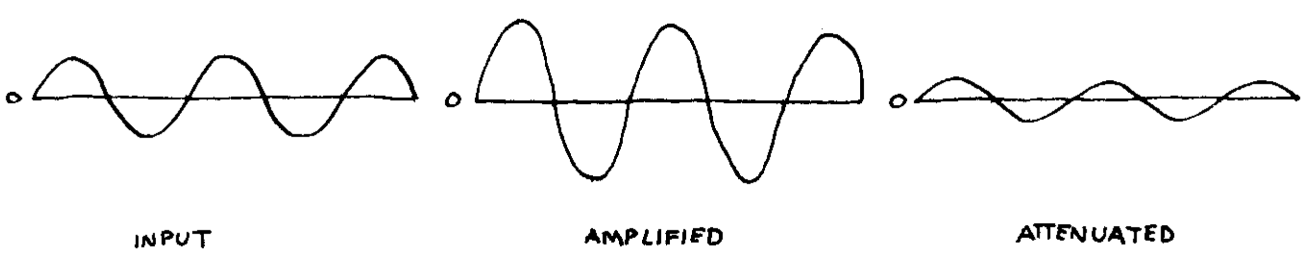
2.3 Try adding a third Sine wave to make a tri-tone.

2.4 Unpatch all but one of the OSCs. Turn up its mixer pot and note that the ear hears the sound as unchanging EXCEPT that it gets louder and louder. This is comparable to the way the eye sees a photograph and its blow-up as identical only the blow-up is larger.

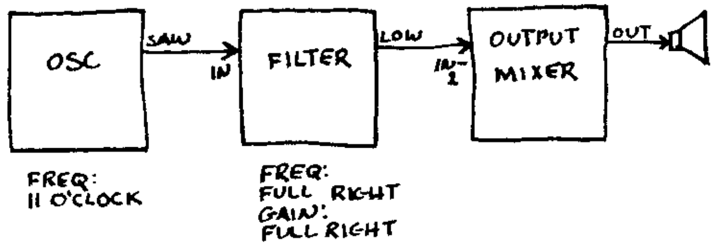
2.5 Very few sounds in the world have a steady amplitude or gain. How a sound's gain changes is one of the clues as to what is vibrating. It is one of the components of the over-all feel of a sound. For instance, a piano note gets very loud very quickly when struck, then slowly gets softer and softer. If the way this amplitude changes were altered, we would not easily recognize it as a piano sound. This amplitude shape is called the ENVELOPE of a sound, because like a letter in an envelope, the sonic information is contained within it.

2.6 Because of the way our ear/brains process sound, the amplitude of a sound must increase exponentially in order for us to perceive it as linearly increasing. For this reason the pots on the mixer are logarithmic.

2.7 When the pot of the associated input is turned to the right, the sound increases in level. Turning it to the left will cause the sound level to decrease. The shape of the wave and its frequency remain the same except for this change in amplitude:



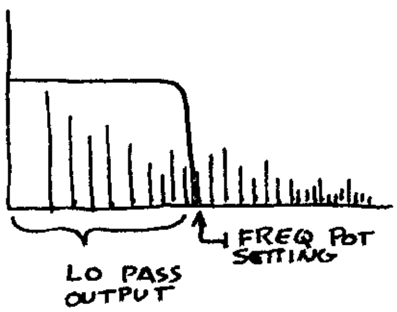
STEP THREE



3.1 A FILTER is a module which makes it possible to eliminate certain components of a sound, depending on its frequency. As we said earlier, every sound can be thought of as the summation of a number of sine waves, each with a different frequency. The Filter allows us to listen to those Sine waves in a sound which fall above, below or directly around a Frequency set by the Pot labelled "FREQ" on the filter.

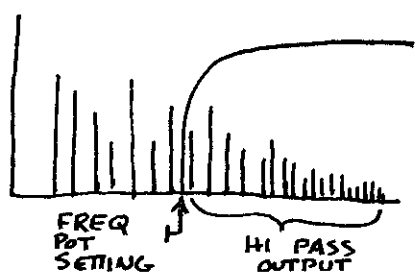
3.2 While there are a number of different outputs on the filter, all outputs can be thought of as different combinations of HI pass and LO pass outputs.

3.3 A LO PASS filter lets PASS through to the output all those sine wave components in the input sound which are LOWER than the Frequency set by the FREQ Pot.

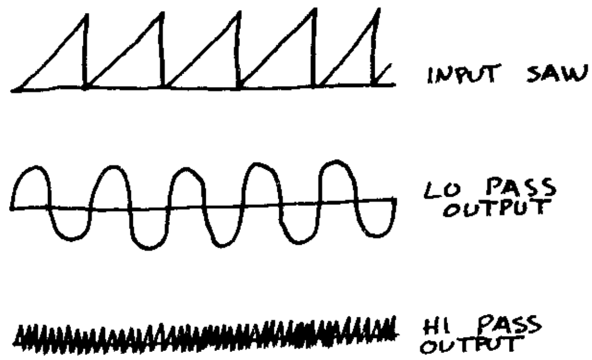


3.4 Slowly turn the FREQ. Pot to the left and the "hissy" sounds will start to disappear. As turning of the pot continues the mid-range will disappear, and finally there will be nothing left but a very low sound, the fundamental, of the oscillator. If the FREQ pot is turned even further, it will eliminate this sine wave as well, leaving no sound.

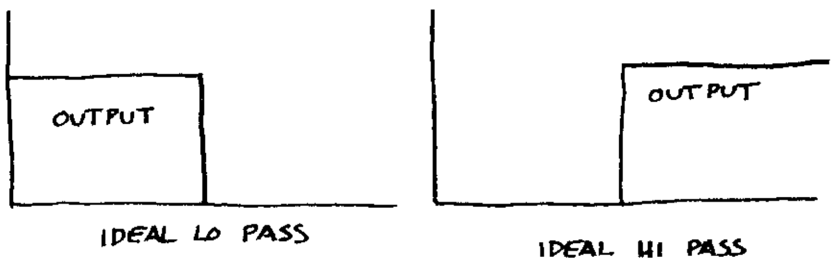
3.5 Re-patch the above patch using the HI PASS output. Now the filter lets PASS only those sounds which are higher than the frequency set by the FREQ POT. It lets pass to the output only the High frequencies of the input sound. Starting with the FREQ pot full left and slowly turning it right, the fundamental will drop out and then the mid-range. ONLY the hiss, or very top part of the spectrum, will be left of the input sound.



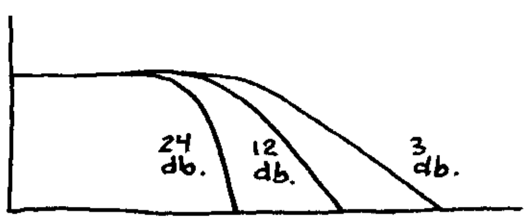
3.6 In terms of waveshape the LO pass filter SMOOTHS out a wave. It finds those components which change the least. Mathematically, it can be said to take the integral of the wave. A HI PASS filter takes the derivative of a wave. That is, the HI pass filter finds those parts of the wave which change the fastest. Below are some typical waveform outputs from HI and LO pass filters:



3.7 An "ideal" filter would not allow any sounds HIgher or LOwer than its cut-off frequency to Pass. It would look like this on a spectrum chart:



But all filters fall short of these ideals, not only because no technology is perfect but because such filters do not produce very musical sounds. The cut-off sharpness is measured in db/oct with 0 db/oct being no cut-off at all and 60 db/oct being about as sharp as we can hear. Most synthesizer filters are in the 3 to 24 db/oct range. The Variable Q Filter has a 12 db/oct cut-off. The Variable Slope VCF's cut-off can be varied from 0 to 12 db/oct using the second POT below the input labelled VC SLOPE.

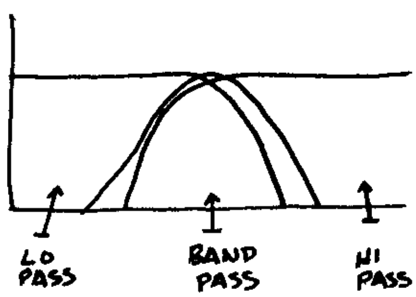


LO PASS CUT-OFFS

Another phenomenon of filtering available on the Serge is called the "Q". Most filters tend to amplify the frequencies near the the cut-off. The more these frequencies are amplified, the higher the Q of the filter. In most cases, the higher the Q, the sharper the cut-off. Knocking on the table is a typical low Q sound from the natural world. A drum head has a medium Q and a bell has a high Q.

3.8 On the Variable Q VCF the Q can be adjusted by using the POT just below the VCQ label. On the Variable Slope VCF when the slope is set so that it is very sharp (full right) the Q is very high. Using a very HI Q it is possible to "scan" through the overtones of a sound by slowly turning the FREQ Pot of the filter. Everytime the Freq. is the same as an overtone it will amplify that overtone.

3.9 The BAND output of a FILTER filters out everything but a area around the Frequency set by the Freq pot. It is useful for listening to a single part of a more complex sound. Below is a diagram of how HI, LO and BAND Pass are related to each other.

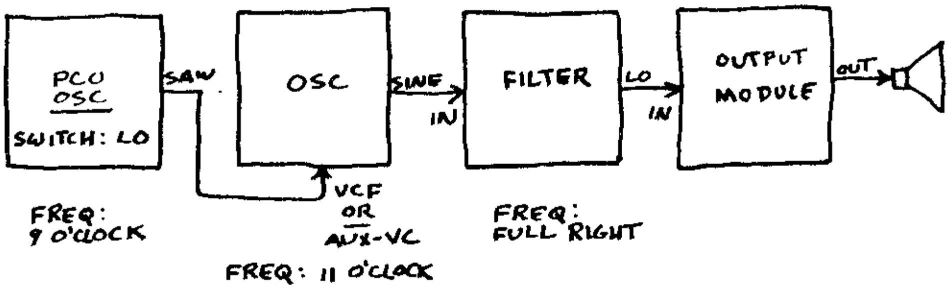


3.10 While HI pass filtering occurs only rarely in nature (a cheap transistor radio tends to be a hi-pass filter to music by cutting out the lows), LO pass filtering abounds. In many musical instruments, a piano for instance, once the string is struck the highs tend to be filtered out leaving only the lows-- the typical action of a LO pass filter. The human mouth is also a LO pass filter and is responsible for our vowel sounds, which again are LO pass filter sounds.

3.11 The GAIN pot on the Variable Q VCF controls the level of the signal input exactly like the POT on a mixer. It must be turned up to hear any output. If the Q of the filter is set high, the GAIN should usually be turned down so that when the FREQ of the filter and the frequency of an overtone coincide, the filter is not overdriven. (Sometimes this is the desired effect. Even though the filter will overload, no damage will be done.)

3.12 The Variable Slope VCF has two independent inputs which can be manually "cross-faded" or mixed together using the MIX pot. If IN-1 is used as the input jack, be sure that the MIX pot is set to the left; vice versa for IN-2.

STEP FOUR



STEPS One, Two and Three could have been been set up in a classical music studio. STEP FOUR begins the exploration of Voltage Control, a technique which extends electronic synthesis to its modern form.

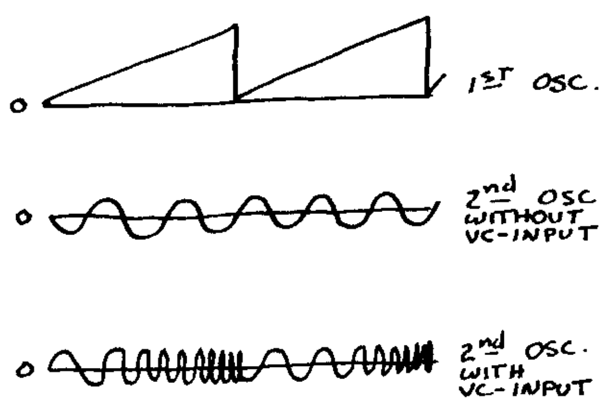
4.1 Patch the SAW output of a Precision VCO to the VC-AUX or VCF input of a second OSC, either a PCO or NTO. The PCO is used as the first OSC because it has a range switch allowing it to oscillate at very low frequencies. This range switch should be set to LO.

A SAW wave is a voltage which rises from 0 to 5 volts and then swiftly drops back to 0 volts. It does this over and over again.

When an OSC is voltage controlled it is like TURNING its FREQ pot by REMOTE CONTROL. When this controlling frequency is rising it is exactly like turning the FREQ pot to the right. When the controlling voltage falls, it is like turning the FREQ pot to the left.

4.2 Turn the VC AUX or the VCF POT on the second OSC full right and the GAIN up on the audio mixer until the sweeping sounds of the oscillator can be heard. The sound will rise higher and higher and suddenly fall back to a very low sound only to begin rising again. The pitch is produced by the second oscillator. The first oscillator's SAWtooth wave is causing it to rise and then swiftly fall.

This is a stylised picture of the pressure wave being produced:



If the TRIANGLE or SINE output of the first OSC are used instead of the SAW, the following waveshapes are produced. These can be heard as different patterns of rising and falling pitches.

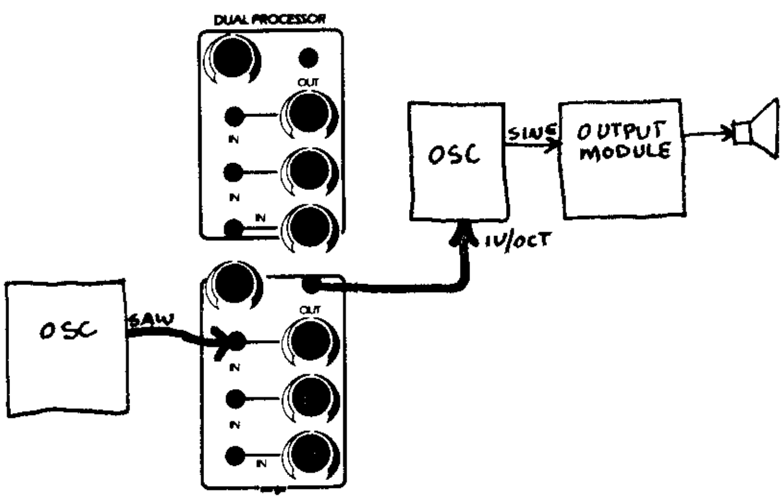
SINE WAVE MODULATING SINE WAVE

TRIANGLE WAVE MODULATING SINE WAVE

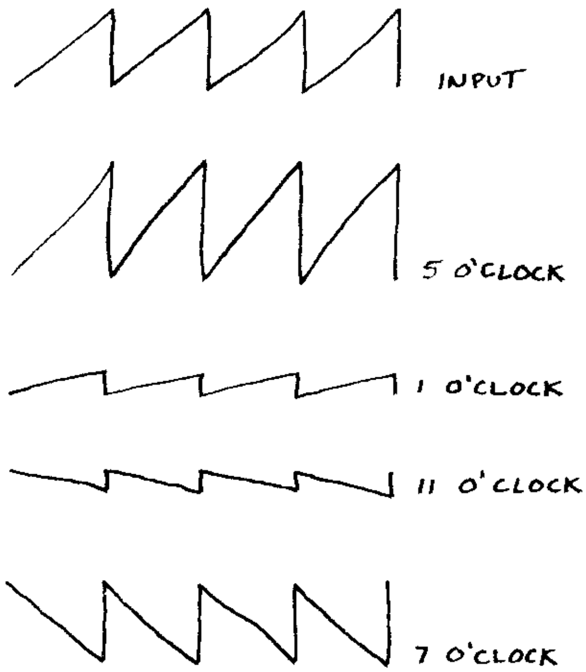
4.3 Increase the first oscillator's frequency slowly, listening carefully to the results. At first the sweeping will get faster and faster until a frequency approaching 20 Hertz is reached, at which point the sound takes on a multi-harmonic quality. This is called Frequency Modulation, or FM, because the frequency of the second oscillator is being changed or "modulated" at a rapid rate by the first. FM is a major technique of audio synthesis.

4.4 Set the FREQ of the first OSC so that the sweep of the second takes a few seconds. As the Pot associated with the control voltage input on the second OSC is moved from its full right position to a 12 o'clock setting, the sweeps will become shallower and shallower, although the time they take remains the same. As this Pot is turned to the left, the sweeps will have a greater and greater gain but an inverted one. Whereas a Pot set to the right causes the sweep to go upward and then suddenly fall downward, when it is set to the left the sweep is downward and the jump up. Control voltage inputs that have Pots of this type associated with them are called "Processed Inputs".

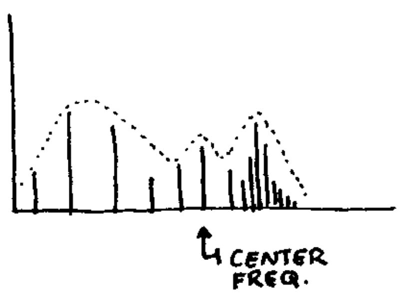
These Pots control a device internal to the module which can amplify, attenuate and/or invert a control voltage input. It is because of their extreme usefulness that they are they are the typical control voltage inputs of the Serge. The Serge also has Processor Modules which can be patched to serve the same function.



Below are some of the possible outputs of a Processor with a SAW input.



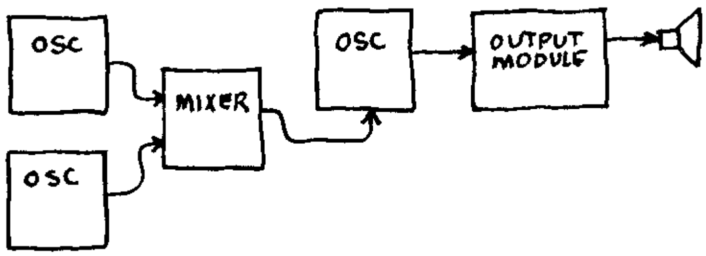
4.5 By setting the first OSC to HI (its range is switched to 20 to 20,000 Hertz, the audio range), an extremely wide range of sounds is possible with different combinations of FREQ and Processing pot settings on the two OSCs. This range can be extended even further by using different waveforms. The first OSC is referred to as the Modulator (or the signal, a term from radio broadcasting); the second OSC is referred to as the Modulated oscillator, or the Carrier. The setting of the processor, which determines the relative gain of the two OSCs, is called the Index. The frequency of the two oscillators and the setting of the Index determine the output of the modulated OSC. While the mathematics of FM is not simple, particularly with waveforms other than Sine waves, in general the spectrum of the output looks something like this:

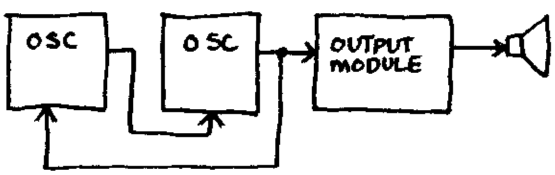


The frequency of the Modulated OSC sets the center frequency. There is an "overtone" or "undertone" every "f" Hertz where "f" is the frequency of the Modulator. The amplitude of these over/under-tones is determined by the Index and the frequencies of the oscillators. The overall shape of the amplitudes is butterfly and is called a Bessel function. In FM, sub-harmonics which would fall below 0 Hertz are "folded back" up to their "absolute" value. If the Modulated OSC is set at 200 Hertz and the Modulating OSC is at 60 Hertz then there should be sub-harmonics at 140, 80, 20, -40 and -100 Hertz. However these will be heard as 140, 100, 80, 40, and 20 Hertz.

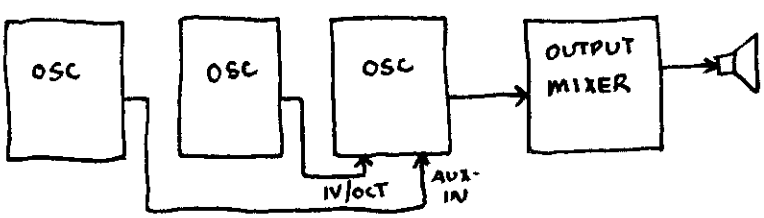
Such mathematical descriptions, while interesting, are not vital to electronic music. Working with the synthesizer is rather like clay sculpture--you can work at the sound until it is right.

Keeping in mind that the output of an OSC, either modulated or unmodulated, is a varying voltage, and that such voltages can be used to control the frequency of other OSCs there are innumerable complex patches available to the synthesist.

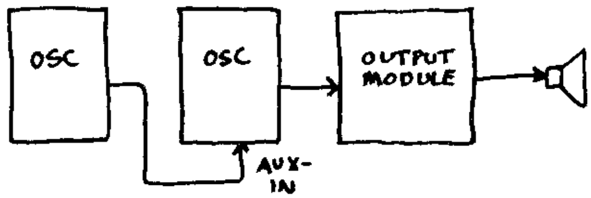




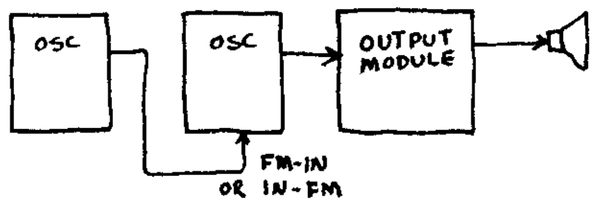
4.6 1V/OCT. The 1V/OCT control voltage on the oscillators is an extremely precise control voltage input whose effect is calibrated with detailed attention. The relationship of input voltage to output frequency is this: For every volt increase at the 1V/OCT input the OSC will rise EXACTLY one octave. One reason that such an input is valuable is that most synthesizer keyboards and other electronic music devices have output voltages that are set exactly to this relationship. Both the NTO and the PCO have two 1V/OCT inputs. (The second 1V/OCT input on the NTO is labelled Portamento In. It has another function associated with the pot and Control Voltage inputs below it. For now the pot should be turned full right.) When two different control voltages are received by an OSC they are added or summed together, after processing, so that both have an effect on the modulated oscillator and yet do not interact with each other.



4.7 FM-IN.



In the above patch, if both OSCs are set to audio frequencies, very interesting shifts in timbre occur when the Processing Pot is turned to different positions. However, when the AUX-IN is used, there are also apparent pitch changes.

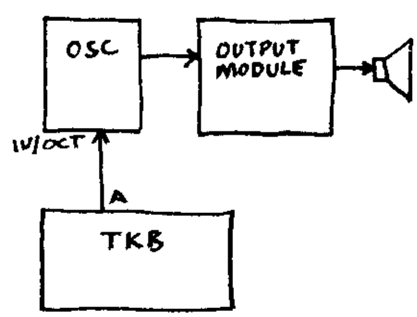


4.8 In this patch if the modulated OSC is a New Timbral Oscillator, then connect the modulating signal to the IN-FM. If the Precision VCO is used, use the FM-IN. Sweeping the associated pots of these inputs sets the Index. The sound produced should be similar to that produced by an audio voltage to the AUX-IN except that the over-all pitch does not seem to change as the INDEX is changed. The FM-IN and the IN-FM signal inputs are LINEAR, that is, equal rises of voltage produce equal increases in cycles per second.

STEP FIVE

All the modules in the previous step could have been found in a classical music studio except for the voltage controlled oscillator (although even it was found in some.) It is a powerful group of modules, with the oscillators providing the basic pitch material, the mixers adding these sounds together and adjusting their volumes, and the filters altering the timbre of the sound. Yet with only these modules many of the simplest sounds and patterns in music could not easily be achieved. In most musics there are discrete pitches whereas with the modules in the last step there were only sliding tones. Secondly it was hard to get non-repeating patterns.

The Touch Activated Keyboard Sequencer Module, or TKB, is a single large module designed specifically to produce control voltages. As discussed earlier, there is no physical or electrical difference between audio and control voltages other than that MOST audio voltages are between -2.5 volts and +2.5 volts, and all audio voltages are between 20 and 20,000 Hertz; while control voltages are between -12 and 0 volts, or 0 and +12, with frequencies anywhere between 0 and 500 Hertz. The actual difference between the two voltages are the uses to which they are put. The same voltage can be used in different ways. In one case it could be an audio voltage, in the other it could be a control voltage. However, some voltages are simply more useful in one situation than the other. The voltages produced by the TKB are designed to be used as control voltages.



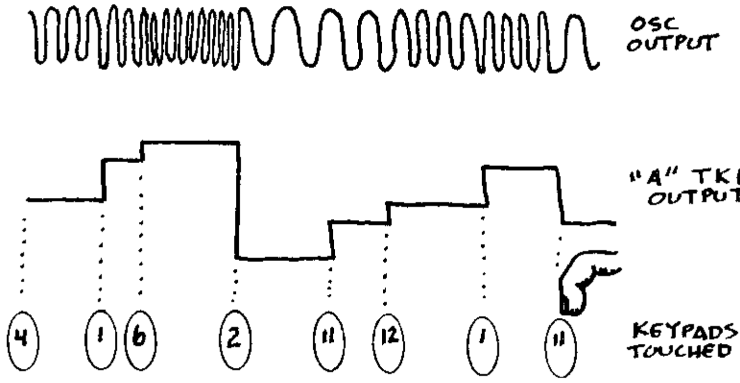
The TKB has four rows of pots across, labelled A,B,C, and D, and one row of keypads. There are 16 columns each with four pots (one from each row) and one keypad. At any given instant ONE and ONLY ONE column is activated and this is indicated by an LED (Light Emitting Diode) on the keypad of the respective column. These columns will be referred to from now on as STAGES.

5.1 The main outputs of the TKB are located at the top left-hand section on the module, enclosed in a border. There are five main voltage outputs (blue jacks) labelled A,B,C,D and ABCD. Patch the A output of the TKB to the 1V/OCT input of the OSC as shown in the above diagram. The OSC should be set to an audio frequency and its output sent to the output modules. Turn KEYS switch on and make sure that no other cords are patched to the TKB.

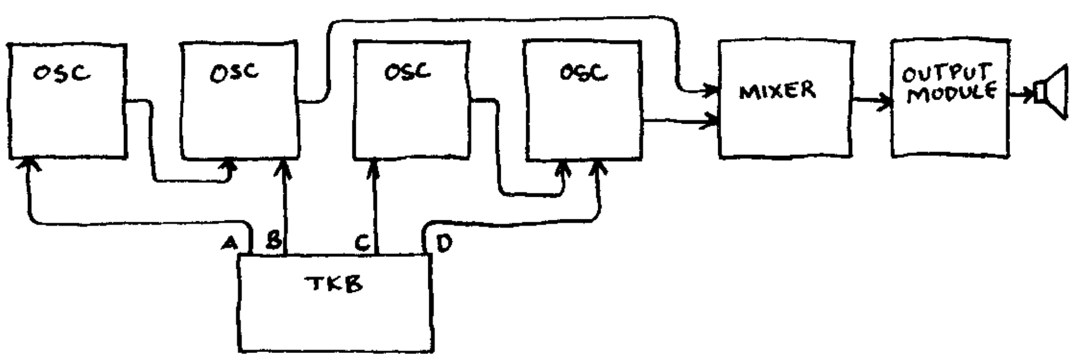
5.2 Touching keypad #1 activates stage #1 which is indicated by the LED that lights on keypad #1. Turn the pot in stage #1 and in row A (the top pot in stage #1) right and left. The OSC's frequency should shift up and down correspondingly. This pot is now remote-controlling the frequency of the OSC using a voltage that is appearing at output A.

5.3 Touch keypad #2 and set its A pot to a different setting then the A pot of stage #1. By alternately tapping keypads #1 and #2 you can get the OSC to produce two different "notes" or pitches without sliding from one to the other. This same procedure can be used to tune all 16 pots in row A. This is the tuneable keyboard.

The output of the TKB is NOT an audio voltage but rather a series of steady, or DC (direct current) voltages which are CONTROLLING the setting of the OSC (or whatever module or parameter the output is patched to). The OSC is designed to respond to these control voltages exactly like it responds to the turning of its pots. Just as the notes on a singer's score do not oscillate, so the voltages from the TKB do not oscillate but merely specify the OSC frequency. Below is a diagram of the voltage outputs of the TKB, and the OSC.

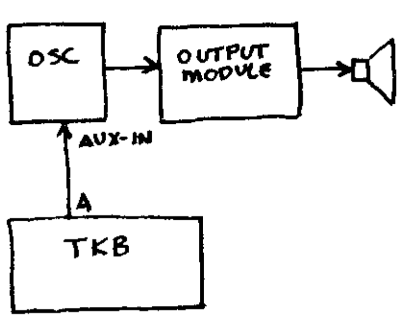


5.4 Patched in this fashion, none of the pots in Rows B,C or D have any effect. However, if it is re-patched so that the output of the TKB is taken from the B output instead of the A output then only the pots in Row B will be active. The same is true for Rows C and D. It is possible to use all four of these outputs (or as many as needed) SIMULTANEOUSLY as in the Patch below:

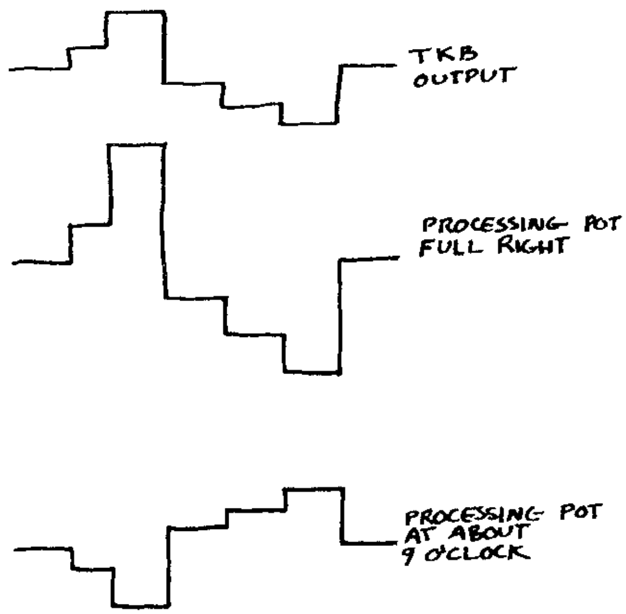


Now at each stage the pot in Row A controls the frequency of the Modulating OSC while the pot in Row B controls the base frequency of the modulated OSC. Row A and B could be replaced by any two rows. By touching the sixteen different keypads and setting the appropriate pots, sixteen different sounds can be set up and recalled in any order at the touch of a finger.

5.5



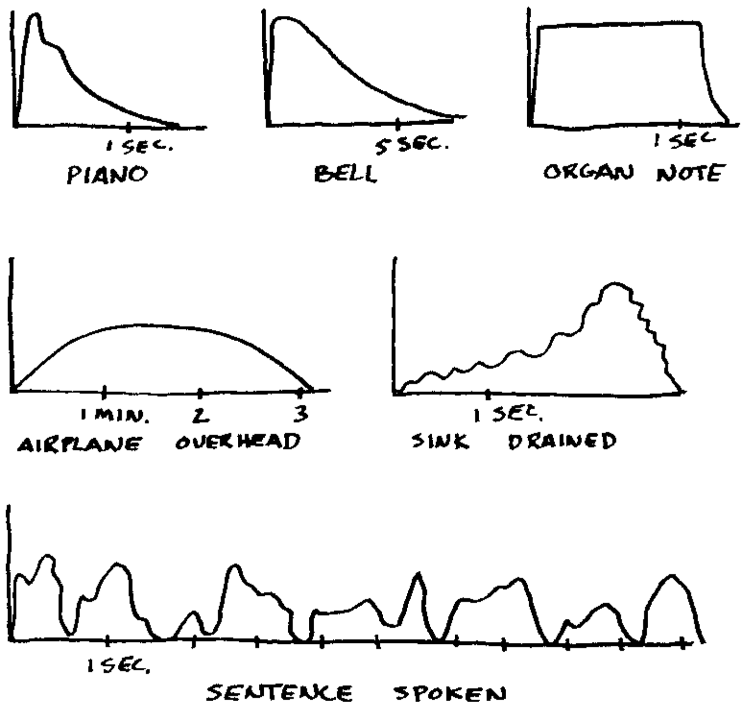
In the above patch the processing pot associated with the AUX-IN processes the incoming voltage from the TKB. Below are some typical processed TKB voltages:



5.6 It is convenient to think of the TKB in this manner: All the pots in each row are tied to a common output (output A for the pots in row A for instance) but only one pot is activated and that is determined by which keypad was last touched. Since there are four rows, four parameters or modules can be controlled in 16 pre-set ways and these pre-sets, or stages, can be accessed directly by the touch pads.

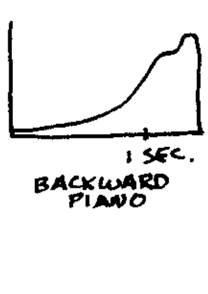
STEP SIX

When a piano note is struck, a bell gonged, a table tapped, an airplane flies overhead, a sentence spoken, a sink drained, an organ note sustained or when any other object makes a sound, that sound has an amplitude shape to it, an "envelope", that grows louder and softer in various ways as time passes.

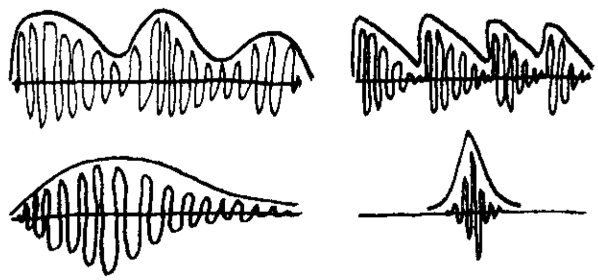


In these charts the loudness of the sounds is measured in db (decibels) while the duration is measured in units of time (seconds, minutes, etc.). The envelope refers only to the loudness of the sound, not to the frequency content of the sound. (We can again compare this "envelope to an envelope which holds a letter and think of the content of the sound as the letter.)

Every sound we hear has an envelope, and this is one of the ways various sounds are distinguished from one another. In this sense the envelope can be thought of as part of the timbre of a sound. Even artificial sounds have envelopes, for instance, this is the envelope of a piano, reversed:



AN ENVELOPE IS THE TRACE OF THE PEAK VOLTAGES OR PRESSURES OF A WAVE.



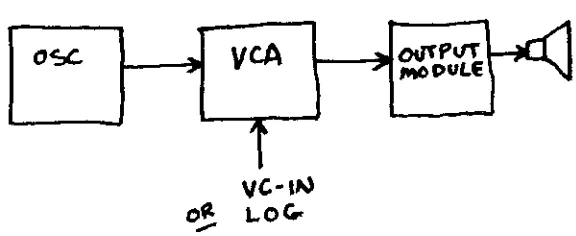
ENVELOPES

The Gain Pots on the mixer, when turned from left to right and then back to the left, can give the input signal an amplitude envelope. The device or module that automates this function is the Voltage Controlled Amplifier (VCA) or Gate and can be found both as an independent module and/or as part of most output modules. These modules are listed in STEP #6 of the first learning patch.

Every VCA has a signal input and a signal output and, like a mixer, has a Pot which can adjust the amplitude of the output in relation to the input without affecting any other parameter of the sound. In addition, the VCA has at least one Voltage Control input. As the control voltage rises, the amplitude of the output increases; as the control voltage falls so does the output amplitude. The control voltage, in effect, turns the Gain pot of the VCA by remote control.

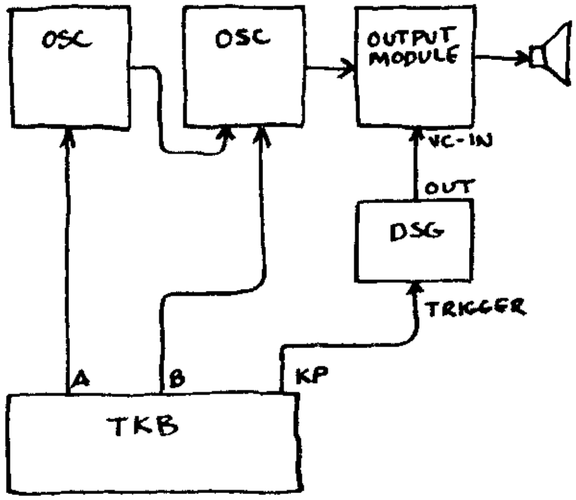
6.1 The Serge system has a wide range of VCAs, but for the purpose of this patch, the VCAs on the output module will be used. The signal input and output remain the same. Each input has an associated VC-in, usually located below or above the signal input (see the section on Output Mixers if your mixer is not mentioned in Learning Patch #1). Since Input #1 is used, VC-input #1 must be used to control its gain. Make sure there are no other patch cords connected to the module. The Gain associated with the input should be set to 10 o'clock. At this time, no sound should be heard from the VCA.

If the system being used has a separate VCA module, that module may be used instead of the VCAs on the output module in the following manner:



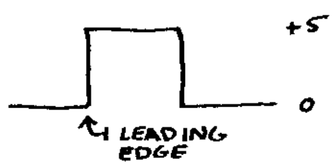
If the VCA is a dual or quad module make sure that the correct inputs, outputs and VC-ins are used.

6.2 Patch this patch:



The Dual Universal Slope Generator contains two identical modules both of which will be referred to as a DSG. In these patches either one can be used.

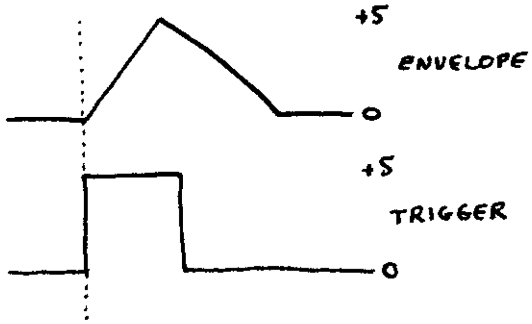
The KP output of the TKB is in the same bank of outputs as the A and B outputs. It has a RED jack which indicates that it is a TRIGGER output which is the third kind of voltage on the system, Audio and Control being the other two. It's function is to either turn something on or turn something off. A Trigger voltage is always either 0 volts (its low state) or +5 volts (its high state). That moment when it goes from 0 volts to 5 volts is called its "positive transition" or "leading edge". It is this transition which turns functions on and off. All trigger outputs and all trigger inputs are RED jacks. It is possible in some cases to use appropriate control voltages to trigger certain modules, particularly if the control voltage has a sharp leading edge. (There are also a few places where these Trigger pulses can be used as audio waves if they are fast enough, or as control voltages if a two-level control voltage is desired.) If a control voltage can be thought of as turning a knob by remote control, a trigger is like pressing a button or tapping a key by remote control. A trigger looks like this:



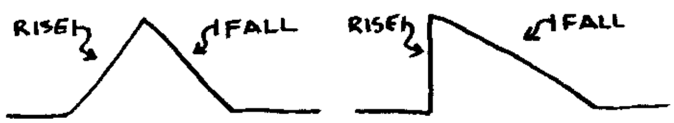
TRIGGER

Each time a keypad is touched on the TKB, a trigger pulse appears at the KP (Key Pulse) output. It will remain in its HI state as long as the key is being touched.

In the above patch this trigger is sent to the TRIG-IN of the DSG at the bottom right hand of the module. When this module receives a Trigger pulse it produces exactly one VOLTAGE ENVELOPE.

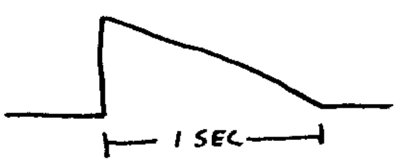


This voltage envelope is a common, simple acoustic envelope similar to many musical envelopes such as piano, guitar, etc. It has two basic parts: The RISE and the FALL.



These two slopes are set by the two pots on the DSG labelled RISE and FALL. With these two pots the Rise and Fall time can be set anywhere from 1/1000th of a second to about 5 seconds.

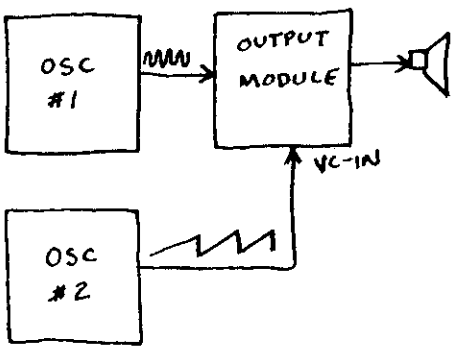
6.2 Patch from the KP out on the TKB to the TRIG-IN on the DSG and tap a keypad on the TKB. Directly above the output jack on the DSG is an LED (light emitting diode-- a small red light) whose brightness is proportional to the voltage of the envelope output. That is, as the envelope rises in voltage, the light gets brighter. Set the Rise and Fall pots to about 11 o'clock. The LED should take about one second to go from off to fully lit to off again. Different settings of the Rise and Fall pots will produce different timings. Set them so that they produce an envelope of this type:

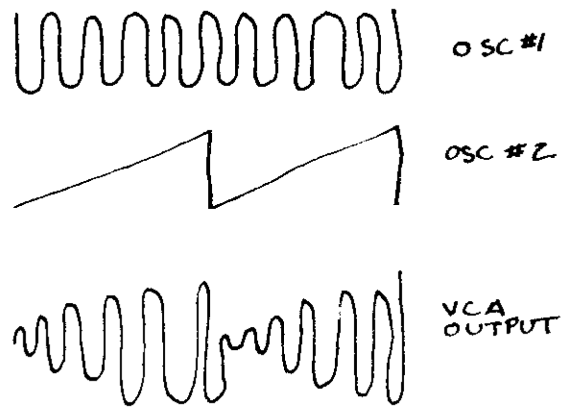


6.3 Complete the patch from the output of the DSG to the VC-in of the VCA (the VC-controlled mixer will be referred to as a VCA when being used in that function). When doing so make sure that the Gain pot is set to the appropriate setting. If there is a small amount of sound "leaking" through, turn the Gain pot slowly to the left just until nothing is heard. Some VCA's (the UPAP, QMX, QCA, QVM, and SMX) can be overloaded if the initial gain is set too high. This will not damage the module, but it might overload your amplifier or speakers. Use caution on these modules, always starting out with the pot turned down, then increasing the gain with the control voltage applied until the sound is the proper level.

6.4 Touch a keypad on the TKB. This will cause a number of things to occur simultaneously. First, as already discussed, it will cause the Stage of the TKB that has been touched to be activated. At the same instant it causes a Trigger Pulse to be produced at the KP output of the TKB. This pulse Triggers the DSG to produce its envelope. This voltage envelope is patched to the VC-in of the VCA where the effect is as if turning the GAIN pot up and then down by remote control. As the voltage of the envelope increases the gain of the VCA increases. When the envelope starts its Fall, the gain of the VCA begins to decrease.

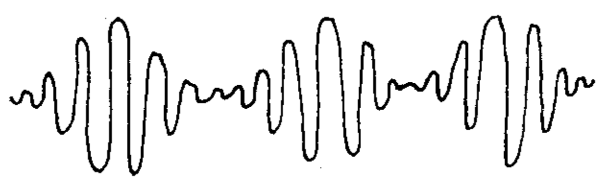
6.5 In general the voltage controllable parameters of a module are the same functions that can be controlled with its pots. For the VCA, then, the controllable function is its GAIN, where a High voltage to its VC-input creates a high GAIN and a low voltage produces a low GAIN. Once again, any voltage can be used to control the VCA, including an OSC. For instance



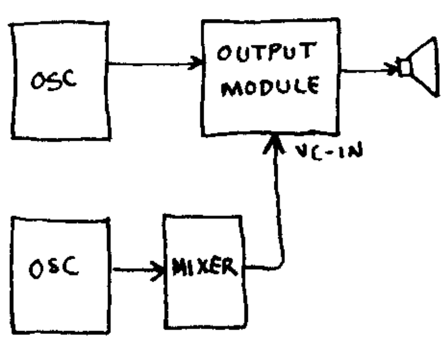


This will be heard, if the control OSC has a low enough frequency, as a kind of backwards sound which slowly gets louder and louder and suddenly cuts off -- only to begin again.

Using a SINE wave as the controlling voltage produces this effect:



6.6 Slowly increase the frequency of the modulating OSC. As you do so the sound will "beat" faster and faster. When this beating approaches 20 times per second (20 Hertz) a more complex sound appears that is somewhat similar to FM modulation. This sound is called Amplitude Modulation or AM. Like FM the sound is dependent on the Frequency of both OSCs and the relative amplitude between them. This relative amplitude, or Index, can be set in the following manner:

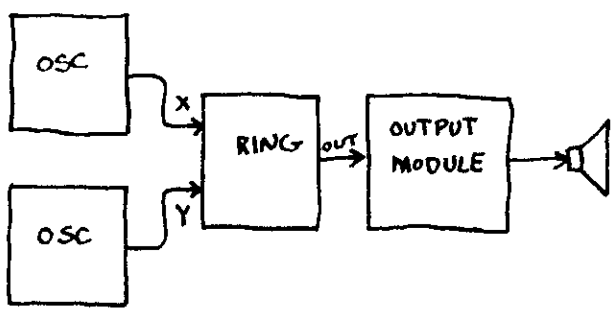


In these patches the GAIN on the mixer or VCA determines the Index. This technique can provide a wide range of sound types from tremelo to a hollow reedy sound to very complex sounds when non-SINE waves are used. In terms of the frequency spectrum, if a sine wave modulates a second sine wave, two NEW frequency components are produced, one being the sum and the other the difference between the two original sine waves. The original frequencies appear as well. That is, if the two original waves are 60 and 200 Hertz, then the output will be a mix of 60, 200, 260 and 140 Hertz waves.

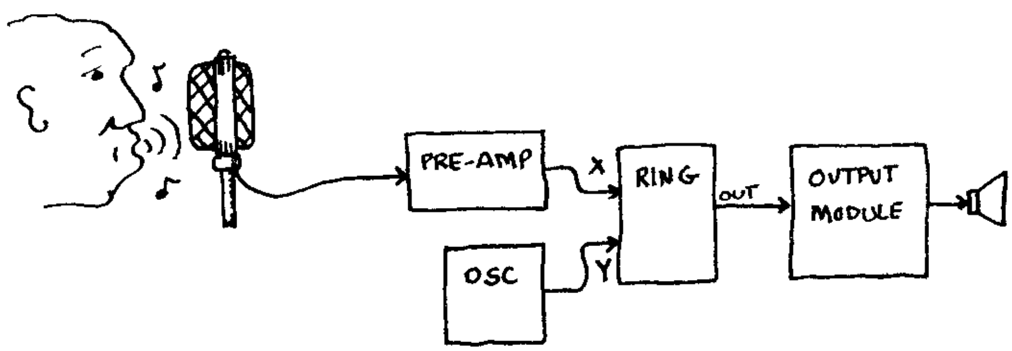
A module closely related to the VCA is the RING Modulator, which provides a third type of modulation along with FM and AM. Ring modulation is one of the oldest electronic music techniques and it is useful for producing complex and "odd" sounds similar to, but thicker than, the input sounds.

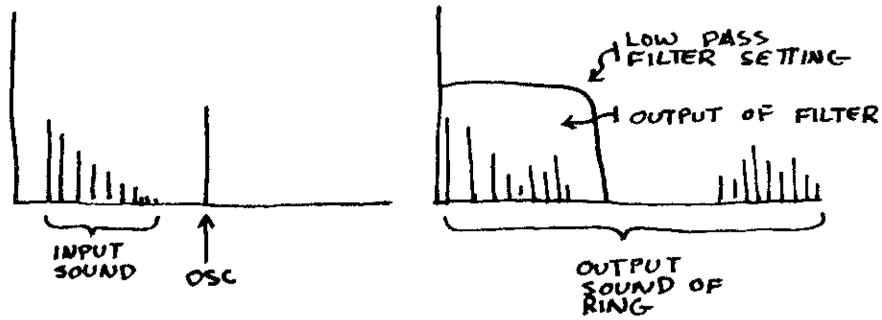
In its most basic mode a RING modulator takes two input frequencies and outputs the sum and difference frequencies ONLY. That is, if one input is 500 Hertz and the other is 160 Hertz, then the output is a 650 Hertz and a 350 Hertz wave mixed together. This differs from AM in that the original signals are cancelled out. If the input signals are complex, containing overtones, then every overtone of one wave is summed and differenced with every overtone of the second wave.

6.7 On Serge system RING modulators the two inputs are labelled X and Y (though in some ring modulators these inputs may be labelled as simply #1 and #2, or as signal and carrier). The output is labelled OUT. The pot at the bottom of the module is not a GAIN pot, but rather a pot which changes the function of the module from a standard VCA (full left) to a RING modulator when it is nearly full right.

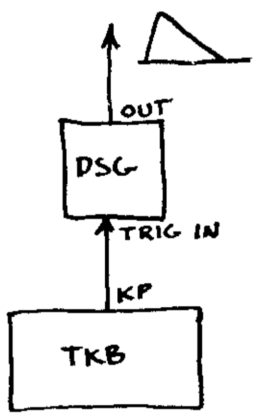


The RING modulator is often used in conjunction with sounds from the "real" world ("concrete" sounds) to give them an electronic feel. In this case, the "concrete" sound is fed to one input of the RING and the electronic sound to the other. It can also be used as a kind frequency shifter where a sound is shifted to a higher or lower frequency. For this, a filter must be used in conjunction with the ring modulator to filter out either the sum or difference component. This kind of frequency shifting alters significantly the harmonic relations of the overtones of the sound being shifted.



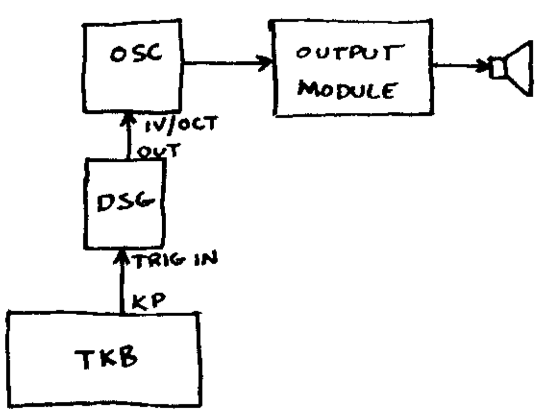


6.8 Some of the RING modulators on Serge systems have two auxiliary inputs, labeled VC-Y and VC-X, which act like VCAs for their respective input. They are useful for bringing out the original sound amidst the RING MODULATED sound.

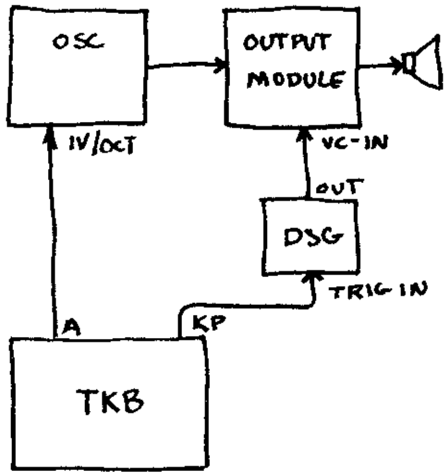


The DSG produces one voltage envelope every time it is Triggered. This trigger pulse may come from many places on the synthesizer but in the above patch came from the TKB each time a keypad was touched. The slope of the rise and fall of the envelope is set by the two pots labeled RISE and FALL.

6.9 While this voltage envelope is often used to control the amplitude or gain of a sound, it may be used to control any controllable module. In the following patch the envelope is controlling the frequency of an OSC:



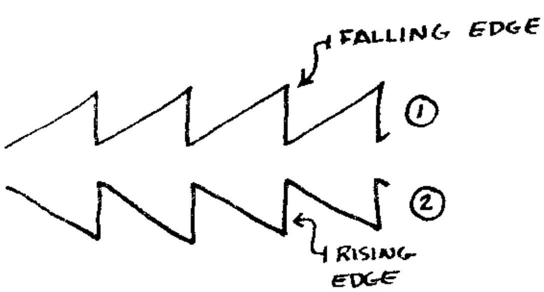
The following patch uses the TKB to trigger the DSG and control the frequency of the OSC.

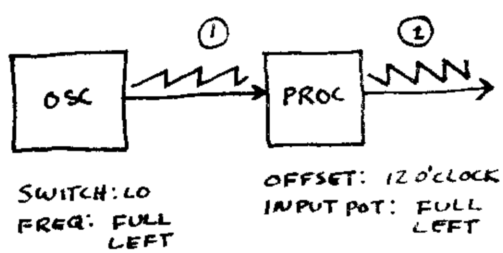


There are other features on the DSG and other ways of controlling it to extend its use far beyond this simple control function.

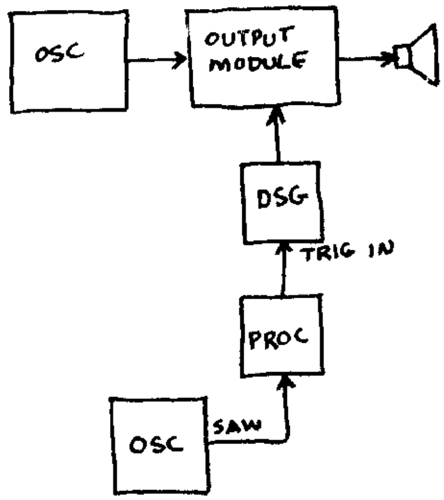
TRIGGERING FROM OTHER WAVEFORMS. The DSG, and in fact all TRIGGER-activated devices on the Serge, are triggered by the positive or rising edge of the Trigger pulse and not by the falling edge or the +5 voltage level itself. Not only a Trigger pulse from a trigger output but any sufficiently fast rising edge will trigger the DSG. The SAW wave output of PCO looks like this:

Note that it only has a Falling edge and therefore cannot trigger the DSG. However, this wave can be inverted by a Processor. The saw output of the OSC is patched into any of the three inputs of the PROC. The associated pot of the input should be set full left. Settings to the left of 12 o'clock on a processing input produce inverted outputs. While Processors usually accept control voltages, they can also accept audio voltages.





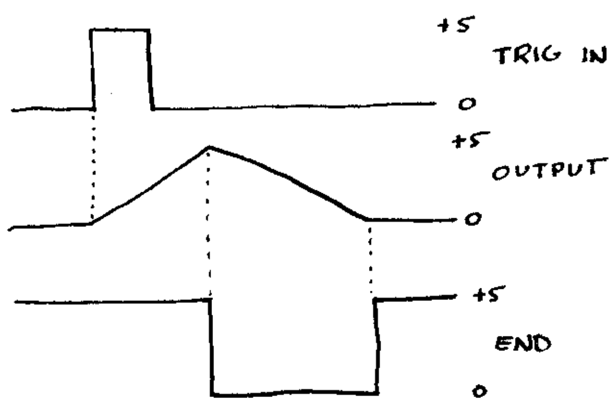
If the PCO is set LO and the frequency set very low (to the left), the output of the processor will be a series or "train" of rising edges that can be used to trigger the DSG over and over again.



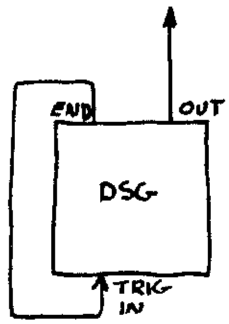
Some processors have an OFFSET pot that adds a set voltage to the output depending on its settting. If the processor being used in the above patch contains an offset pot it should be set at its 0 volt position (12 o'clock).

The DSG should be set so that the duration of the envelope as a whole is shorter than the "period" (the period of a wave is how long it takes to complete one cycle) of the OSC's sawtooth wave so that a full envelope can be generated before a new one is triggered. A DSG will not respond to a new trigger until it completes its entire Rise-Fall cycle.

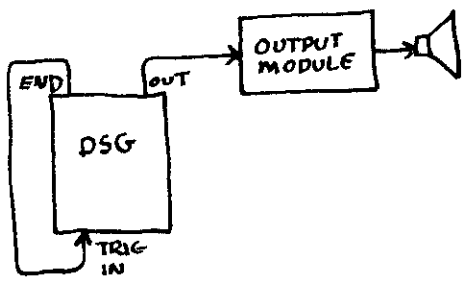
SELF-TRIGGERING and DELAY. The DSG has an END output that generates a rising edge at the completion of each envelope and REMAINS high until after another envelope is triggered.



This END Trigger can be used to Trigger any triggerable module, including ITSELF.

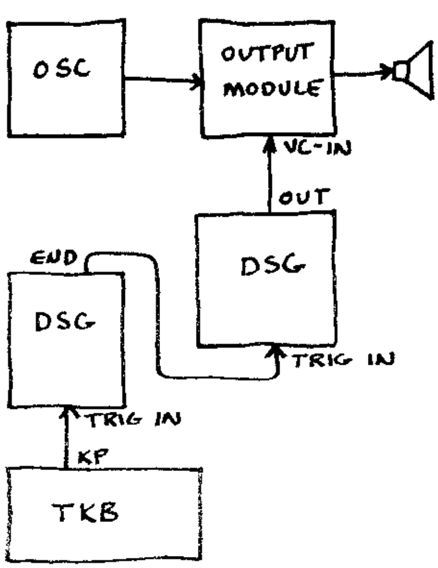


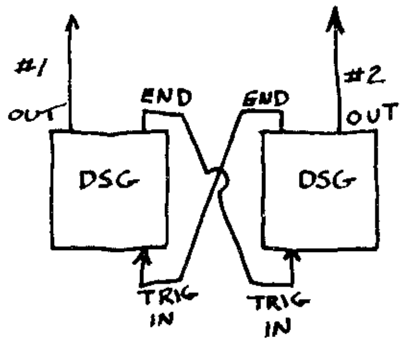
When the envelope has completed its cycle, a Trigger appears at the END jack. Since END is patched to TRIG-IN, the module is re-triggered and the cycle begins again. This patch turns the DSG into an oscillator! When the Rise and Fall times are set short enough, so that the total rise and fall time is less than one twentieth of a second, this OSC is within the audio range and can be heard directly through the speakers.



By adjusting the RISE and FALL pots different waveshapes can be achieved from saw to triangle.

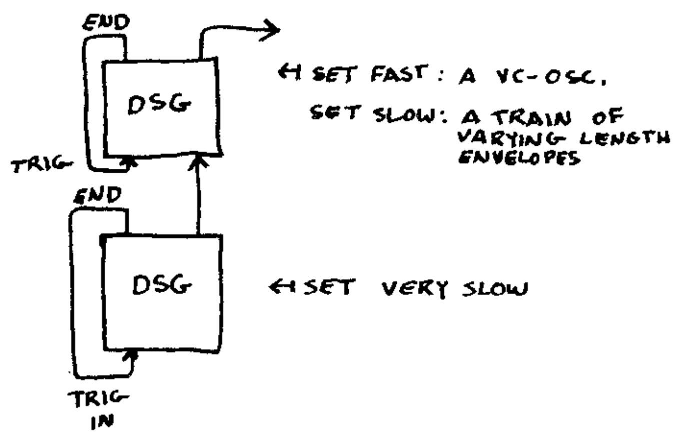
By using two DSGs a delay can be created between a trigger and the generation of an envelope or between successive envelopes of a DSG that is oscillating.



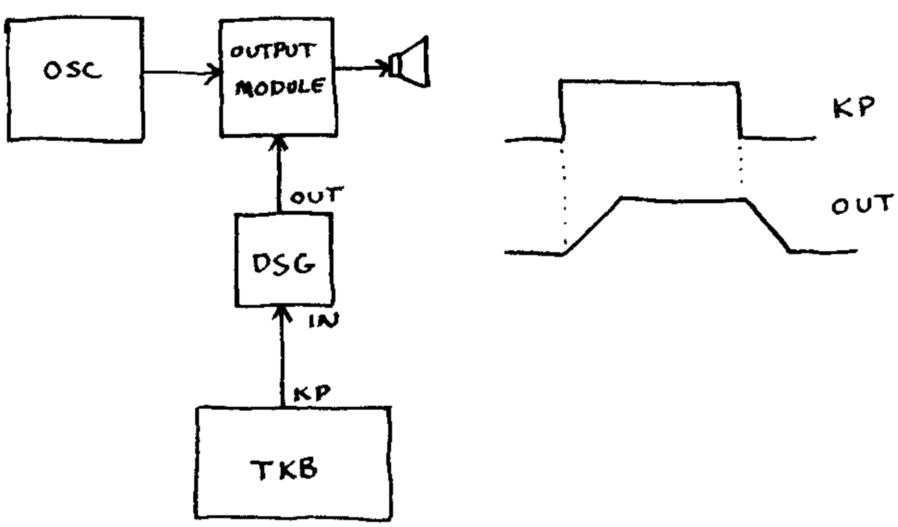


In these examples the length of the second envelope determines the delay. This envelope is not "heard" in any other way.

VOLTAGE CONTROL. Naturally the DSG can be voltage controlled itself. In this module different control voltages produce different Rise and Fall slopes and thus different length envelopes. The VC-IN has an associated 3-way switch which allows for 3 possible modes of control. When the switch is positioned to either RISE or FALL the control voltage controls EITHER the Rise or the Fall. In the center position a control voltage will control both Rise and Fall simultaneously. The VC-IN has an associated control voltage processor so that the control voltage can be amplified, attenuated and/or inverted. One good place to get control voltages to control one DSG is from another DSG:



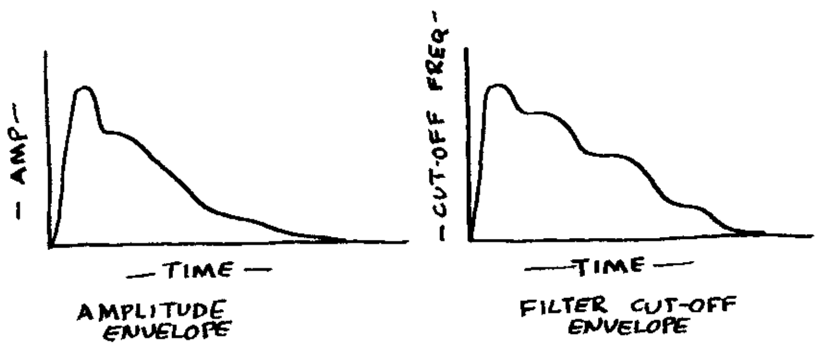
ENVELOPES WITH SUSTAIN. The DSG has an input labeled IN which accepts a voltage. If this voltage is higher or lower than the output voltage of the DSG, then the output voltage will rise or fall to the input voltage at a rate set by the RISE and FALL pots. This input is useful for making envelopes which sustain as long as the Trigger pulse remains high. For instance, the KP out on the TKB remains at 5 volts as long as a finger is held down on the keyboard.



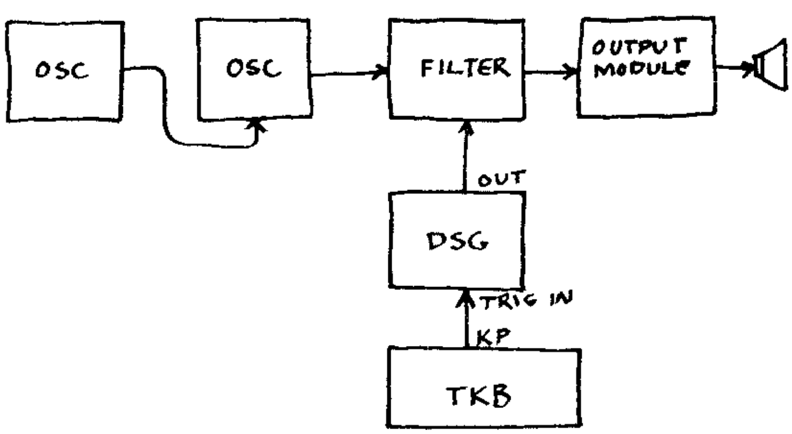
STEP SEVEN

When a piano note is sounded, not only does it have an overall amplitude envelope, but each harmonic or overtone has its own envelope. In most acoustic instruments the higher the frequency of the overtone the faster it dies away. The lowest tone, the fundamental, dies away last. This pattern is very much like closing down a fully opened LO PASS filter.

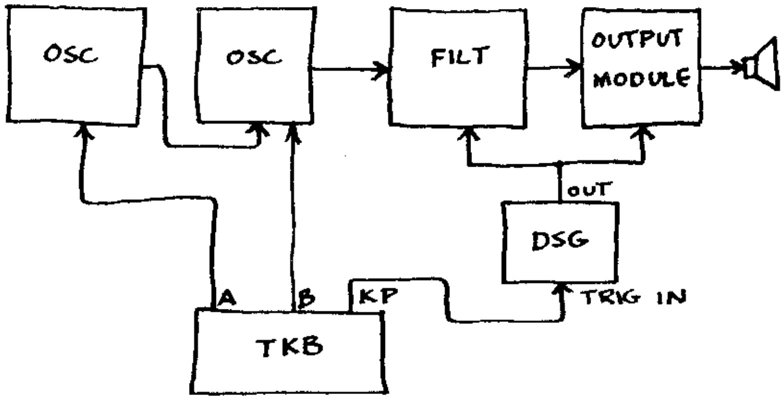
The control voltage applied to the filter sets the cut-off frequency. Usually the higher the voltage the higher the cut-off frequency. What makes it easier to simulate "natural" sounds using a VCA and a filter is that the amplitude envelope is often similar to the "harmonic spectrum envelope".



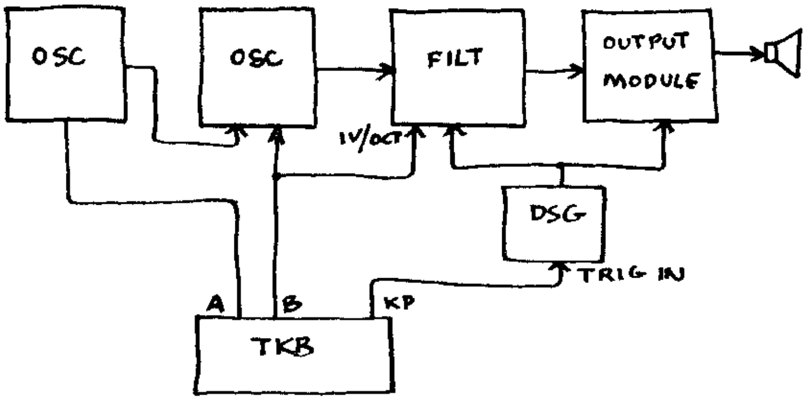
7.1 The similarities of these envelopes combined with the tautology "if all the harmonics die away the sound has died away", make it possible to simulate natural sound even without the use of a VCA. The cut-off frequency of the filter should be set low enough so that no sound gets through unless an envelope is applied. The envelope should be applied to the VCF input and the processing pot should be turned full right. The envelope should be set so that it rises rapidly and falls slowly. If the two OSCs are set to produce a fairly harmonic output, a bell-like sound should result.



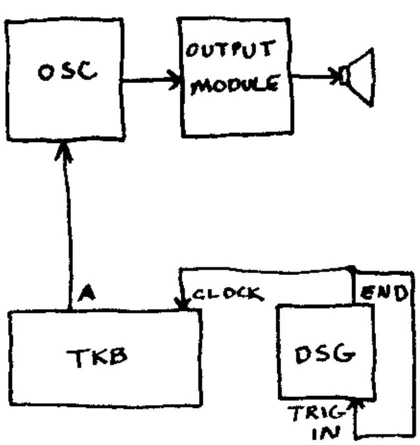
7.2 When this same patch is combined with a VCA controlled by the same envelope, and if the two OSCs are controlled by the A and B outputs of the TKB, the result can be an interesting keyboard instrument over which the composer has a lot of control.



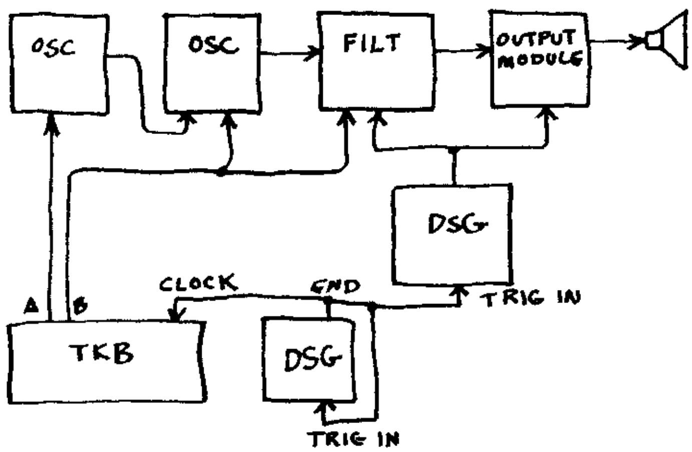
7.3 A limitation of this patch is that the initial cut-off frequency of the filter is always the same while the frequency of the OSC shifts under control of the TKB. A way to correct for this is by patching row B not only to OSC #2 but to the 1V/OCT input of the filter. Since these inputs are very precisely calibrated, and since they are being controlled by the same voltage, the filter and the OSC will "track", so that the cut-off frequency of the filter will follow the frequency of the oscillator.



7.4 This patch uses the TKB in only one of its two major modes: the keyboard mode. It is possible to use it in an automatic or SEQUENCER mode where different stages are accessed automatically. Near the upper right-hand corner of the TKB is a CLOCK input which accepts a trigger pulse (it is red, indicating a trigger in). Every time the TKB receives a trigger pulse at its CLOCK input it steps one stage to the right. If it is at stage 8 it will step to stage 9. If it is on stage 16, however, it "wraps around" to stage 1. Using a DSG set up as a slow OSC to provide trigger pulses, the following patch will step the TKB through its stages:

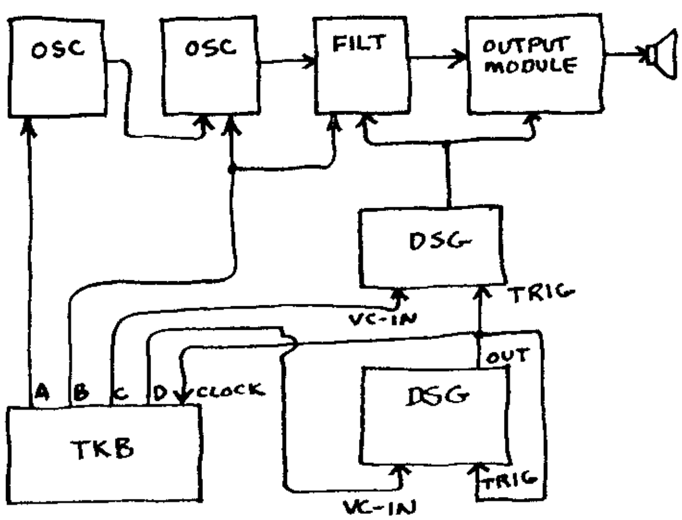


7.5 For this patch it is helpful to set all the pots in row A to different settings so that the different stages are distinguished from each other. Below is a logical extension of this automated TKB combined with the instrument sound we previously developed:



In this patch DSG #2 acts like a clock for the whole system. As it "ticks" it steps the TKB along and simultaneously triggers the #1 DSG.

7.6 A musical drawback with this patch is the regularity with which the system moves along. But since the DSG is voltage controllable, we have a way of altering the clock's rate by using a row of the TKB to Voltage Control it. The speed of the clock now has become an intregal part of the "musical instrument" that was constructed by patching together modules. By setting the pots in the row controlling the DSG, it is possible to set the time at each step-- in other words, to control the rhythm. Furthermore, row D can be used to control the length of DSG #1, the envelope to the VCA and to the Filter. With a thoughtful setting of the pots, 16 different sounds in a desired order, in any rhythm, can be produced and repeated:



THE OVERALL DESIGN OF THE SERGE SYSTEM

With this last STEP all three kinds of voltage and all four major types of modules found on the Serge have been used.

The Three Kinds of Voltage:

AUDIO VOLTAGES: Black Jacks. 20 to 20,000 Hertz. Output voltage typically -2.5 to 2.5 volts. Audio voltages produced by blue jacks typically 0 to 5 volts. However, any voltage range, so long as it oscillates in the audio range can be used as an audio voltage. Black inputs are typically AC coupled, meaning that the slow or non-changing aspects of the voltage are blocked.

CONTROL VOLTAGES: Blue Jacks. Typically 0 to 500 Hertz but can be higher particularly in the case of FM and AM. Usually either -5 to 0 volts or 0 to 5 volts but can range over -10 to +10 volts. Blue inputs are DC coupled meaning they respond to the full range including negative voltages.

TRIGGER or PULSE VOLTAGES: Red Jacks. Either 0 volts or 5 volts with a fast rising edge between 0 and 5 volts. Some red outputs can hold the high level indifinitely, others fall back to 0 in a set time. Red inputs are triggered by the rising edge and therefore other voltages, such as inverted saw waves, can be used to trigger. Some red inputs control certain functions of a module as long as the voltage remains HI. In these cases any 5 volt level will sustain the function.

The Four Major Kinds of Modules:

SIGNAL GENERATORS. These modules produce audio voltages as their output. The oscillators are examples of this kind of module. The Noise Source module is another.

CONTROL VOLTAGE GENERATORS. These modules produce control voltages as outputs. Envelope generators and sequencers are examples of this type of module.

AUDIO PROCESSORS. These modules input audio voltages, operate on these voltages and output a related audio voltage. Filters are an example of audio processors. In general they operate on the timbre of the sound. Another type of audio processor inputs two or more audio signals and combines them in various fashions. Mixers and ring modulators are examples of this type of module.

VOLTAGE PROCESSORS. These modules input a control voltage and output a related control voltage. A processor is an example, in which case a control voltage is the input. The output might be the same voltage inverted.

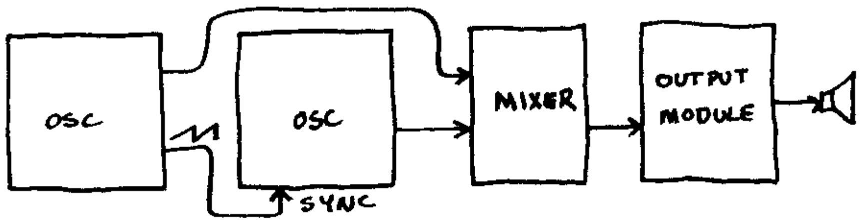
A fifth type of module that has not been dealt with yet is an Audio-to-Control-Voltage converter. One example might be a module that inputs an audio signal and outputs a control voltage representing the envelope of that sound. Such a module is called an envelope follower.

Once the concept and basic principles of these five types of modules are understood, an infinite array of new "instruments" can be made or "patched" out of the modules available on the Serge. New modules, once their basic type is determined and their internal workings understood, can be easily added to existing modules. In general, modules of the same type can be substituted for each other.

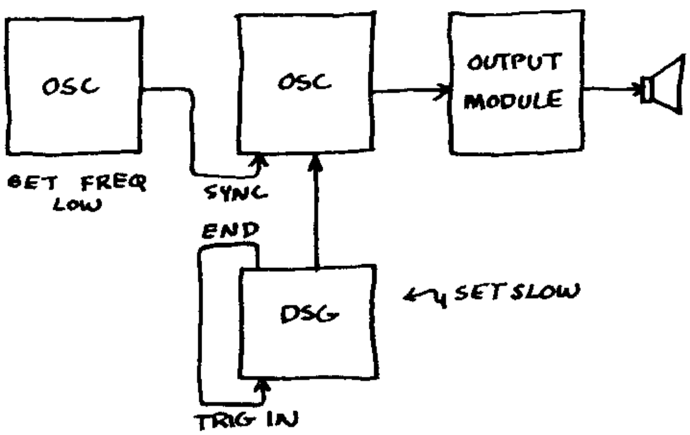
Each of the next five chapters will cover one of these five types of modules, presenting modules and functions not yet covered. It is suggested that each module be explored as it is presented by setting up patches using it with modules already understood.

SIGNAL SOURCES

OSCILLATORS. Both the NTO (New Timbral Oscillator) and the PCO (Precision Controlled Oscillator) have been treated earlier in this manual as to their ability to generate audio frequencies. Both the NTO and the PCO have an input labelled "SYNC". This input allows two OSCs to be locked together so that they will not drift apart in frequency. Two OSCs that have drifted just a few Hertz apart can cause a "beating" to occur at their difference frequency. Sometimes this is the desired effect, producing a choral quality to the sound. When using the SYNC, one OSC is locked to the fundamental OR to a strong overtone of a second OSC. Locking onto an overtone is useful in the setting up of chords.

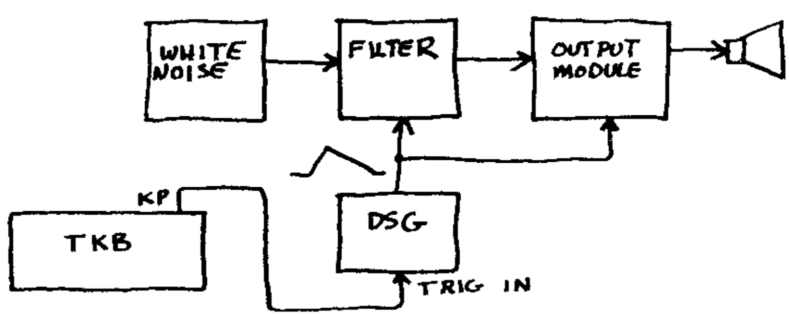


In the above patch OSC #2 is locked to OSC #1. It cannot drift. An interesting phenomenon occurs if OSC #1 is set very low and its SAW wave is used to SYNC OSC #2. If OSC #2 is now swept upwards over its range, either using its pot or a control voltage, you will hear it locking onto one overtone after another, creating a "just-intoned" stepped scale.



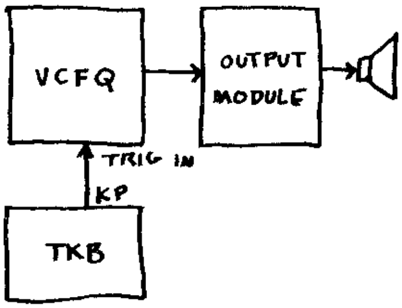
WHITE and PINK NOISE. White noise is a complex wave in which ALL frequencies appear mixed together. The sound is a sort of hissing sound. Since it contains all frequencies it can be filtered in various fashions to produce bands of sound in many different frequencies. This is the ultimate material for subtractive synthesis. It is also useful for producing percussion sounds such as snare drums.

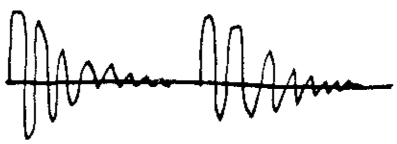
WHITE NOISE



Pink noise is like White noise except that it sounds lower, more like a waterfall. The low frequencies have more amplitude.

RINGING FILTER. The VCFQ can be "rung" much like a gong, with a trigger pulse to the TRIG-IN. By adjusting the FREQ, different pitches can be achieved. Adjusting the Q will alter the sound from percussive clicks to bell-like sounds. The output is a damped SINE wave.

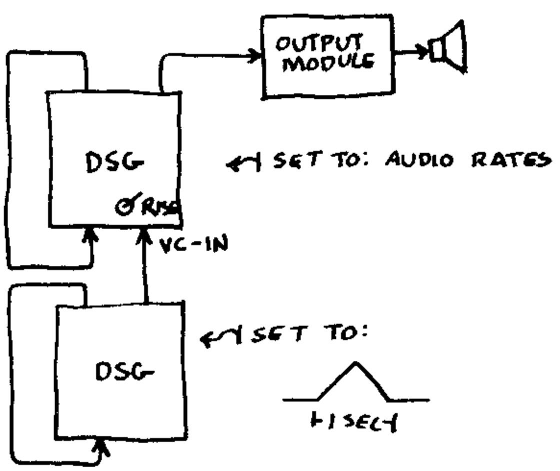




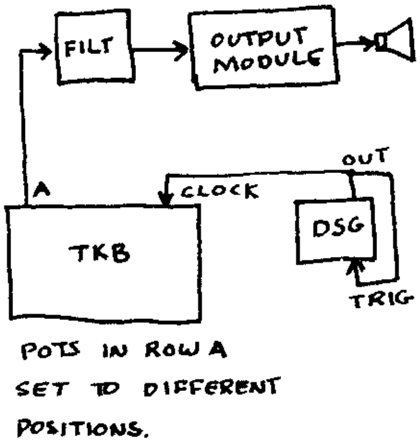
DAMPED WAVES

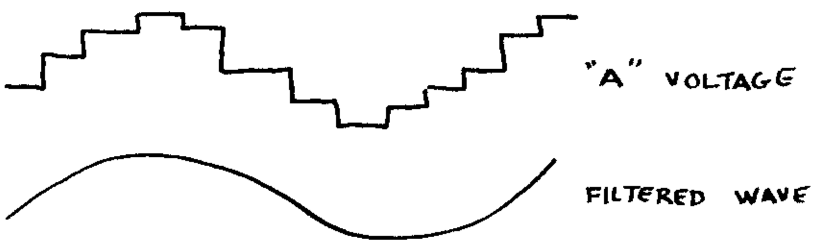
This technique can be used when other signals are applied to the INPUT of the filter to produce a wide range of interesting sounds.

SLOPE GENERATOR. As already discussed, the DSG, can be patched to trigger itself to produce an OSC. The frequency is set by adjusting the Rise and Fall time either with the pots or with a control voltage. The DSG can have a voltage controlled wave shape if the switch on the VC-in is set to either RISE or FALL.



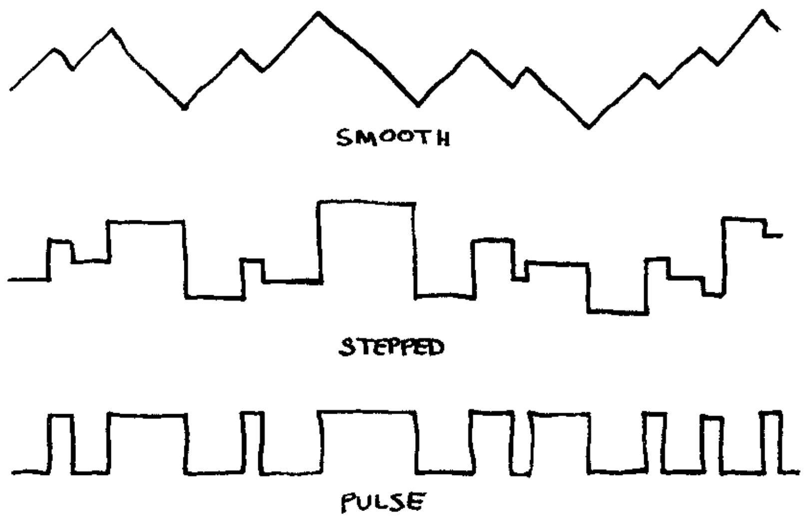
AUDIO SEQUENCES. A sound source of a more unusual nature can be found in the TKB. To use the TKB as a sound source the CLOCK trigger must be well into the audio range. The output is taken from either A, B, C or D outputs and sent directly to the output or to a audio processor such as a filter. Each pot on the chosen row defines the voltage of the wave at one point, so the waveshape is composed of sixteen levels. The frequency is one-sixteenth of the frequency of the clock. Interesting waveshape variations can be produced by adjustng the position of the various pots.





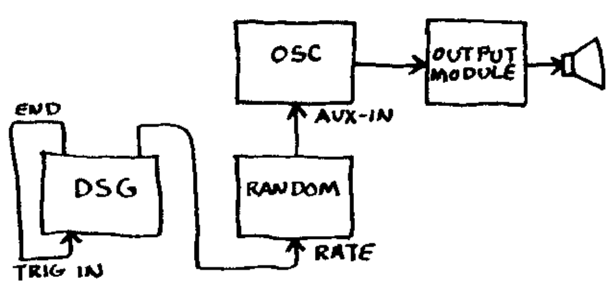
RVG and the 2RVG (The Random Voltage Generator). Very often you will want a changing voltage. Either it won't much matter what voltage it is, or you may want a surprise. Such situations come up when working with certain kinds of modern music such as Stochastic or Aleatoric, as well as other musics such as symphony and rock and roll. In particular it sounds more "animated" to have the timbre of an electronic sound slightly changing in a random or non-consistent manner.

The Serge provides three kinds of random control voltage: stepped, smooth, and pulse. These are diagrammed below:

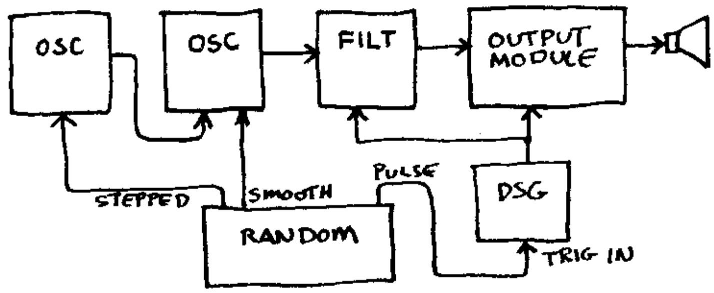


The RATE pot at the bottom of the module determines the overall rate of change, a function which can be voltage controlled.

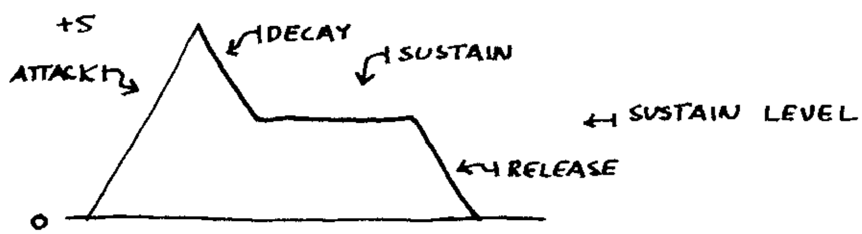
The rate of randomness can be controlled on the random module, and a Processor or a processing Input can scale the random output to any desired level. The following patch is useful for exploring the possibilities of the smooth and the stepped random output:



If your Serge has more than one random module it is possible to use a random control voltage to control the rate of the second random voltage. The pulse output can be explored using the following patch to provide a random rhythm:



ADSR. The Extended ADSR is an envelope generator that can produce multi-segmented envelopes of a more complex variety than a single DSG is able to provide. In certain kinds of synthesis this is necessary, since few natural envelopes are a simple rise and fall. The ADSR is able to provide a four-part envelope labelled Attack, Decay, Sustain and Release, as in the following diagram:

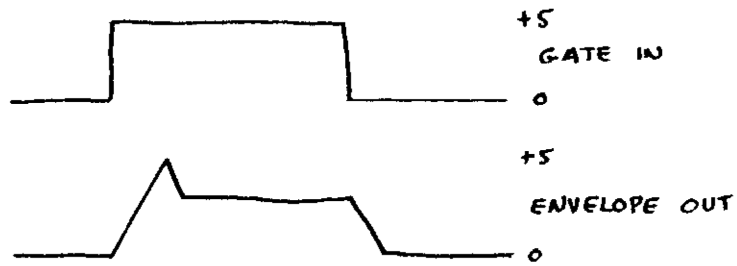


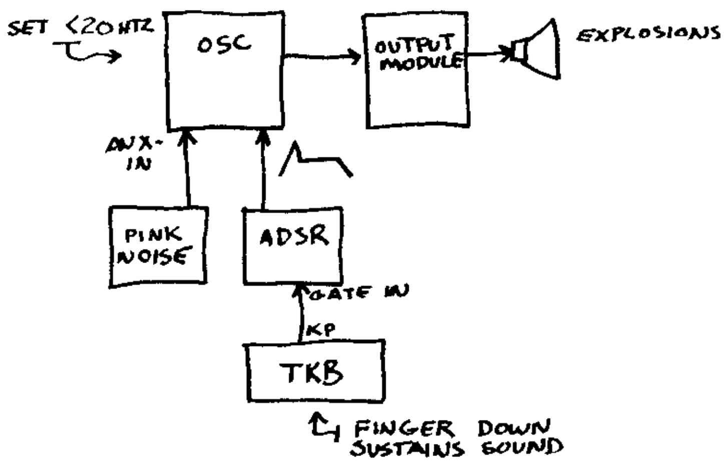
This envelope, in a general sort of way, represents the envelope of trumpet or any instrument which can sustain a note at a steady level. The sustain section is settable to different Sustain Levels by means of a pot and voltage control. In addition to these functions the ADSR also provides a delay that sets an amount of time between the receiving of a trigger and the onset of the envelope itself. This is useful when triggering related envelopes with the same Trigger or Gate.

The module is triggered by a pulse to its GATE input. Usually this trigger comes from a keyboard device such as the TKB's keyboard output (KP) which is a trigger that stays at a +5 volt level as long as a finger remains on the key:

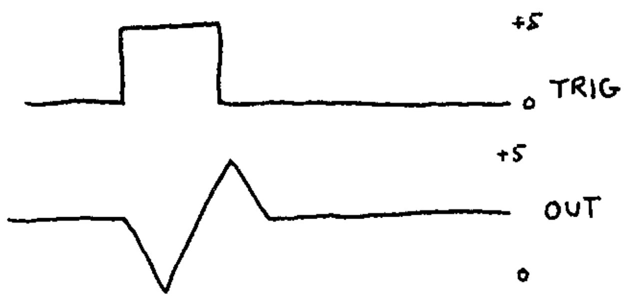
This sustained high level will be used in the timing of the ADSR's output.

The ADSR has five pots, each with an associated control voltage input jack. These control voltages affect the same segments as the corresponding pots. The uppermost pot/control voltage input sets the length of the delay between receipt of the trigger pulse and the onset of the envelope. The further left the pot is set the longer the delay. The second pot controls the slope of the Attack in much the same way as the Rise pot on the DSG. It too, is voltage-controllable. The third pot/VC-input controls the slope of the initial Decay, which falls to the voltage level set by the fourth pot (Sustain). The ADSR will sustain the voltage output set by this fourth pot AS LONG AS THE GATE INPUT REMAINS HIGH. In the case of the TKB's KP output, this is as long as one holds a finger down on the keypad. When the GATE goes low, the fifth pot determines the slope of the final decay.



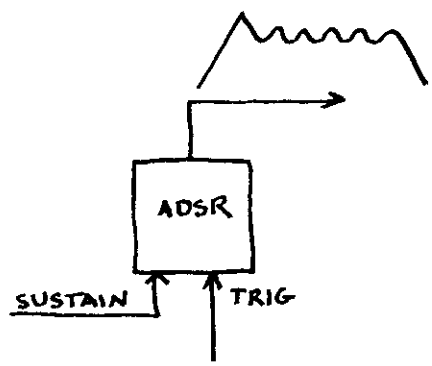


The ADSR functions in a slightly different fashion if the trigger pulse is applied to the TRIG input and not the GATE input. When there is no input to the GATE, the output of the ADSR remains at the voltage level set by the Sustain pot (pot #4). When the ADSR receives a trigger pulse the voltage drops to 0 volts from this level at a rate set by the Release Pot (#5). The voltage then rises to the peak voltage at a rate set by the attack pot and finally drops back to the level set by the sustain pot at a rate set by the decay pot.



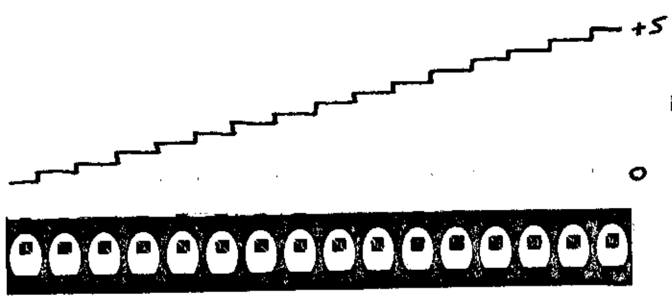
Using both inputs, more complex envelopes are possible. For example, during the sustain time set up by a GATE pulse, a trigger received at the TRIG will cause a new attack to start.

While in the sustain mode of an envelope, the ADSR will respond to changing control voltages at its sustain input. This makes complicated sustains possible.

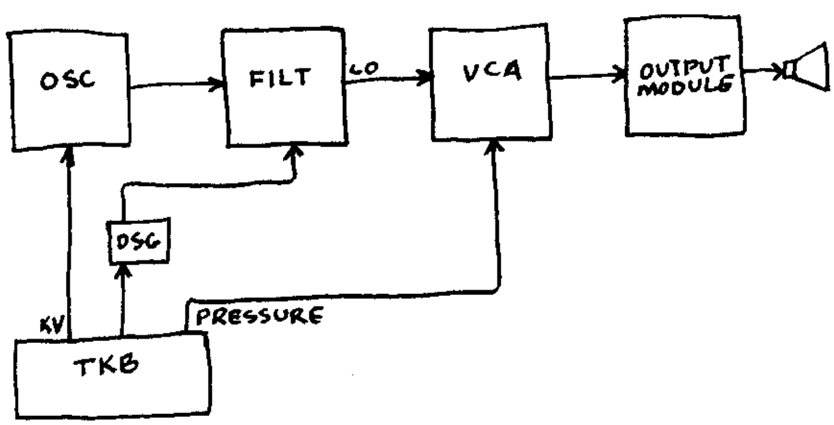


TOUCH PAD KEYBOARD on the TKB. The TKB can be thought of as two separate but interconnected modules: the Sequencer and the Keyboard. The two modules can be partially separated from each other by switching the KEYS switch to OFF.

The Keyboard provides the user with three voltage outputs. KV outputs a voltage depending on which keypad was last touched. Each keypad is assigned a voltage such that keypad #1 has the lowest voltage, keypad #2 the second lowest and so on up to keypad #16 which has the highest voltage. The voltage increase between any two adjacent keys is equal and can be used whenever an equal-tempered scale is needed. A processor or processing input can be used to calibrate an oscillator to produce any desired equal-tempered scale including the western 12 divisions to the octave.

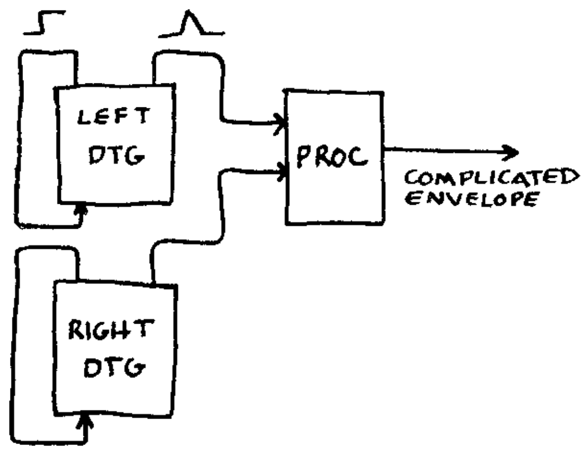


The Pressure output is a voltage proportional to the amount of pressure applied to the keyboard with the finger. When used to control a VCA, it can act much like an expressive envelope generator simulating the "piano-forte" (soft-loud).

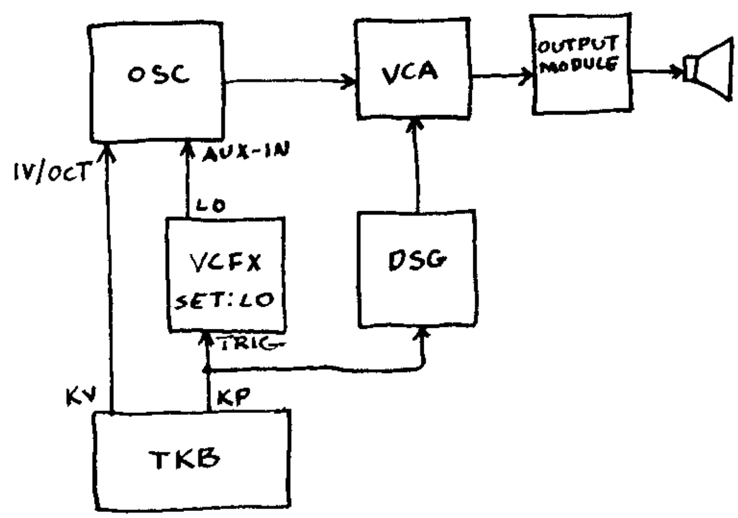


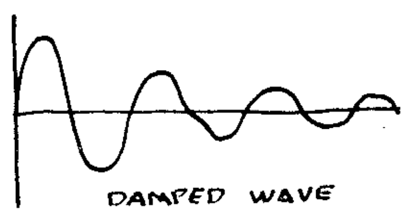
The pressure output works by using a capacitance detector. You will find that the position of your other hand in relation to the faceplate of the synthesizer affects its output.

TRANSIENT GENERATOR. The DTG is a smaller version of the DSG. It is a dual module, the modules being side by side. The Rise and Fall are only voltage-controllable simultaneously and can not be controlled separately. Each DTG has two outputs: a final pulse which can be used for recycling itself or for triggering another function, and an envelope output.

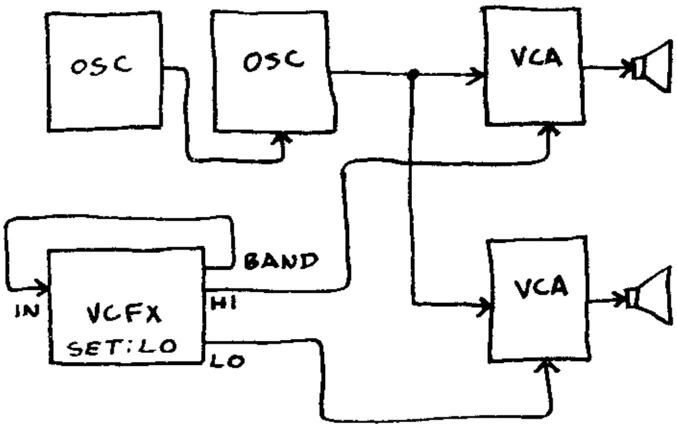


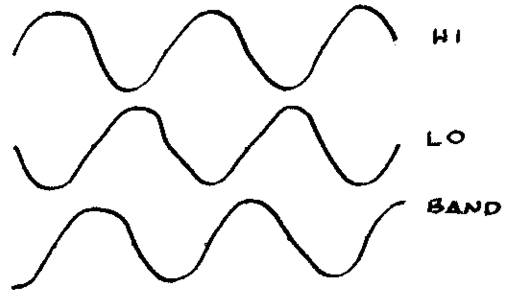
EXTENDED RANGE VCF. A surprising source of control voltages is the Extended Range VCF (VCFX) which, like the Variable Q filter, can be made to "ring" -- in this case, to ring at sub-audio or control voltage frequencies. The VCFX should be set to its LO setting. A trigger applied to its TRIG in, when the Q is at a high enough setting, causes the damped oscillation which is a useful control voltage for tremolo effects.



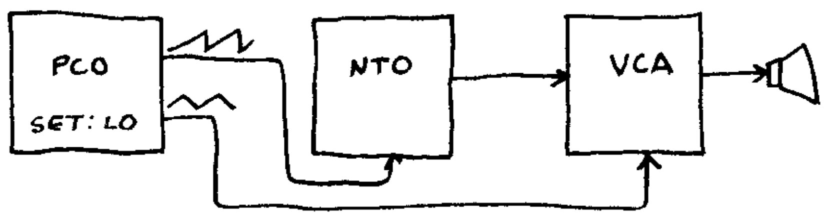


When the BAND output of the VCF is patched back into its own input to produce a feedback loop, the VCFX will oscillate. Sine waves are then available at its HI, BAND, and LO outputs. these sine waves will be 90 degrees out of phase with each other and are useful for creating stereo and quad panning patterns.





LOW FREQUENCY OSCILLATOR. the PCO in its LO setting produces voltages of sub-audio frequency. This frequency can be made even lower by the application of a negative voltage, or a positive voltage inverted by its processing input.



AUDIO PROCESSORS.

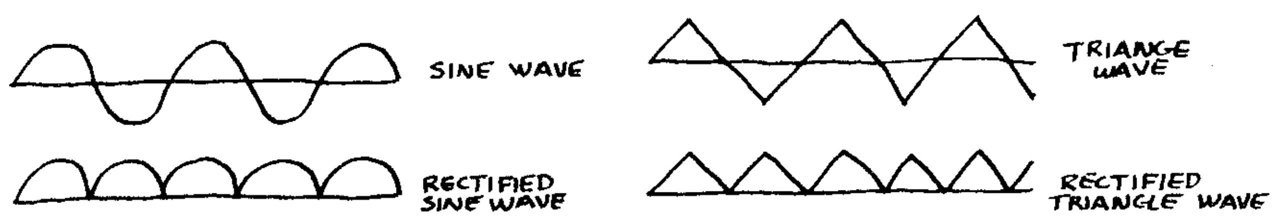
There are four major paths to analog electronic music synthesis.

ADDITIVE SYNTHESIS: Since any sound can be shown to be made of sine waves, it is possible to construct any sound by adding the appropriate sine waves together. While conceptually this seems to be the most flexible method of synthesis, in reality it is a difficult and time-consuming procedure except in some limited cases. Often it is more practical to mix already complex sounds together.

SUBTRACTIVE SYNTHESIS: The opposite of additive synthesis is subtractive synthesis. In its ideal form, one can take white noise, which contains ALL frequencies and subtract the ones not wanted, much like the sculptor chipping away at a block of stone. More commonly, the synthesist takes approximate waveforms, such as sawtooth waves, or a mix of waves, and "chips" away at these sounds.

MODULATION: There are a number of electronic processes that take one simple waveform and modulate it, or alter it, with a second waveform. This would include AM, FM and Ring modulation. The resultant waveforms are then often subjected to either additive and/or subtractive synthesis.

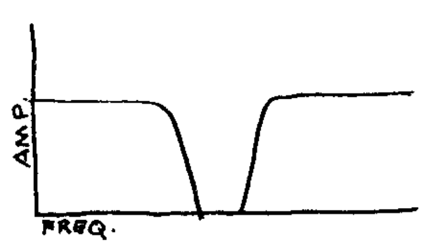
WAVESHAPING: Waveshaping is a technique where a given wave is input into a device and a related but different wave is output. For instance, a simple waveshaper is a "Rectifier" which outputs the absolute value of its input wave.



Modules that waveshape signals, add signals together, Subtract parts of signals, or that Modulate signals are called Signal Processors. The Serge system is a Signal Processor-rich synthesizer and includes many processors that are not found on any other synthesizer. So far in this manual the processors dealt with have been Mixers, Voltage Controlled Amplifiers and Filters.

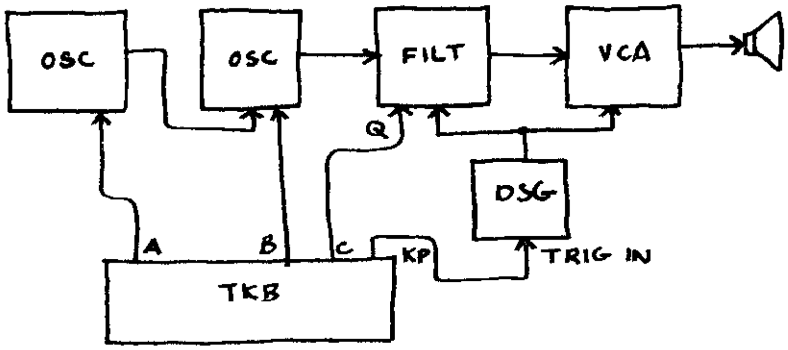
VARIABLE Q VCF (VCFQ). The VCFQ has all the functions of a regular VCF plus a few unique features.

NOTCH OUTPUT. This output allows all the frequencies of the input signal to pass EXCEPT those in a band directly around the setting of the FREQ pot and control voltage. It is useful for removing a certain sound from a more complex one.



NOTCH OUTPUT

VCQ or RESONANCE. "Q" is the property of a filter where the cut-off frequency is amplified and fed-back into the original signal to produce a bell-like or ringing sound. Different levels of Q produce different tonal qualities. In the VCFQ this function is both adjustable by a pot and voltage controllable by the VCQ input.

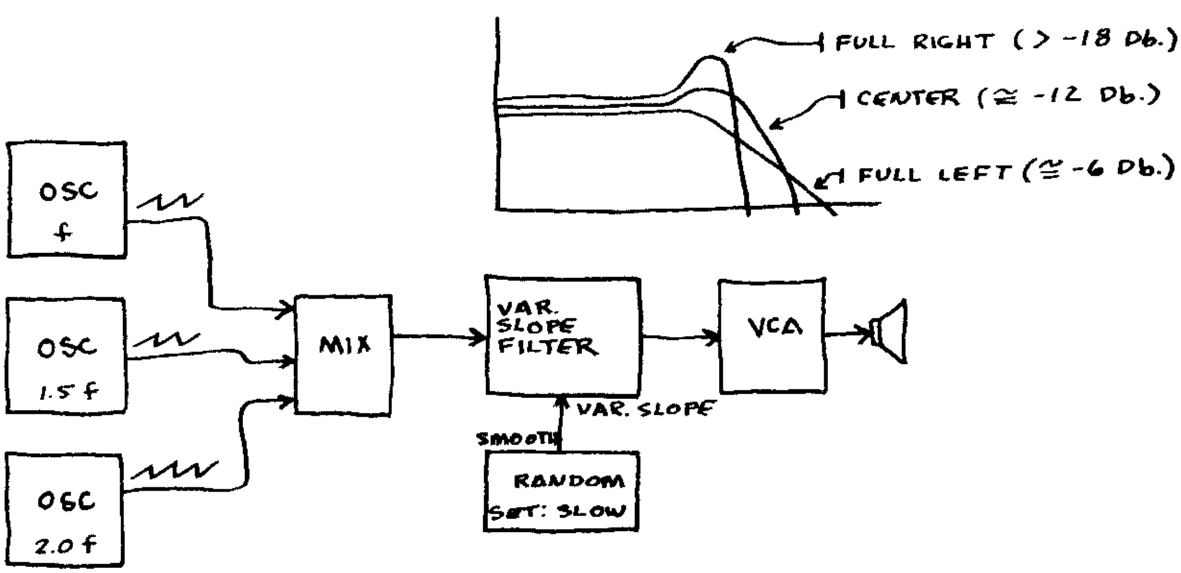


AUTOMATIC GAIN CONTROL. One of the problems with a Hi-Gain Q filter is that if the cut-off frequency hits an overtone that is higher in amplitude than expected, it can overload the filter and/or the speaker system and cause distortion. The AGC keeps the output gain constant at the cut-off frequency and is activated by inputting the signal into the AGC-IN. The regular input (IN) does not have an AGC, but does have a GAIN pot associated with it to attenuate the signal to desired levels.

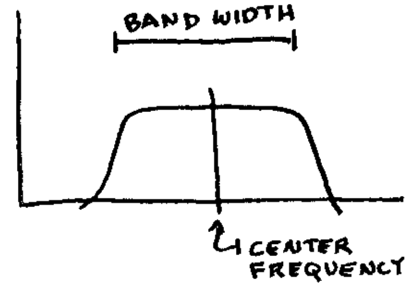
VARIABLE SLOPE VOLTAGE VCF (VCFS): Other than having the regular features of a voltage controlled filter, the VCFS has the following features:

MIXED INPUTS. Two inputs IN-1 and IN-2 which can be mixed using the MIX pot situated between the two inputs. Full right allows only IN-2 to be processed while full left allows only IN-1. Pot settings between these extremes allow a mix of the two inputs, the 12 o'clock setting being an equal mix.

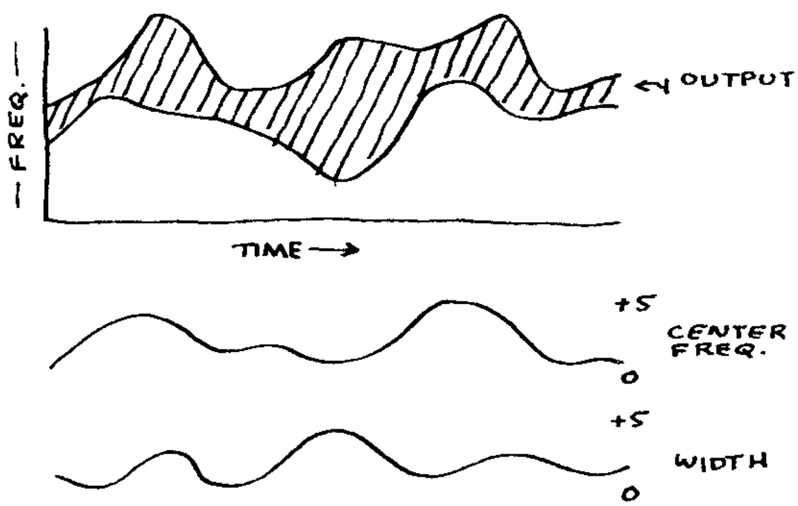
VOLTAGE CONTROLLABLE SLOPE. The Slope of a filter is the amount of attenuation per octave the input signal is attenuated beyond the cut-off frequency. Though an "ideal" filter has a 60 db/oct cut-off (or even greater) slope, this is not a particularly useful slope musically. In fact, different slopes have different tonal and timbral effects which can be used with good effect. In the VCFS the slope can be adjusted either by using a control voltage (the input has a Processing pot associated with it) or manually by using the pot directly beneath the VC SLOPE Processing pot.



VARIABLE BANDWIDTH VCF (VCF2). A band pass filter is useful for listening to a part of a much larger sound. The nature of this band can be described by noting its Center Frequency and its Band Width.



The VCF2 allows the user to either manually control or voltage control both of these parameters. The center frequency is controlled manually by the FREQ pot and voltage controllable by the VCF input (processed) and a 1V/OCT VC input (precision calibrated to be able to track the oscillators). The band width is also controllable by voltages using the VC-BW input (processed), the 1V/OCT VC input (precision calibrated) and by a manual pot located just below the processing pot.



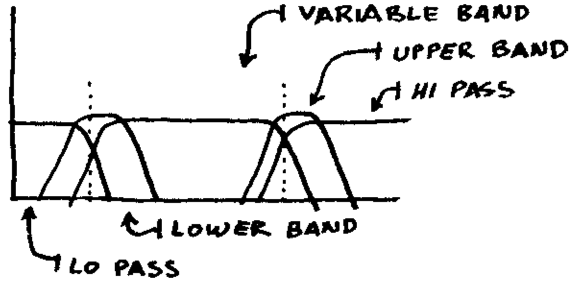
The VCF2 also has four other outputs:

LO PASS. The cut-off frequency being the LO edge of the Variable Bandpass.

HI PASS. The cut-off frequency being the HI edge of the Variable bandpass.

LO BAND. A fixed width bandpass filter whose lo cut-off frequency is the lo edge of the variable width bandpass.

HI BAND. A fixed width bandpass filter whose hi cut-off frequency is the hi edge of the Variable width bandpass.

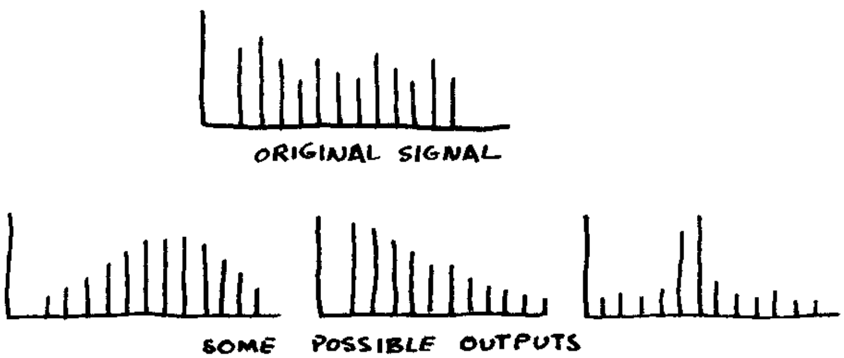


The VCF2's output on all five bands is flat, that is there is no resonance or Q, and therefore is ideal for processing concrete sounds, or for studio use since the filter does not change the quality of the sound that it is passing.

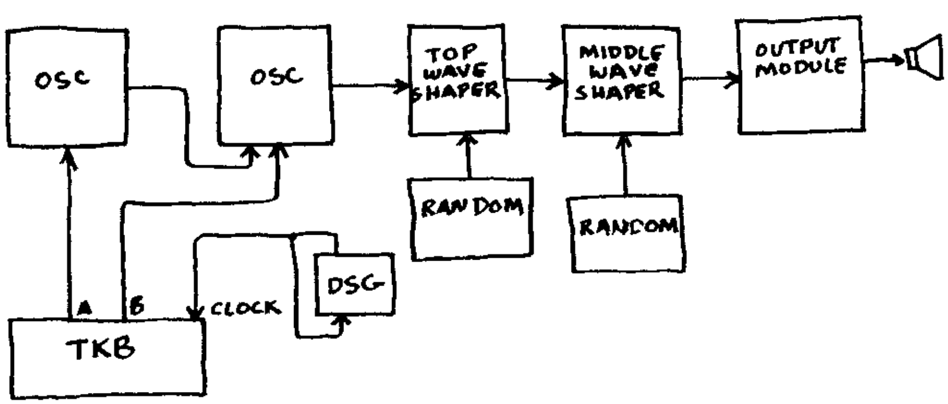
RESONANT EQUALIZER (EQ). An Equalizer or Comb filter is a bank of band pass filters that covers the entire spectrum and whose outputs are mixed together such that the amplitudes of each filter can be controlled. With this device the sound as a whole can be adjusted and balanced to suit. The Resonant Equalizer has ten bands with each band's output being controlled by a pot that is labelled with the center frequency of the band. When the pot is turned to the right, the band it controls is amplified up to about the 3 o'clock position (this is 12 db higher than the input signal). Past 3 o'clock the band is given more and more resonance. If the pot is turned to the left the associated band is attenuated further and further.

The EQ has three outputs. The EQ output sums together all ten bands while the remaining two outputs each sum together alternating outputs (the lower jack outputs 61, 218, 777, 2.8K and 11K Hertz while the middle outputs the other bands).

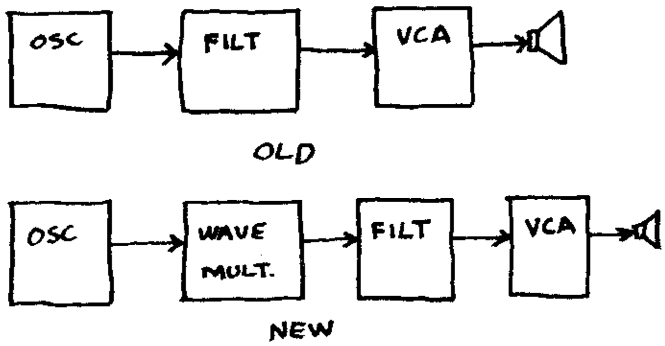
The bands are arranged in sevenths so that a false tonic does not develop. A Level pot at the bottom adjusts the over-all gain of the output and prevents overload when the resonance is set high. These fixed resonant bands are common in almost all timbres produced by musical instruments, and it is the skill of the violin or piano manufacturer in tailoring these resonances that partially determines the quality of the instrument.



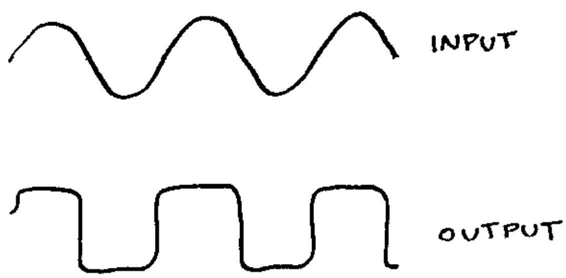
TRIPLE WAVESHAPER. (TWS). The Triple Waveshaper module contains three identical devices which can be used to convert sawtooth waves into sine waves and can provide a wide range of other forms of sound and timbre modification. The timbre can be affected by a manual pot and two different VC inputs which operate on the sound in two different ways. It is a useful module for producing interesting and changing sound timbres, something difficult to achieve in other synthesizers.



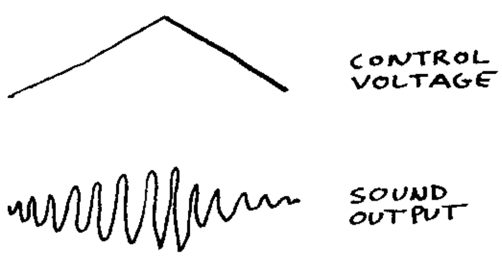
WAVE MULTIPLIERS (VCM). The Wave Multiplier Module is a triple module that, unlike most other multi-modules on the Serge, contains three DIFFERENT modules. Each of the three modules operates on its input in a unique fashion, transforming simple sounds into musically complex and interesting ones. They should not be confused with such devices as Ring modulators which multiply their input signals in a linear fashion -- the Wave Multipliers are highly non-linear in their action. In many ways these modules represent a new node in the typical synthesizer patch.



UPPER WAVE MULTIPLIER: The Upper module of the trio is the simplest of the three. It has a switch for two different settings characteristics. In the HI setting it acts to moderately "square up" or soft clip the signal. The soft clipping is amplitude dependant, producing changes in timbre as the loudness increases.

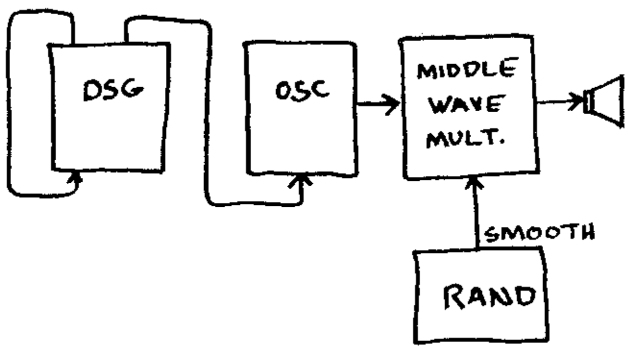


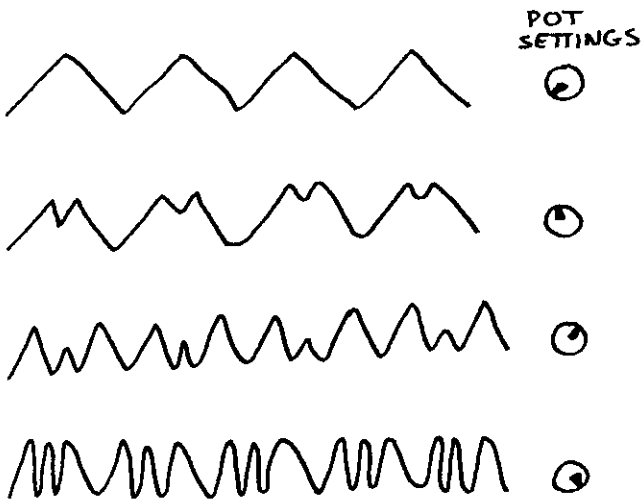
In the LO setting it acts like a linear VCA, a device useful for producing different types of AM sounds.



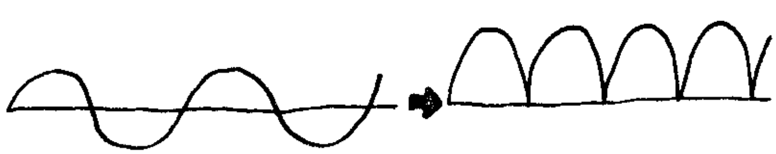
In both settings the module can be controlled either manually or with a VC (control voltage.)

MIDDLE WAVE MULTIPLIER: The Middle Wave Multiplier has two inputs, each producing a slightly different result at the output. One input is DC coupled and has a blue jack. The other input is AC coupled and has a black jack. A sine wave will sound the same when connected to either input, but a triangle wave will produce different effects. These inputs can be used together to provide unusual effects. The general effect of the module is to produce new odd overtones from a sine wave input when the manual pot is turned or when a voltage is applied to the VC input. However, control voltages of complicated natures or inputs more complex than sine or triangle waves can create shimmering bodies of sound somewhat reminiscent of over-blown wind instruments. The VC input can accept AC signals, allowing for complex modulation.





BOTTOM WAVE MULTIPLIER: Like the Middle Multiplier, the Bottom one also has two independent (but identical) inputs. Both inputs are AC coupled. The general effect of the module is that of a full-wave rectifier for audio signals, which means that negative voltages are "flipped" up into the positive.

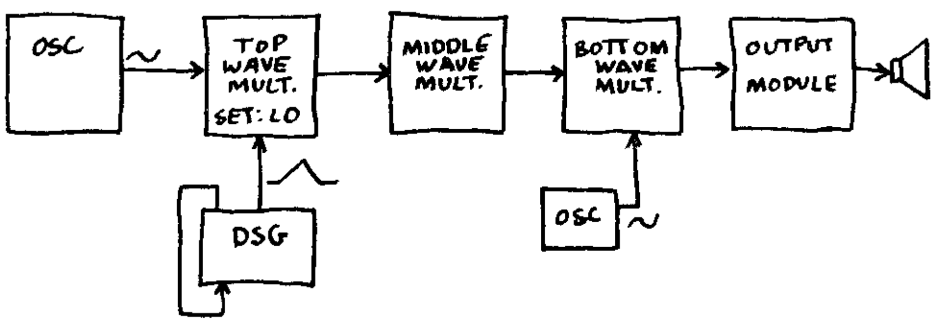


Such a rectified sine wave contains only even harmonics and is one of the few waveforms to contain only these harmonics. The Bottom Wave Multiplier, in actuality, contains three waveform-transforming circuits in a carefully controlled series.

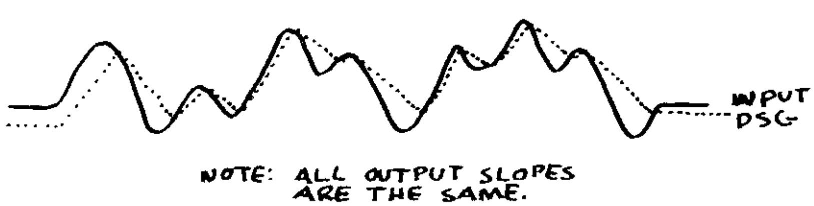
Like the upper two wave multipliers the module provides both manual and voltage control over the output. Unlike its companion modules, however, there are two distinct outputs, OUT 1 and OUT 2. OUT 2 provides a "squared up" version of OUT 1.

One important feature of this lower module is that, unlike simple rectifiers, the amplitude of the output does not decrease through succesive rectifications.

Overall, these three Wave Multipliers provide a method of producing timbres as rich and as varied as acoustic sounds and yet having the precision and repeatablity of analog synthesis.

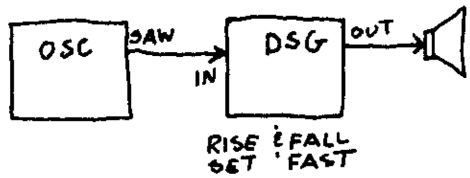


DUAL UNIVERSAL SLOPE GENERATOR (DSG). The DSG can function as a non-linear lo-pass filter essentially by softening the slopes of the IN signal. Generally speaking, the less steep the slope of a waveform the fewer high frequencies it contains. To accomplish this the RISE and FALL times must be set quite fast. Increasing either the Rise or Fall time will increase the filtering action.

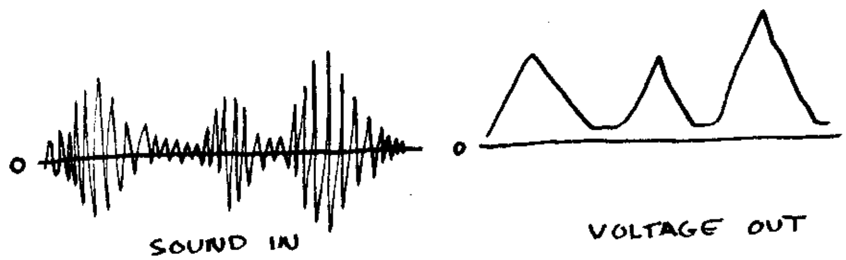


NOTE: ALL OUTPUT SLOPES ARE THE SAME.

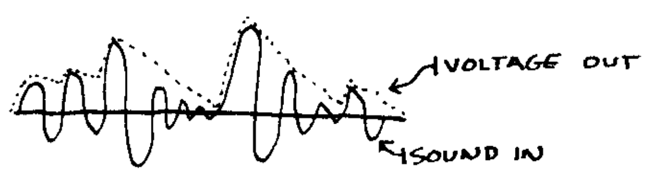
Because the RISE and FALL time on the DSG is voltage controllable, when used in this fashion, the module becomes a voltage controlled filter.



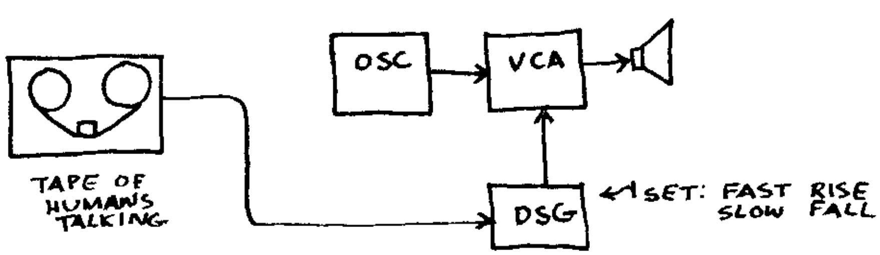
Closely related to this patch is the use of the DSG as an ENVELOPE FOLLOWER. An envelope follower is a device or module that inputs a complex sound and outputs a control voltage proportional to the envelope of the input.



To create an ENVELOPE FOLLOWER it is not desirable to exactly follow the voltage, for that will simply reproduce the wave itself, perhaps with a slight delay or softening of the slopes. Rather, an envelope follower should follow the rising voltages as closely as possible, but have a very slow FALL time. When this is done, the uppper edge of the waveform alone is traced, this being the envelope.

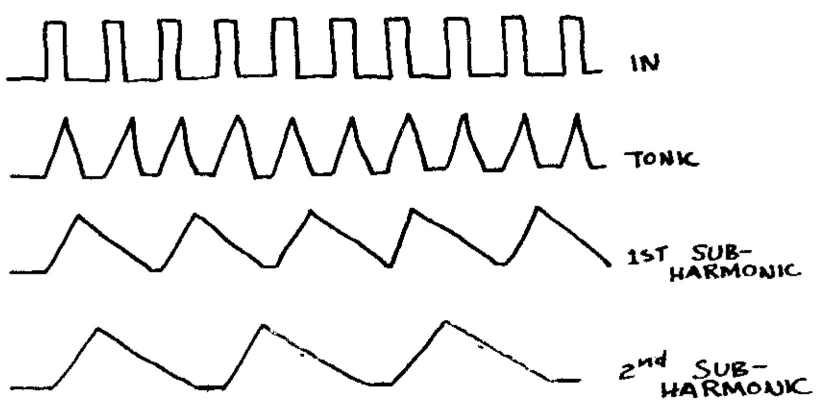


On the DSG this is easily accomplished by using a very fast RISE time and a slow FALL time.



If the DSG is set with a fast FALL time and a slow RISE time, the DSG will follow the negative peaks of the sound. Usually these peaks are almost identical to the positive ones, but not necessarily always. The negative envelope can be used directly to "shut down" a VCA. This can be useful for suppressing backgrounds during solos and for inverting dynamics.

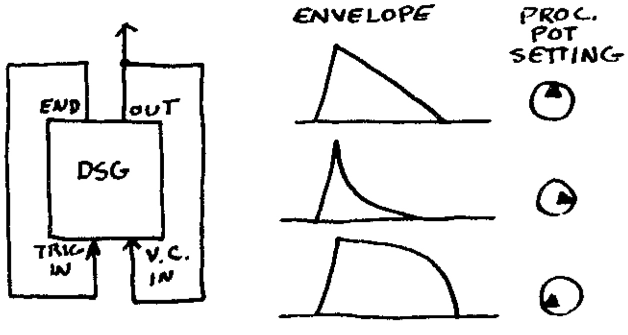
Another use of the DSG is that of an SUB-HARMONIC or UNDERTONE GENERATOR. This is accomplished by applying a very fast pulse train to the TRIG IN and by having the DSG set to audio frequencies. The DSG will not respond to a second trigger until its envelope is complete. If the duration of the envelope is set (manually or with a control voltage) longer than the time frame between the pulses in the train, it will "skip" one (or more) pulses. If it misses a single beat, the frequency is lowered by an octave; if it misses two, the frequency is lowered by an octave and a fifth. Note that this wave has an inverted trapezoidal shape.



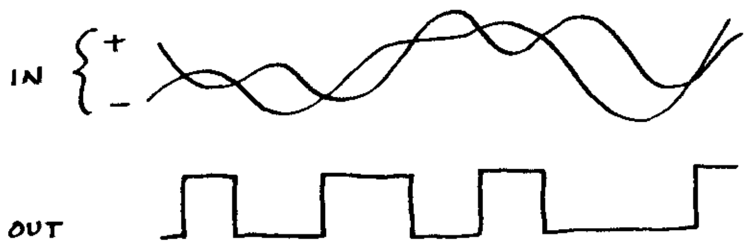
It is sometimes desirable to shape the envelope so that it has a NON-LINEAR SLOPE. this is often the case when producing long sustained sounds or sounds that gradually change in loudness over a long duration. The DSG, when patched to the VCA, will seem to have little effect on the loudness of sound at the start and the end of the long envelopes. This is because of the wide range of the VCA's and the exponential relationship between the voltage and the amplitude.

By "feeding back" the output of the DSG to the VC-IN and setting the processing pot to the left, the final output is made non-linear. This phenomenon occurs because higher voltage to the VC input causes the slope to decrease, so the DSG remains longer at the higher voltage levels.

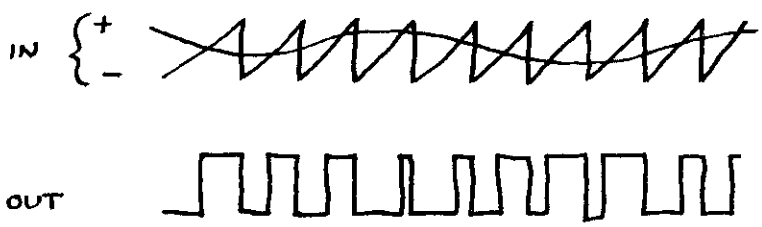
If the processing pot of the DSG is turned to the right, the feedback has the opposite effect: the envelope becomes a sharper and sharper spike, useful for creating short percussive envelopes.



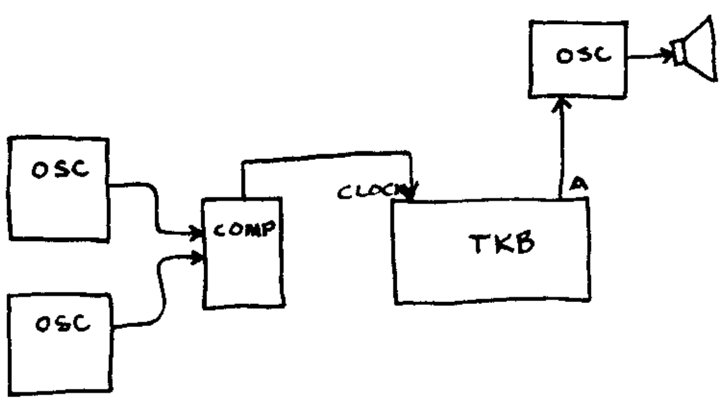
COMPARATOR (COM): The action of this module is this: When the voltage of the "-" input is greater than the "+" input, the OUT goes high to +5 volts. Otherwise the OUT is 0 volts. The pot sets a threshold voltage which is added to the "+" input. If there is no "+" input, this pot alone sets the voltage which the "-" input must rise above for the OUT to go high.



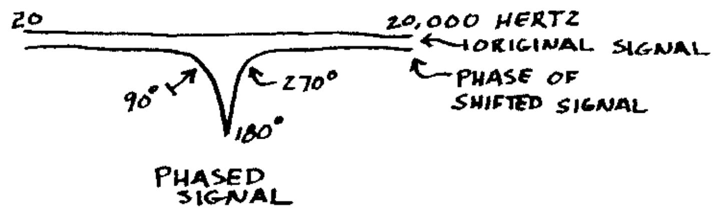
This device can be used to create rectangular waves of various duty cycles. By inputting a sawtooth wave into the "-" input (which is labelled with a "sawtooth" wave to indicate this) and inputting a control voltage into the "+" input, a voltage controlled pulse width generator can be created. These pulses sound like a certain kind of filtering or phasing. A square wave contains only odd harmonics but different rectangular waves contain different harmonics depending on the "duty cycle" (amount of ON time to OFF time). It is interesting to note that rectangular waves with duty cycles of: 1 to 2 have these harmonics missing: 2,4,6,8,10 etc. (a square wave has only odd harmonics); 3 to 1 have missing harmonics 3,6,9,12 etc.; 5 to 1 have missing harmonics 5,10,15,20 etc.

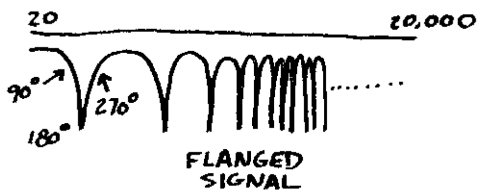


The pulse outputs of the COM can be used to trigger any device on the Serge that requires a trigger pulse.

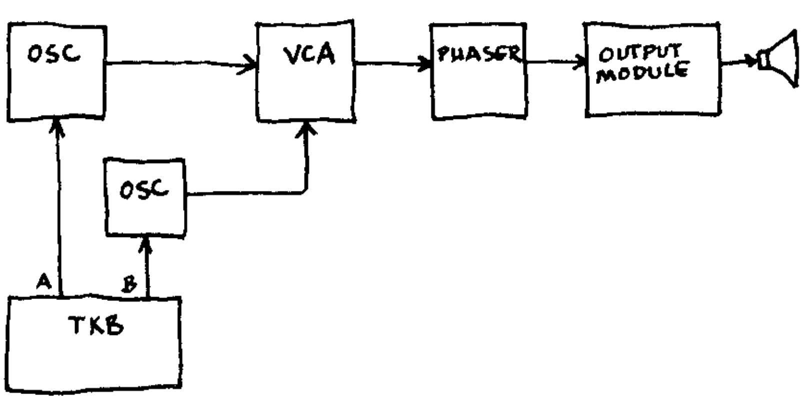


PHASER (PHA). When two identical signals of different phase are mixed together, various frequencies of the signal are cancelled out or attentuated. Which frequencies are cancelled is dependent on the phase relationship between the two signals. "Phasing" is often created when a single sound arrives at the ear at two slightly different times, for instance via two different echoes or two speakers playing the same sound. In these cases one of the sounds arrives slightly delayed because of the extra distance it must travel. In rock and roll this technique is called flanging and was first created by playing the same song on two different tape machines with a thumb on the flange of one of them, slowing it down slightly. Electronic Phasing is a related technique except that instead of delaying the entire signal the delay, and hence the phase shift, occurs only at one specified frequency. This phase shift can be pictured as a well with fairly sharp sloping sides with a phase shift of 180 degrees at the bottom.

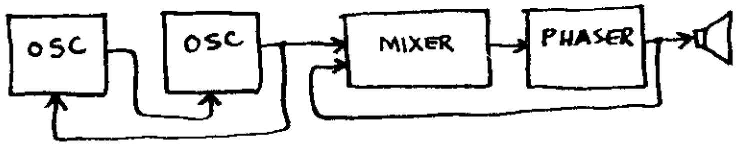




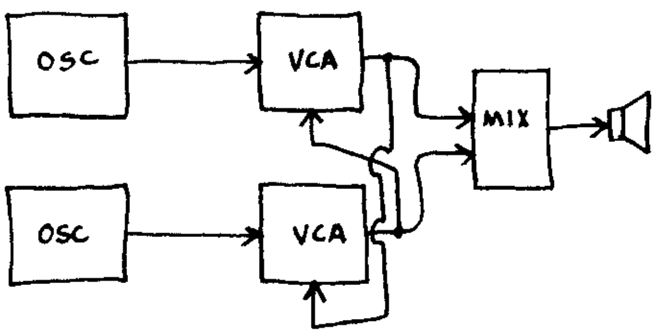
The Serge Phaser is a three-stage phaser providing outputs at each stage (360, 720, and 1080 degrees), however, the characteristic sound of phasing only occurs when these phase-shifted signals are mixed back into the original signal. (The shifted signals by themselves are useful for certain spatial effects, making a sound appear to be moving through space. Our ears are particularly sensitive to phase in relation to position as it is the phase difference between our ears which gives us a clue as to direction.) A mixed output of the Phaser appears at the MIXED output, adjustable by using the manual pot labelled MIX. Full right is the unphased signal, full left is the fully phased signal. The center of "frequency shift well" is determined manually with the PHASE pot and/or with a control voltage (VC in). The VC In can be attentuated by the pot below it. These controls are very precise and log-conforming, allowing the frequency of the Phaser to follow the shifting of its input signal if desired.



This module also comes in a DUAL PHASER Module (2PHA). The upper phaser has 720 degree phased output as well as the mixed out, while the lower phaser has a 360 degree out and the mixed out. If desired, these two modules can be strung together to produce a single, deeper phase shift. It has been found that if the mixed output is mixed back in with the input signal an extraordinarily deep, resonant sound can be produced.



QUAD VCA. (QCA). The Quad VCA contains four independent VCAs each with an audio input, a voltage control input, an output and a GAIN pot. These are logarithmic VCAs and are useful for putting envelopes on sounds, and for Amplitude Modulation.



2-VCA RING MODULATOR

OUTPUT MIXING

For three million years, in fact up until eighty years ago, EVERY sound heard by a man or a woman was integrally associated with its source. The sound of a lion implied a lion, the sound of a snapping twig implied a snapping twig, the sound of somebody calling out your name implied somebody calling out your name. This changed suddenly and irrevocably with the invention of the record player. The sound of a lion could now be a speaker cone vibrating. The sound of somebody calling out your name could be a telephone.
Despite this recent change, we humans still hear sounds as distinct entities. Even though all the sounds in a room combine to form a single complex pressure wave which vibrates our ear drum, we still hear the faucet dripping, the clomp of shoes in the apartment above, the cars wooshing by outside, the conversation in the other room; and if the radio is on not only do we hear music, we can hear the singer, the bass player, the piano, the drums and even something else called the "hiss". All these separate "sound sources" are MIXED in the air to impinge on our ears as a single complex waveform. Our brain easily sorts them out. This ability to sort different sounds is valid even with electronic sounds. A synthesizer can create two different sounds, mix them together, and the ear upon hearing them, can separate them out again. For three million years not only could we tell that a lion was roaring, but we could tell that the lion was roaring over there, that the twig broke behind that bush, that somebody called your name behind you. It was possible to localize the sound in space. While it is important to know that a big cat is around, it is just as important to know where. (You run the other way.) A "sound entity" is located in space by hearing the sound twice, once with each ear. Because sound takes time to move through the air it reaches one ear before the other -- just enough to create a phase difference. The brain can process these phase differences to locate the direction of the sound source. The relative loudness and quality of the sound help to determine the distance of the sound source. A phenomenon called the "Doppler Shift" helps to determine whether the sound is coming or going. Because the brain discovers the direction of a sound by phase difference and distance by relative loudness, location can be simulated with two speakers. With two speakers appropriately placed the brain can locate a recorded lion pacing back and forth. To accomplish this a single sound is PANNED back and forth between the two speakers.

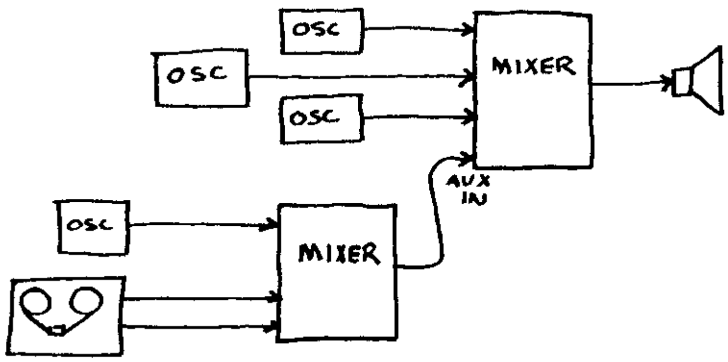
A MIXER is a module that adds together a number of different sounds and sends the mixture to one or more outputs. A MIXER can be categorized by the number of inputs and outputs it has.

A PANNER takes one sound and sends it to two or more different speakers, fading from one to the other. This creates the illusion of movement.

A CROSSFADER takes two sounds and smoothly mixes between them so that as one sound decreases in amplitude, the other increases.

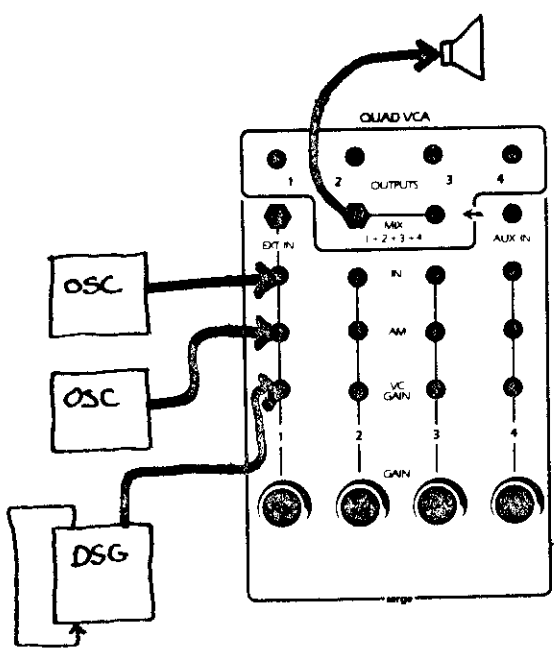
The Serge system offers a wide variety of voltage-controlled Output Mixers, many of which also contain Panners, Crossfaders and voltage-controlled Amplifiers. These mixers also have at least one grounded output jack that can be directly patched to an amplifier. Typically this jack is a Mini-phone, but may be Phone or RCA. This shielded connection output allows the synthesist to run fairly long cords without picking up hum or crosstalk. Many of these Output Mixers also have mini-jacks which allow the synthesist to bring in external sound sources such as tape recorders and pre-ampflified microphones.

DUAL AUDIO MIXER (MIX and MIX2). This module allows a mix of three signals, each with its own level setting with an associated pot. IN-2 can accept either a grounded input from the external world or a signal from another Serge module, but not both at once (connecting to the mini-jack will disconnect the banana jack input). Each of the two mixers on this dual module has an auxiliary input, IN-4. This is a "unity gain" input. The input signal is mixed with the final output. These auxiliary inputs can be used to create a 6-IN 1-OUT mixer.

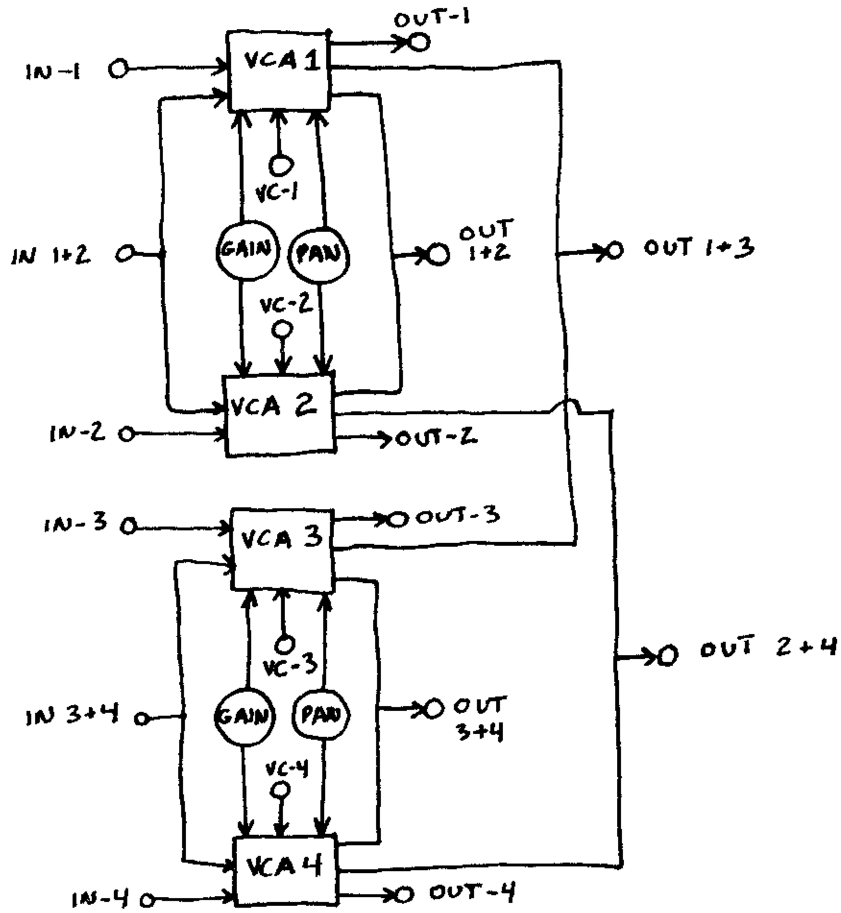


QUAD VCA (QCA). The Quad VCA contains four separate, independent VCAs and provides an additional output Mix of all four VCA outputs. These VCAs are arranged as parallel columns with the outputs at the top of the module. The gain of each VCA can be controlled by a GAIN pot at the bottom of the module or by a VC GAIN voltage control input. This input is used to control the overall level of the sound with control voltages such as envelope generators and TKB outputs. On each VCA there is an AM control input of a lower sensitivity for amplitude modulation of the signal. VCA #1 also has an EXT IN, a mini-phone jack input that can accept external signals.

The MIX output is a mix of the four outputs of the VCAs and has both a Banana output for use within the Serge and a mini-jack output so the signal can be exported to other pieces of equipment. An AUX IN is provided, which is a unity gain input that is mixed into the final 1+2+3+4 MIX. This input is useful for creating larger configurations of mixers.



UNIVERSAL EQUAL POWER AUDIO PROCESSOR (UPAP). At the heart of the UPAP are four VCAs which, though they can be used separately, can also be used as dual VCA units, providing various panning, crossfading and mixing functions. Because of the range of uses it is the most space-effective module for a small Serge System. Below is a block diagram of the module.



Each VCA can be used separately with signal input at 1,2,3 and 4 respectively. A signal to input 1+2 will be sent to VCAs 1 AND 2 for panning and an input to 3+4 will be set to VCAs 3 AND 4. These "dual" inputs can be mixed with the individual inputs to the VCAs.

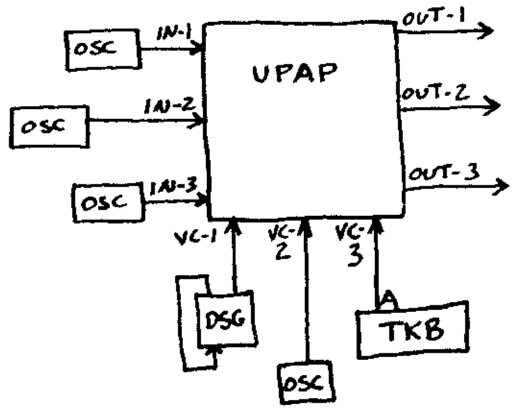
The output of each VCA appears individually at outputs 1,2,3 and 4 respectively. There are four mixed outputs: 1+2, 3+4, 1+3 and 2+4. Outputs 1+3 and 2+4 have, in addition to their banana jacks, mini-phone jacks to be used to send the output to external equipment. These are the usual left and right stereo outputs.

Each individual VCA has its own VC input located directly beneath its signal input. The overall gain of VCA 1 & 2 can be controlled by a single pot labelled 1+2 and its associated VC input. There is an identical configuration for VCA 3 & 4.

Lastly, the gain of the two pairs of VCAs (1 & 2 and 3 & 4) can be controlled by the Pan/Fade pots and their associated VC inputs such that, if set properly, the combined gain of the two remains the same during a cross-fade between them.

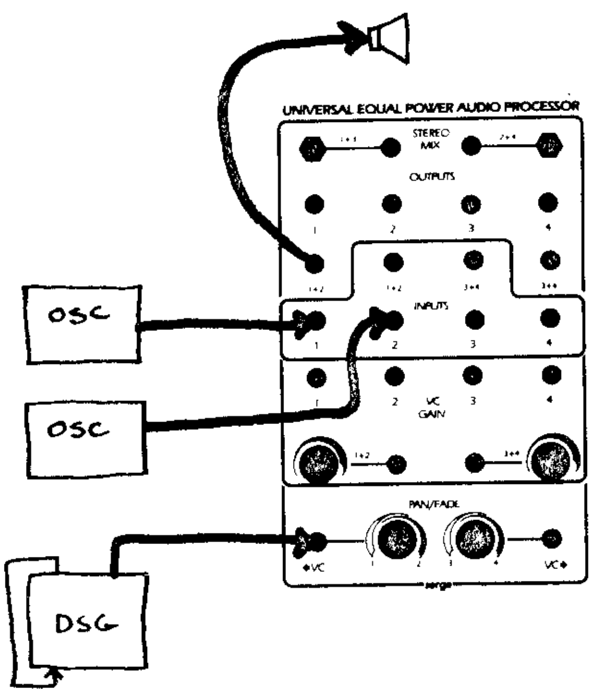
PATCHES:

1. Four VCAs:



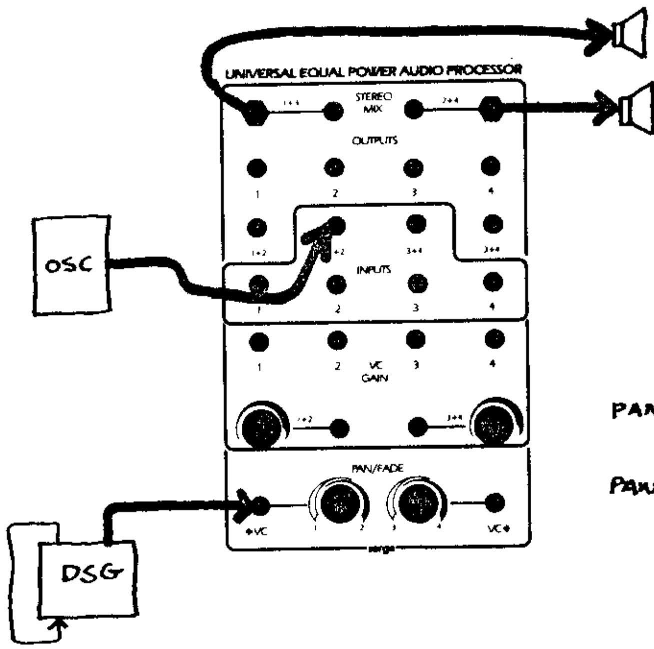
NOTE: PAN/FADE POTS SHOULD BE SET TO 12 O'CLOCK.

2. Two Crossfaders:



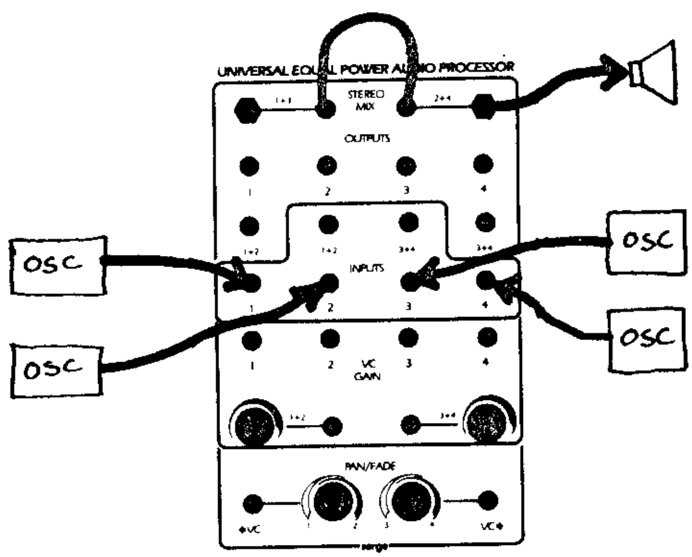
PAN/FADE: 10 O'CLOCK
IF 3+4 USED:
PAN/FADE: 2 O'CLOCK

3. Two Stereo Panners:



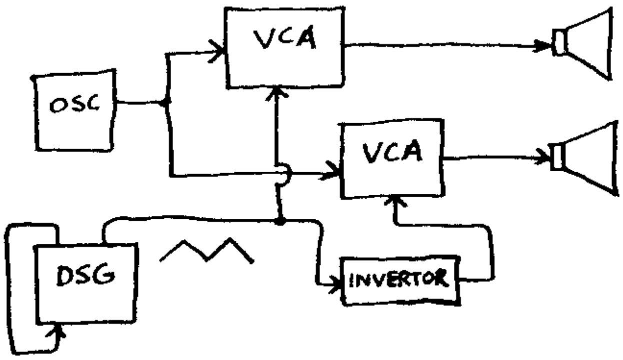
PAN/FADE: 10 O'CLOCK
IF 3+4 USED
PAN/FADE: 2 O'CLOCK

4. Four out mixer:

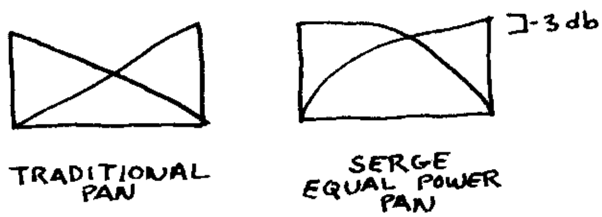


A Note on the PAN/FADE:

The PAN/FADE Pot and VC control allows the synthesist to achieve an equal power pan between two speakers. The typical pan in early synthesizers was achieved by using two VCAs, one controlled by a linear envelope, the other controlled by the inversion of that envelope:



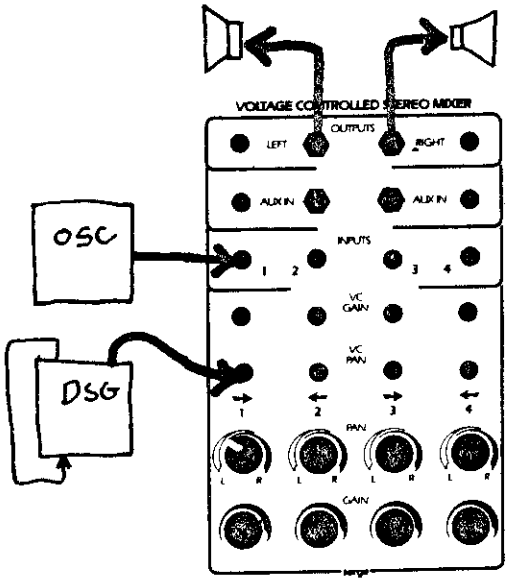
This technique caused a severe dip in overall loudness of the sound at the half-way point. The PAN/FADE control on the Serge corrects for this problem by keeping the total power (and perceived loudness) equal as the sound moves from speaker to speaker. This technique uses sophisticated circuits to compute the amplitude of the sound at each speaker. At the center point the amplitude of each speaker is down exactly 3 DBs. Similar circuits are provided in the quad and octo panners to provide multi-channel equal power panning.



To use the VC input to achieve a full pan, the Pot should be set, on the 1-2 Pan, just to the right of the position where all the sound is sent to speaker #2. A 5 volt peak envelope will then sweep the sound across to speaker #1. For Panning between output #3 and #4, the pot should be set just to the left of the position (all the sound is in speaker #4). The arrow underneath the VC input indicates the direction which a positive voltage "moves the pot".

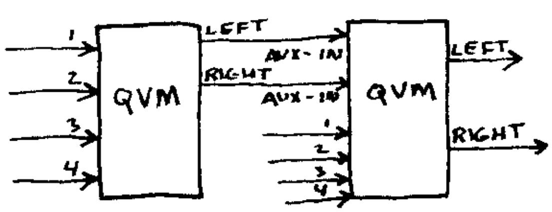
If the pan/fade pot is set beyond these cut-off positions, it will increasingly override the control voltage action. For example, if pan/fade pot 1-2 is turned full left, nothing will be heard from output #2 regardless of other VC inputs.

VOLTAGE CONTROLLED STEREO MIXER (QVM). The QVM is a four-in, two-out mixer. Each input can be individually gain controlled by a GAIN pot and a VC gain input. Furthermore, each input can be directed to either or both of the two outputs by means of a pan pot. This panning function can also be voltage controlled by the VC Pan input. The arrow beneath the VC PAN input indicates the direction of the pan upon receipt of a positive voltage. Like the UPAP this is an equal power pan.



POT SETTING FOR TWO CHANNEL PAN.

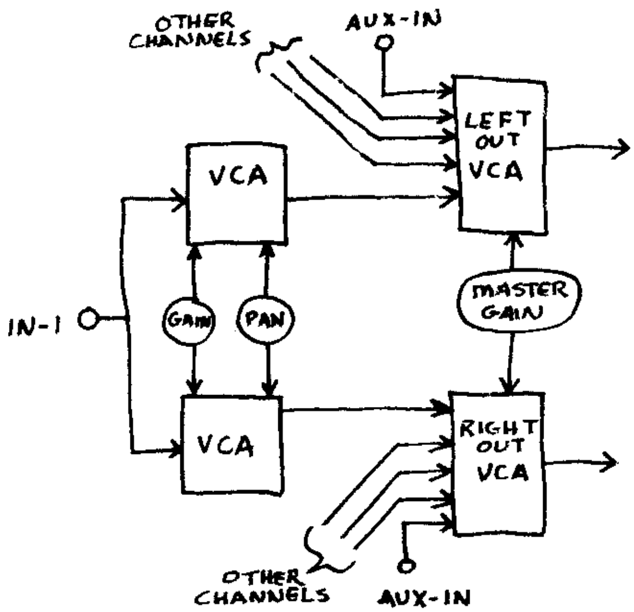
The QVM has auxiliary unity gain inputs for each of its two output channels. These can be used to create an 8-in 2-out mixer by patching the outputs of one QVM into the AUX-INs of another QVM. The AUX-INs also have a mini-jack input which can be used to bring external signals into the Serge.



8 IN - 2 OUT MIXER

The QVM provides a mini-jack output for both the right and left channel, allowing the QVM to be used as an output mixer.

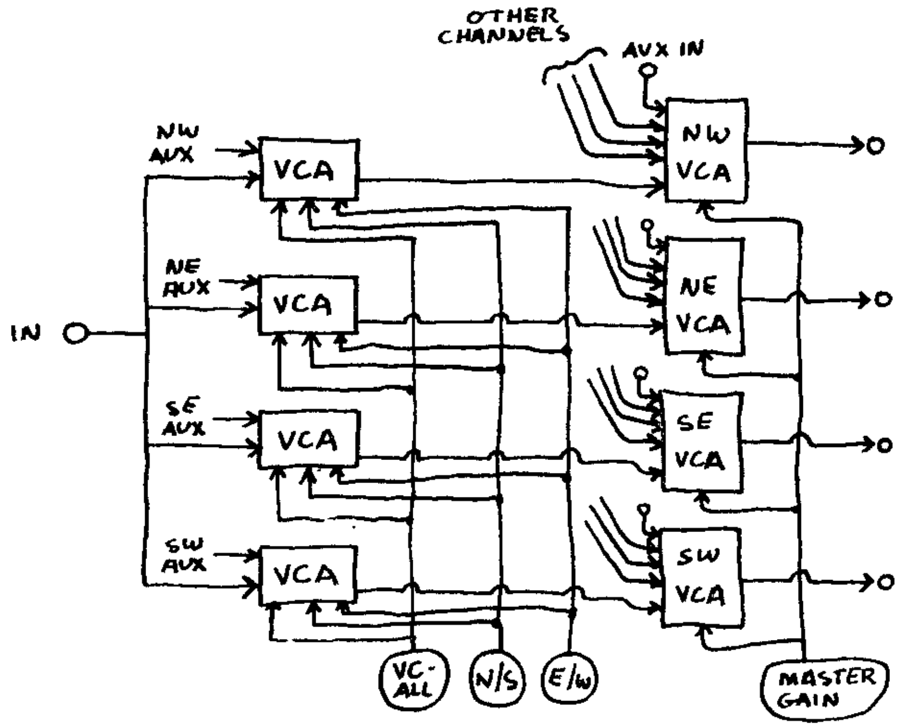
MULTI-CHANNEL STEREO MIXER (SMX).The SMX is a 2X input, 2-out mixer where X is the number of Dual Stereo Panner modules. Typically there will be three or four of these input modules. Each input channel is identical to the input channels of the QVM with individual gain control pot and VC input; and pan control (also in both manual and voltage control modes). Each channel has both a banana input and a mini-jack input to be used with external signals. The Stereo Output Module is identical to the QVM's output section in that it has a left and right output and a left and right auxiliary input (both of which have both banana and mini-phone jacks). In addition, the Stereo Output Module has a Master Gain Control which can attenuate all input signals simultaneously with either manual or voltage control.



1 CHANNEL OF SMX

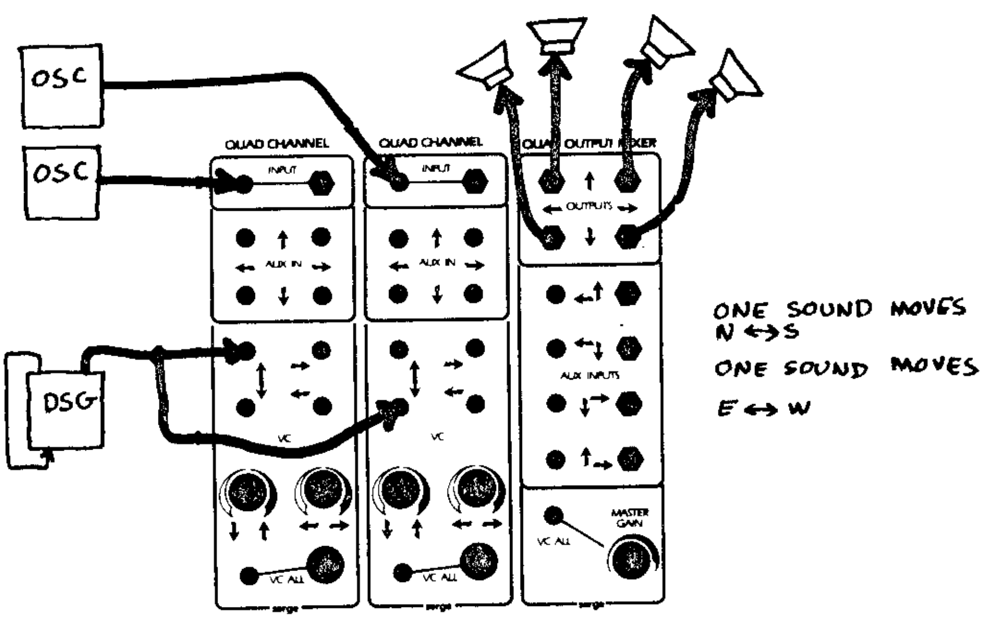
MULTI-CHANNEL QUADRAPHONIC MIXER (QMX). The QMX is an X-input, 4-output mixer where X is the number of Quad Channel inputs (from 2 to 7). Each input channel can be sent to any of the four output channels, either with manual or voltage control. Each input channel has four AUX-INs which are dedicated inputs to each of the four outputs. Each input's overall gain can be controlled either manually or with a control voltage to its VC-ALL.

The Quad Output Mixer module contains four mini-jack outputs and four AUX-INs, one for each of the outputs. There is also a master output gain control that can simultaneously attenuate all four channels, manually or by voltage control to its VC-ALL.

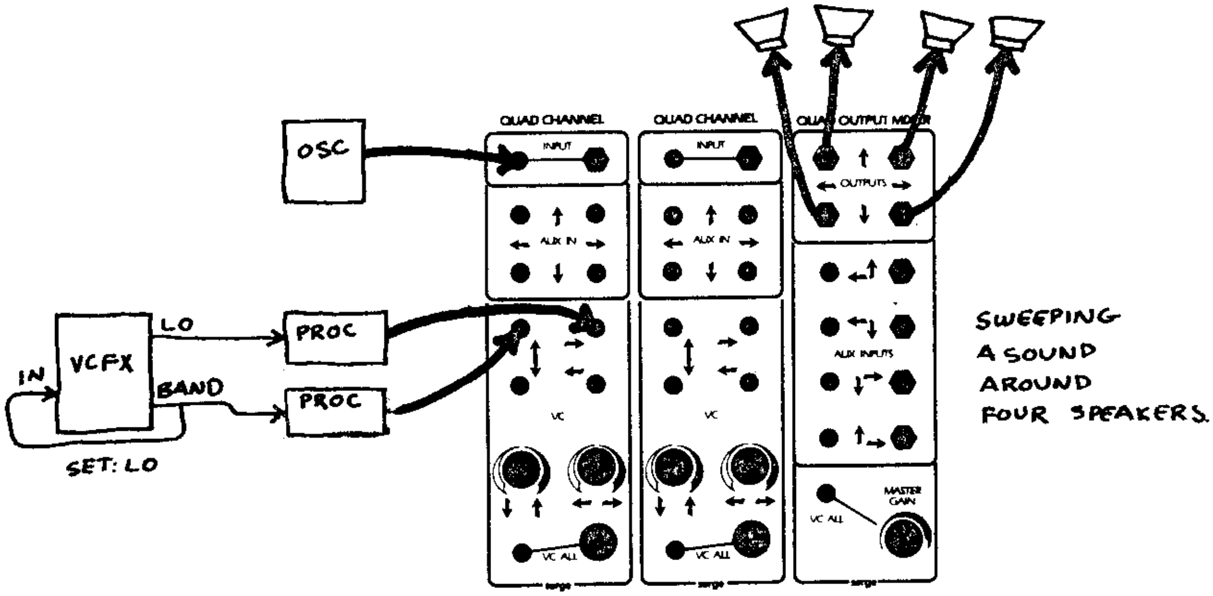


1 INPUT CHANNEL OF QMX

Each input has two panning controls: front to back, and left to right. These panning controls are identical to the panning controls on the UPAP, QVM and the SMX (see these modules for a description of how to effect perfect equal power pans) with one addition. Each of these two pans contains an extra Inverting input that allows a positive voltage to pan in the opposite direction.

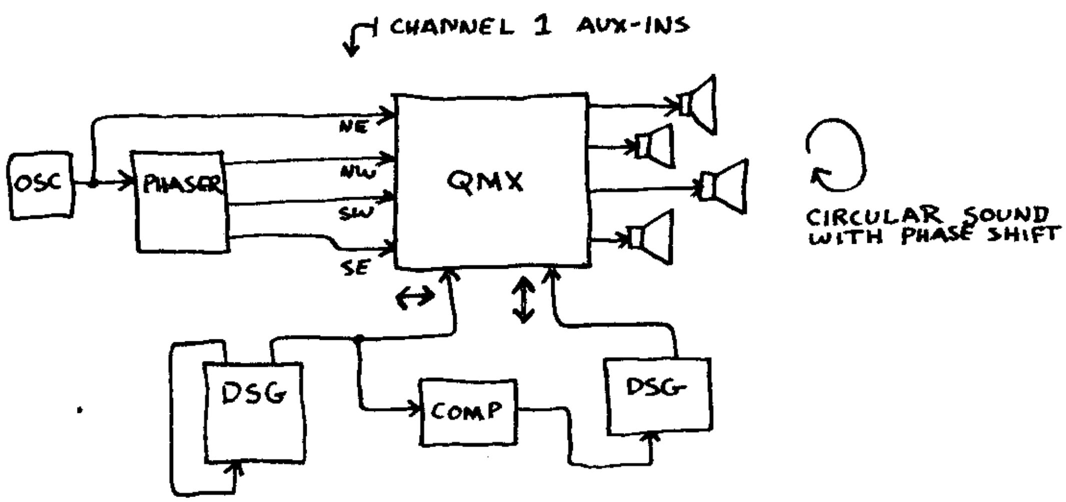


To achieve a perfect rotation of a single sound, two sine waves of identical frequency, 90 degrees out of phase are required. These can be obtained using the VCFX filter by patching the Band output to the IN and setting the filter to its low range. Sine waves 90 degrees out of phase are available at the Band and Lo outputs.



The high quality of the QMX allows rotation of a sound at audio frequencies providing a "spatial modulation" (SM).

Sounds patched into the AUX-INs of the input module will only appear at their assigned outputs and only when that output is gated open by the panning controls.



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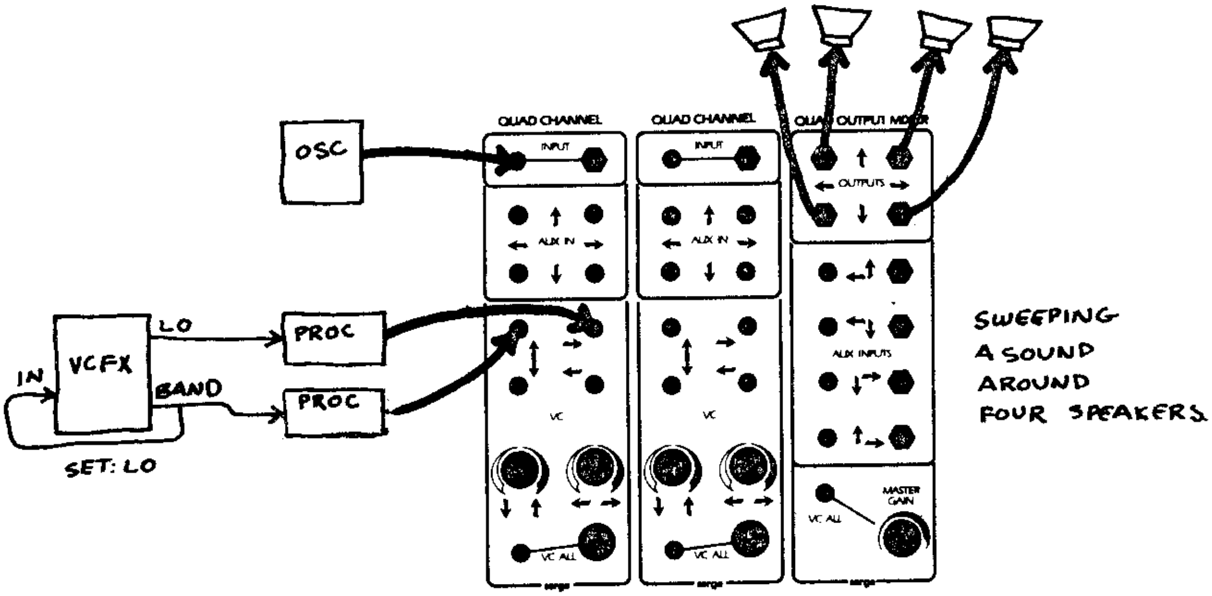

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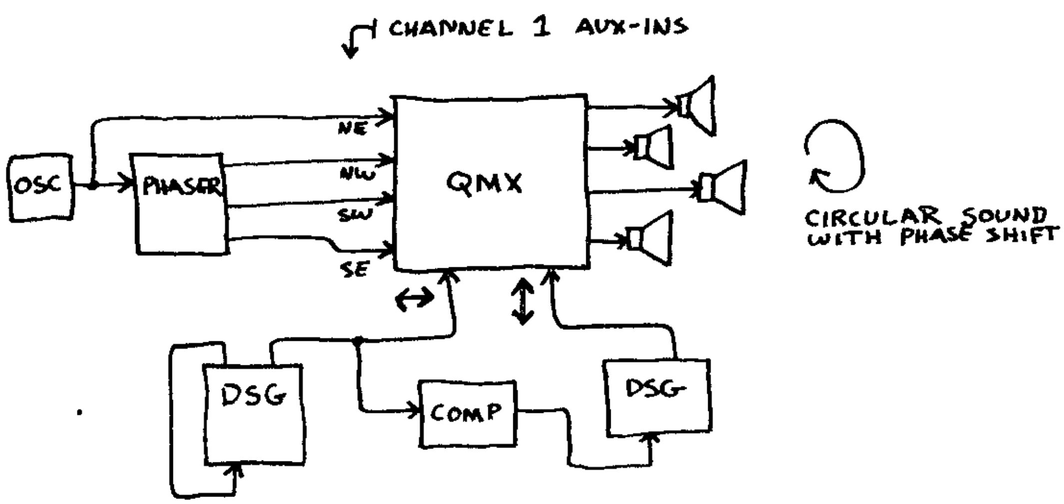
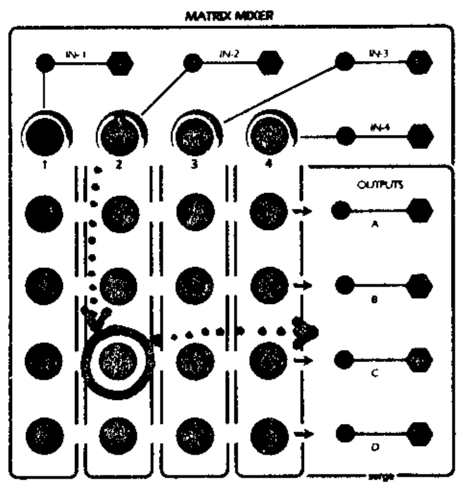


STEREO OUTPUT MIXER (MXP). The MXP is a 4-in, 2-out manual mixer that provides individual manual control of gain and output selection (panning) for each of its four inputs. It also has two AUX-INs that allow the mixer to accept external signals (using the mini-jack inputs) or to be converted into a 8-in, 2-out mixer using 2 stereo output mixers patched together.

MATRIX MIXER (MAX). The MAX is a 4-in, 4-out manually controlled mixer. The level of each of the four inputs at each of the four outputs is determined by the GAIN pot located at the intersection of the appropriate input column and output row. The faceplate below has that pot circled which controls the level of input #2 to output C.

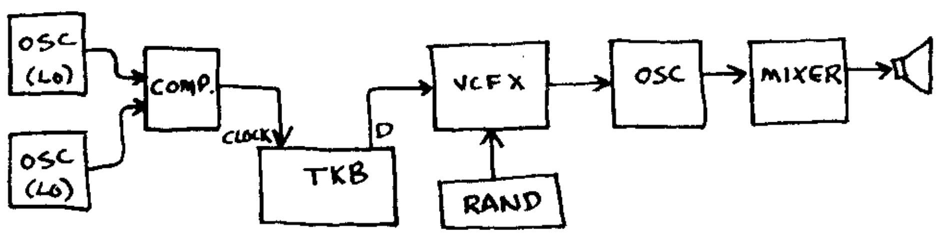


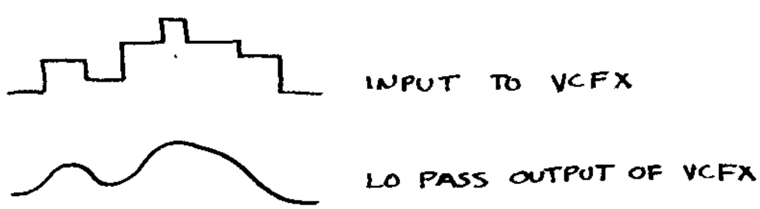
Each input has an overall gain control so that that input can be attenuated equally throughout the mix.

CONTROL VOLTAGE PROCESSORS

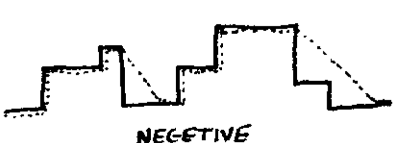
Because there is no difference, other than frequency, between audio voltages (AC) and control voltages (DC) it should not seem too strange that it is possible to have modules which process control voltages in much the same way that there are modules which process audio voltages. Nor should it seem odd that these modules for processing control voltages are themselves voltage controllable. These modules extend the range of shapes and forms that control voltages can have, and thereby extend the possiblities of control. It is the control voltage processors which "mold" the control voltages that determine the complex dynamic shifts so important to interesting electronic music. The simplest of these devices have already been explored--The control voltage Processors that are found on the control voltage inputs of many modules on the Serge system. These processors enable the user to amplify, attenuate and/or invert the control voltage.

EXTENDED RANGE VCF. (VCFX). When the VCFX is in its LO, setting it can be used to filter out the harmonics of sub-audio waves, that is, of control voltages. Like audio frequencies, sub-audio frequencies can be described in terms of the sum of Sine waves of given amplitudes. The VCFX can be used to filter out these waves either in a HI, LO, BAND or NOTCH filter mode. The LO pass output will sound as if the filter is smoothing the control voltage. Note that this smoothing function is voltage controllable since the frequency of the oscillator is controllable. The lower the frequency the smoother the control voltage becomes. In the following patch the smooth random is determining the filtering of the output of the TKB.



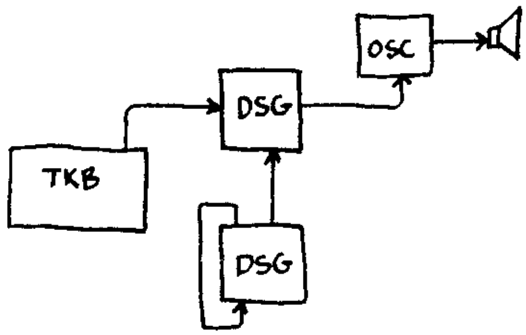


DUAL UNIVERSAL SLOPE GENERATOR (DSG). The DSG can act as a Positive or Negative or Positive/Negative SLEW or Portamento device. A Slew is a device which slides from one voltage to another voltage; a Positive slew affects positive-going voltages (not just positive voltages, but changes in a positive direction), and a Negative slew acts on negative-going changes.



When both positive and negative slews are present the device is often called a Portamento or Glissando device.

Because the DSG will Rise or Fall to the voltage IN, it can be used as a Positive and/or Negative Slew limiter. The slower the Rise or Fall time setting, the more that parameter acts like a slew. A Positive slew, for instance, would have a slow Rise time and a very fast Fall time. Because the DSG is voltage-controllable, it is a voltage-controllable slew.

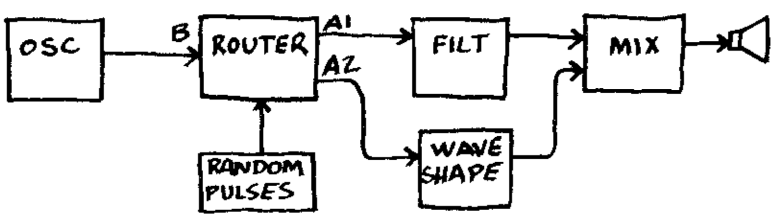


BI-DIRECTIONAL ROUTER. (ROU). This unique triple module can be used in one of two ways.

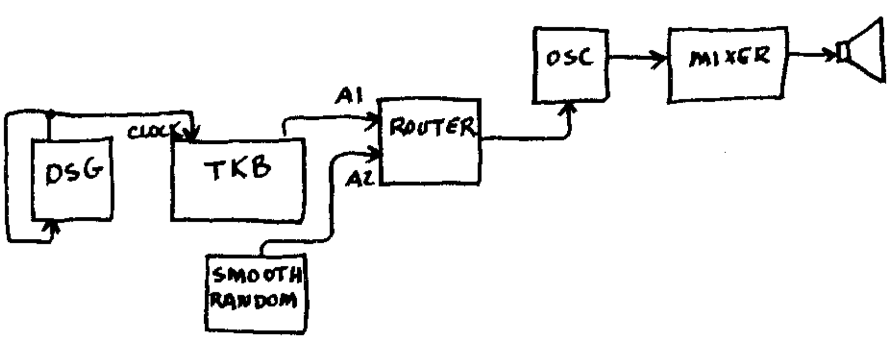
ONE-IN TWO-OUT SWITCH. An input at B can be sent to either output A1 or A2 depending on the state of input A1-A2. If A1-A2 is HI (+5 volts) then B appears at A1, otherwise it appears at A2.

TWO-IN ONE-OUT SWITCH. If there is an input at A1 and a second input at A2 then A1 will appear at the output B if A1-A2 is HI, otherwise A2 will appear at the output.

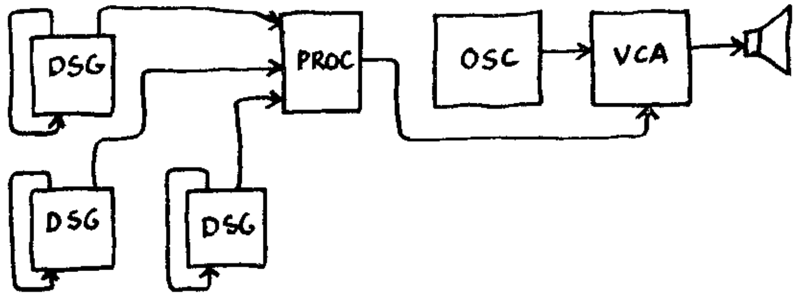
Note that if there is an INPUT at B there CANNOT be an input at either A1 or A2. This would, in effect, short outputs together.



While in the above patches audio voltages are being routed about, control voltages can be routed in much a similar way. However, anytime there is an instantaneous change in voltage there will be a click.



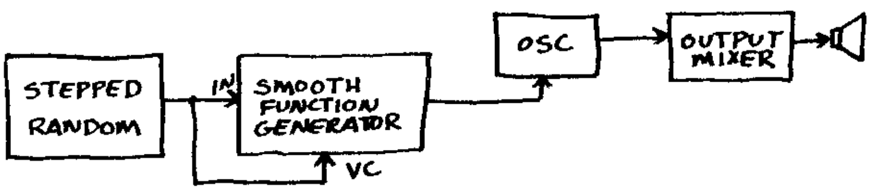
DUAL CONTROL VOLTAGE PROCESSOR (PRC). The Processor is to control voltages what a Mixer is to audio voltages. It can be used to sum together up to three control voltages. Each of its three inputs can independently be attenuated, amplified and/or inverted. Furthermore a manual offset pot sets a fixed voltage that can be added into the mix. This offset voltage is available at the output of the Processor even if there are no other inputs.



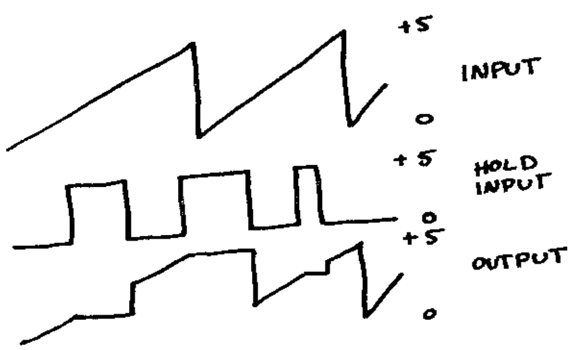
SMOOTH AND STEPPED FUNCTION GENERATOR. (SSG). The SSG is a dual module that contains two different and independent modules (though an internal coupler is provided). These are very versatile modules that can be patched to process control voltages in a wide variety of ways.

THE SMOOTH FUNCTION GENERATOR

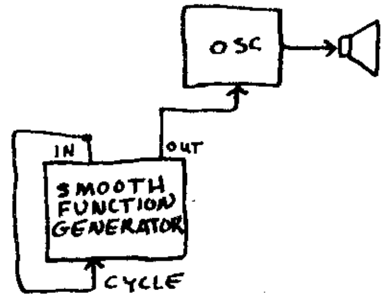
SLEW LIMITER. The Smooth Function Generator serves as a voltage controlled Slew Limiter on its Input. The slope of the slew (both positive and negative) is determined by the manual RATE pot and a VC input with associated attenuation pot.



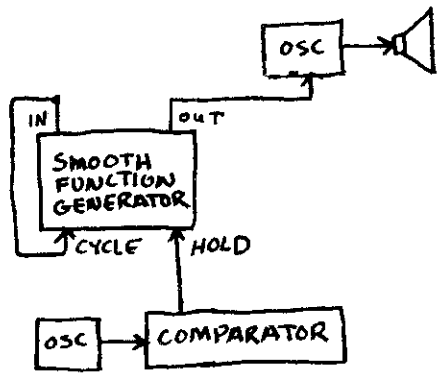
TRACK AND HOLD. A Hold input is provided on the Smooth Function Generator. When this input receives a HI voltage it has the effect of HOLDING the present output voltage level until the HOLD input goes low. If the RATE is set very fast so that the Smooth Function Generator follows its input closely (that is, Tracks the input), the receipt of a pulse train at its HOLD will produce a Staircase-like series of voltages.



LOW FREQUENCY OSCILLATOR (LFO). If the input of the Smooth Function Generator is patched to the CYCLE jack, the output is a triangle wave whose frequency is determined by the RATE and control voltages. The CYCLE output will be a series of pulses with the same frequency as the triangle wave output.

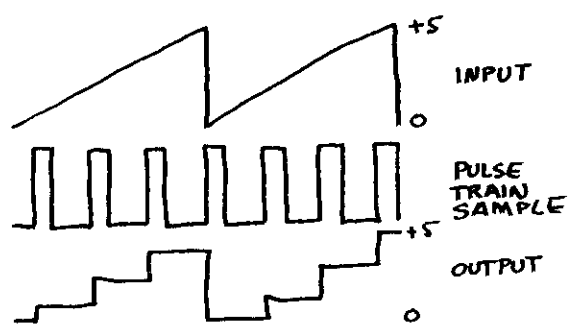


When the Smooth Function Generator is patched to cycle,the HOLD function remains operative to be able to produce up and down staircase-like voltages.

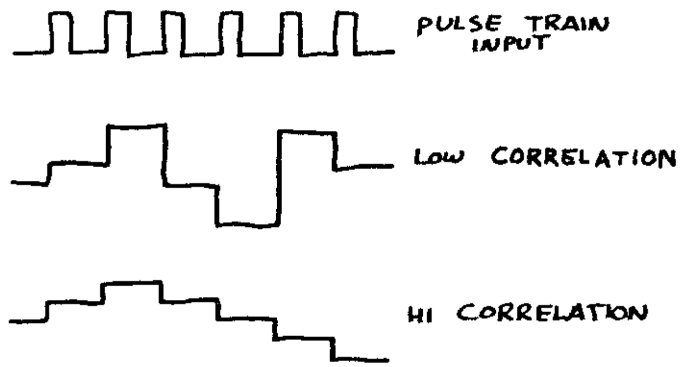


STEPPED FUNCTION GENERATOR

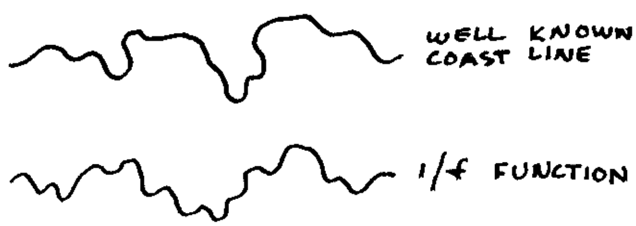
SAMPLE AND HOLD. A Sample and Hold is a device which produces a discrete stepped waveform from a changing input voltage. When a pulse is received at the SAMPLE input, the voltage appearing at that instant at IN appears at STEPPED OUT and is HELD there until another pulse is received at SAMPLE, when this process is repeated.



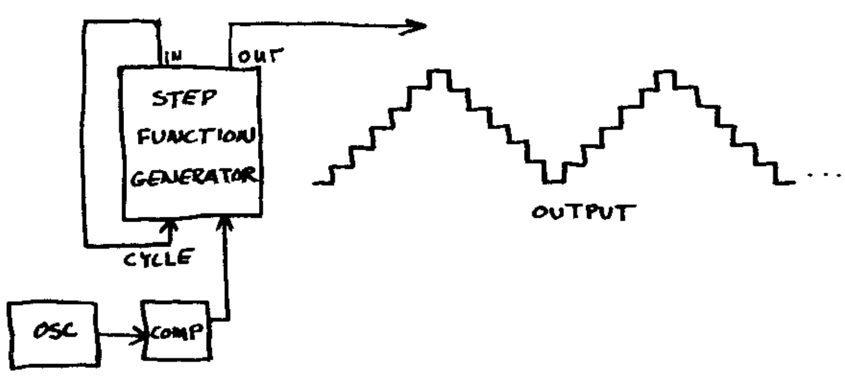
CORRELATION. The RATE pot and the VC input with its associated attenuation pot control the "correlation" of one voltage output level to the previous voltage output level. In the stepped voltage as correlation increases, each step must be closer and closer to the previous step.



When the RATE pot is at a middle position, and the input is a random voltage such as the S/H Source on the NOISE module, the output approximates the function called 1/F. 1/F is a random-like function that describes the shape of natural coastlines, cloud movements and many kinds of music.



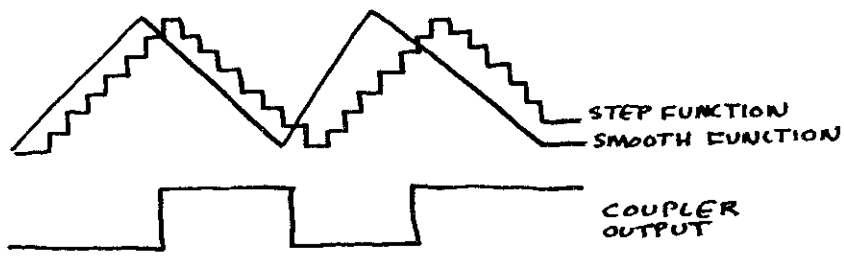
STAIRCASE GENERATOR. When the input is patched to the CYCLE jack and pulses applied to the SAMPLE, a complex staircase wave is generated at STEPPED OUT, determined by the pulse frequency, the position of the RATE pot, and a VC.



RANDOM VOLTAGE GENERATOR. It is sometimes desirable to create random voltages to control the various devices on the Serge, or, if the system already has one random voltage generator, it is sometimes desirable to have a second. Using the Coupler output on the SSG and the S/H Output on the Noise Source module it is possible to Patch the SSG to become a random voltage generator.

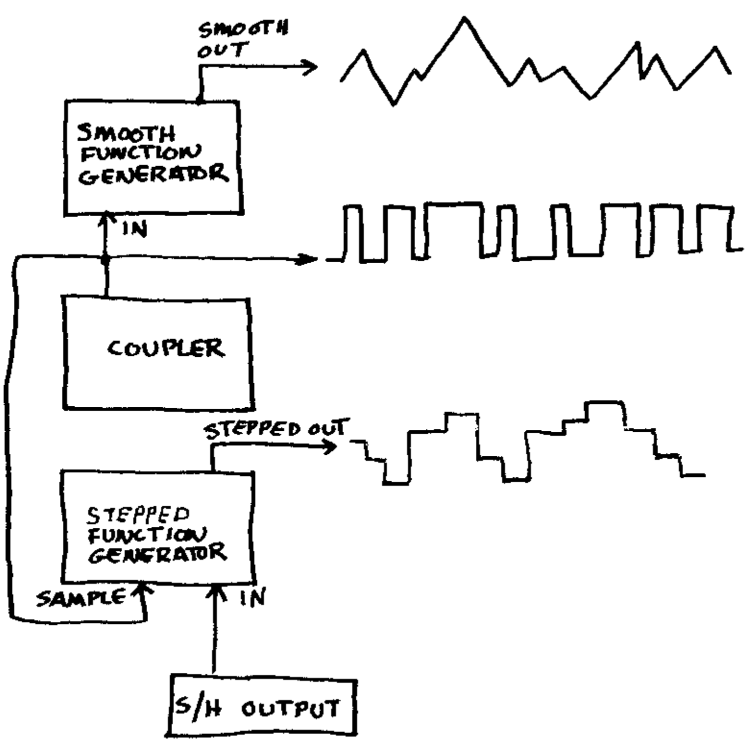
COUPLER

The Coupler is a comparator which is hard-wired (that is it is not patchable, but is pre-wired beneath the panel). The Coupler compares the levels at the Smooth and the Stepped outputs. Whenever the Step Function Generator is HIGHER in voltage than the Smooth section, the output of the Coupler goes HI. Otherwise, the coupler is LO. The coupler has two identical outputs.



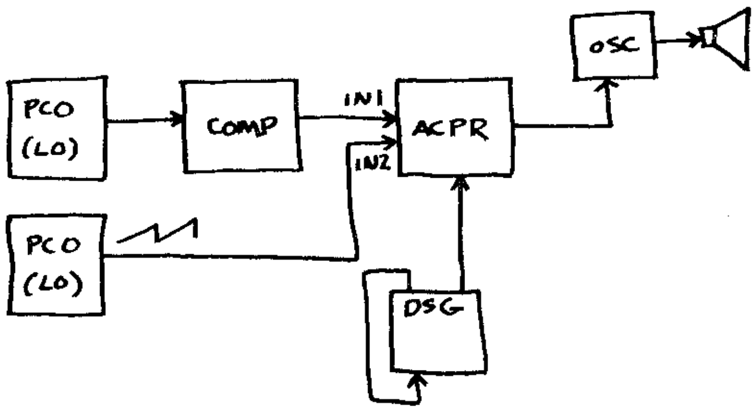
The S/H Output of the Noise Source is a randomly modulated sawtooth wave. This form of wave is good for generating random voltages with sample-and-holds. Since this wave is always going from 0 volts to +5 volts, there is an equal probability for any voltage to be selected. Since the frequency is random, it is impossible to predict what voltage will be sampled.

The S/H Output is patched to the input of the Stepped function generator, whose RATE pot is turned full right (low correlation). The output of the Coupler is used as the input to the Sample on the Stepped Function Generator. This same coupler output is used as the pulse INPUT of the Smooth Function generator, whose own rate is set fairly slow. The output of the Smooth Function Generator is now a continuous random voltage; the output of the Stepped Function generator is a stepped random voltage while the output of the coupler is a random pulse output with random on-times as well as onset times. The rate of change is set by the Smooth Rate pot and its associated VC.

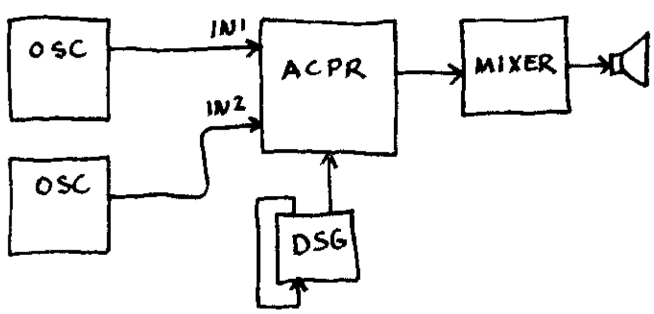


ACTIVE PROCESSOR. (ACPR). This module contains two separate and different modules. The lower module is a simple processor that allows the user to invert a control voltage and apply an offset. This module is useful for processing control voltages sent to modules without processing inputs.

The upper module is the Active Processor. It has two inputs, IN1 and IN2. The output of the module is a mix of these two inputs. The VC input and the manual pot control this mix. A cross-fade between IN1 and IN2 is done by turning this pot from full left to full right and/or by applying 0 volts to 5 volts to the VC. This allows a smooth change between control voltages, for instance, from a sequence to a random voltage control of an oscillator, or between an envelope and its inversion. When the pot is at 12 o'clock an equal mix between IN1 and IN2 appears at the module's output. This module can have the effect of multiplying control voltages. A voltage at IN2 will be multiplied by the VC if the knob is set full left.



This module is able to accept AC signals and can act as a linear VCA as well as a linear cross-fader between two audio signals.

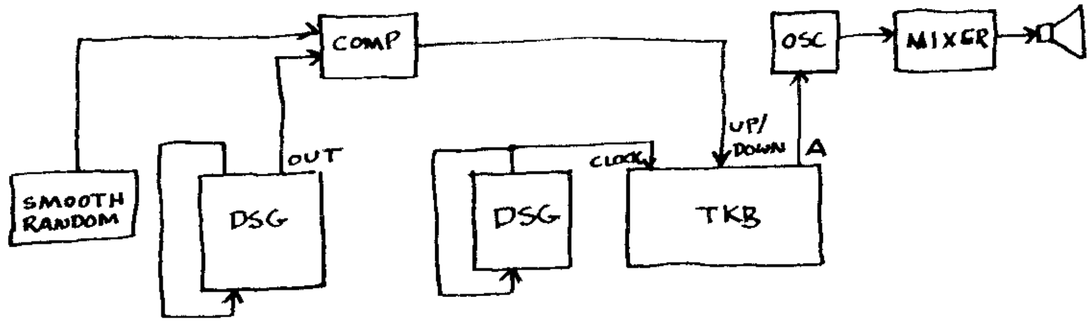


TOUCH ACTIVATED KEYBOARD SEQUENCER:(TKB). It is useful to think of the TKB as composed of two separate parts: The Sequencer and the Touch Pad Controller.

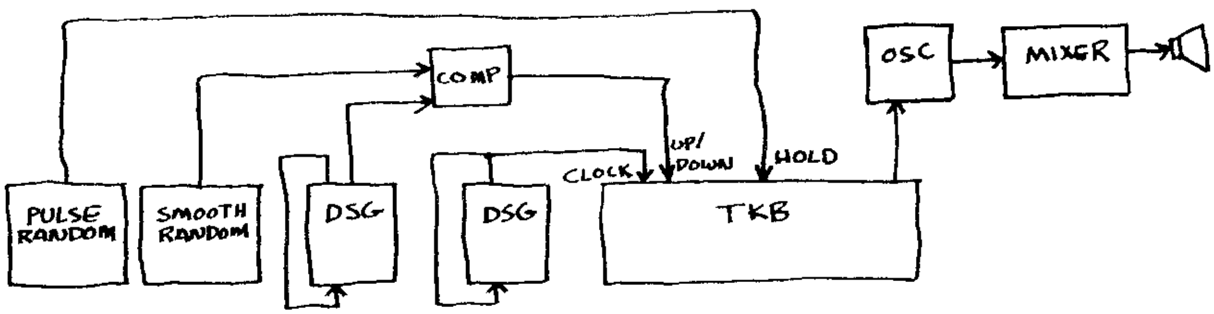
16-STAGE Sequencer. Only one stage can be on at any given instant. Each stage controls a specific column of pots, so that if stage #7 of the Sequencer is on, it activates all four pots in stage #7.

In its normal mode the Sequencer will advance one stage every time it receives a CLOCK pulse. That is, if the Sequencer was on stage #5 and it receives a CLOCK pulse it will advance to stage #6. If it is on stage #16 and receives a CLOCK pulse, it wraps around and activates stage #1. This function was described in an earlier section. There are, however, a number of other ways of controlling the Sequencer to produce elaborate musical patterns.

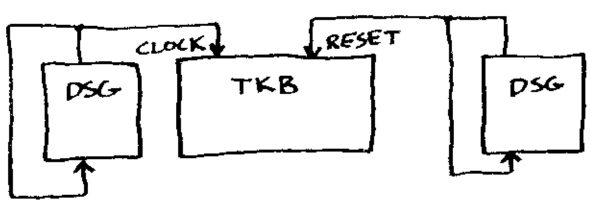
UP/DOWN. If a HI voltage is applied to the UP/DOWN input, the Sequencer will step DOWNWARDS instead of upward when it receives a CLOCK pulse. If it is on stage #1, it will wrap-up to stage #16.



HOLD. If at any time (either in its up or down mode) the Sequencer receives a HI voltage at its HOLD input, the Sequencer will stop until the HOLD input again drops LO. This is useful for producing elaborate rhythms.

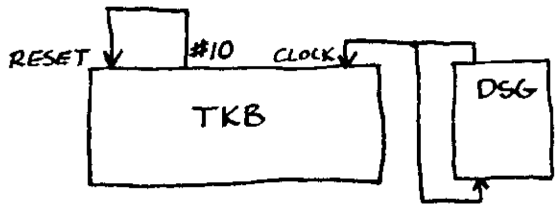


VARIABLE LENGTH SEQUENCES. It is often desirable to have sequences shorter than sixteen stages, or to have variable length sequences. For these purposes two RESET inputs are available at the top of the TKB. The RESET is triggered by a pulse from other pulse outputs on the TKB or by pulses from other modules.



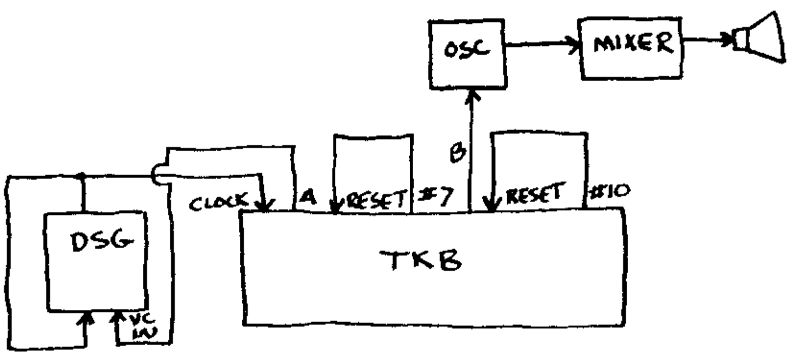
In the above patch, to get the TKB to RESET to Stage #1 you must first touch Keypad #1. The number of stages clocked advanced before resetting is determined by the Transient generator which is Triggering the RESET. Set this Transient generator so that about four stages are clocked through before RESETting occurs. If you touch a different keypad, say #7, you will find that the sequence resets to that stage. Each RESET input resets to the Keypad last touched. Using the above patch, and by touching different keypads, it is possible to produce an interesting interactive sequencer.

Above each stage of the Sequencer is a Pulse output that goes high when that stage is ON. One of their uses is to Trigger the RESETs providing a second way of producing sequences shorter than 16 stages. If the pulse out of stage #10 is patched to the RESET, the sequence will step through to Stage #9 and then RESET instead of activating Stage #10. Note that the sequencer will sequence only to the stage just preceding the stage that is patched to the RESET and will not include that stage.

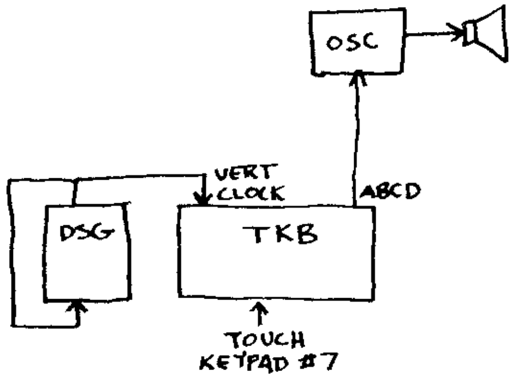


In this patch, just like in the previous one, the sequence will return to stage #1 only if that was the last keypad touched, otherwise, it will reset to whatever pad was last touched. In the above patch you can get the sequence to activate stages 5,6,7,8,9 and then reset to 5 by tapping stage 5's keypad. By tapping different keypads different length sequences can be "played", each ending at stage 9.

The two RESET inputs on the TKB are independent of each other though identical in function. By using them both, two different sequences can be set up and chosen by a tap of the finger. In the patch below the Pulse out of stage 7 is sent to one RESET input and the pulse from stage 15 is sent to the other. By touching keypads 1-6, sequences are activated that start with the touched key and terminate with stage #6. By touching keys 8-14 similiar sequences, but ones that terminate with stage #14 are set up. With a touch of a finger they can be selected. That touch also chooses the beginning stage of the sequence.

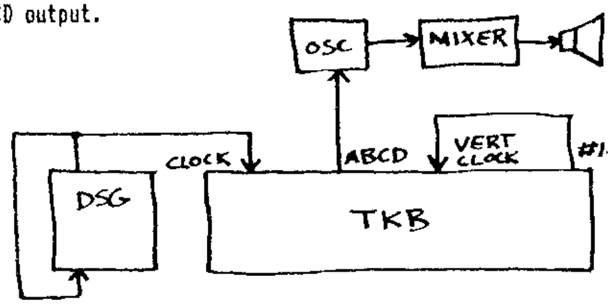


VERTICAL SEQUENCER. The TKB contains a second Sequencer, this one in 4 stages, which is clocked independently of the main 16 stage sequencer. The clock input for this sequencer is labelled VERT CLOCK. The VERT Sequencer's output is labelled ABCD. Every time a Trigger is received by the Vertical Sequencer it steps DOWN one ROW. That is, if it was on Row B, it will progress to Row C. After Row D it wraps around to Row A. The output at ABCD is determined by the pot that is in the activated Stage of the Programmer (as determined by the main Sequencer and the Keypads) AND in the row specified by the Vertical Sequencer. If the activated stage of the Programmer is not changed, then the Vertical Sequencer will have a four stage sequence set by the four pots in the activated stage.



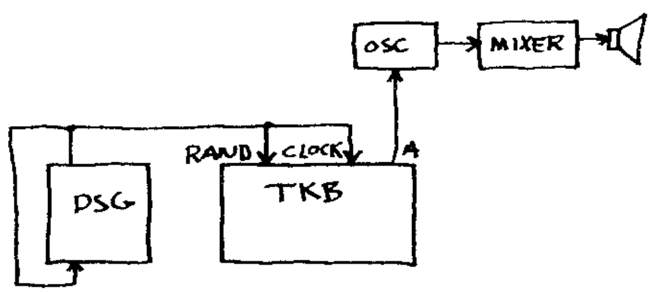
If in the above patch, stage #7 is now activated by touching its associated keypad, the four pots in stage #7 will determine ABCD's output.

A particularly useful function of the this Vertical clock is that it allows the user to produce sequences of up to 64 stages. This is done by having the main sequencer clocked by an external clock and having the Vertical Sequencer clocked by the Trigger out of stage #1. The 64 stage output is found at the ABCD output.



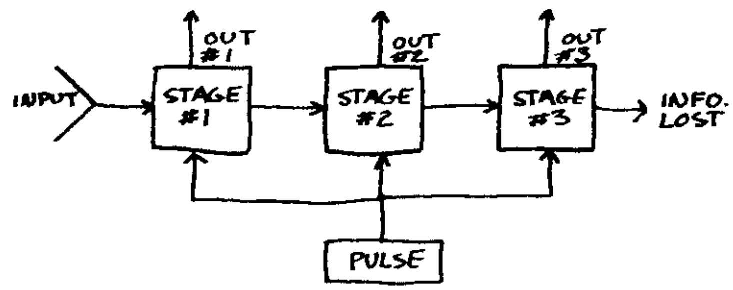
Consider what happens when the vertical sequence starts on Row A. Since the main sequencer clocks through all sixteen stages, all of Row A appears at output ABCD. When the main Sequencer gets to Stage #1, its pulse out, steps the Vertical Sequencer down one row, to Row B. The Main (horizontl) Sequencer now wraps around and clocks all the way across, with the pot settings in Row B appearing at the ABCD output. This will continue, stepping to row C then to row D until the 64 stages have been sent to the ABCD Output. The sequence will then repeat itself.

RANDOM SELECT. If the Random Select is pulsed at the same time as the RESET, the sequencer will reset to a random stage.

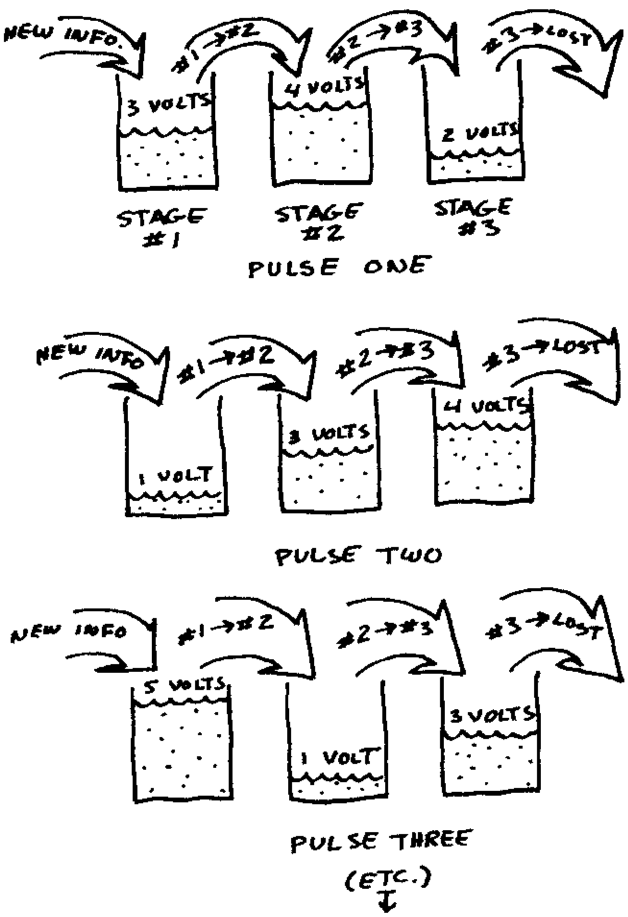


TOUCH PADS. The 16 Touch Pads can be used to interact with the 16-stage Sequencer, or can be used independently. Some of its interactive functions have been discussed already. It is important to note that the KV, KP, and PRESSURE outputs are always selected by the touch pads and not the Sequencer except when the Sequencer is in the Random mode. when the KEYS Switch is ON, the Touch Pads will also turn the associated Sequencer stage on. If the KEYS switch is OFF, then the Sequencer and Touch Pads will be totally independent unless the RESET inputs are used. Both the KV and PRESSURE outputs produce the full range of control voltages, from 0 to +5 volts. The KV output is equal-voltage steps, so with the proper processing, a 12-note equal-tempered scale can be set on the oscillators and filters. Of course, other equal-tempered scales can also be set.

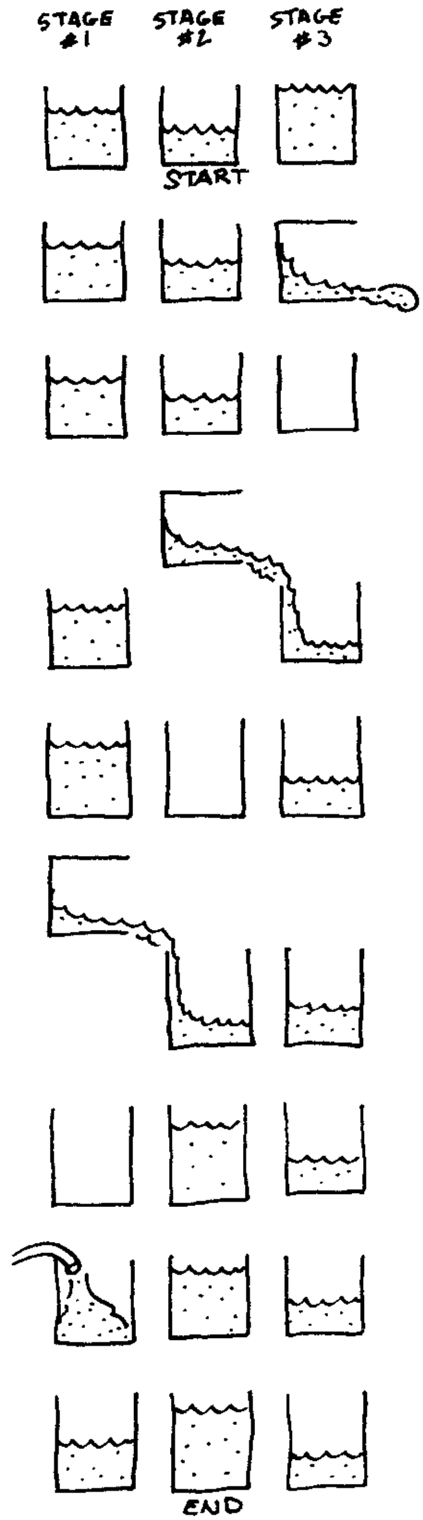
ANALOG SHIFT REGISTER. (ASR). An Analog Shift Register is a sequential sample and hold device. It is also referred to as a Bucket Brigade Delay. The Analog Shift Register available on the Serge has three stages, though two or more modules can be linked to provide longer chains. Each stage of the ASR can hold and save a voltage which is available at all times at outputs 1,2 and 3. When the device receives a Trigger at PULSE IN the voltage stored in stage #1 is moved to stage #2; the voltage at stage #2 goes to #3 and the voltage at #3 is lost. Stage #1 picks up the voltage found at SAMPLE IN. It is this action which gives the device the name of the bucket brigade.



OVER-ALL ACTION

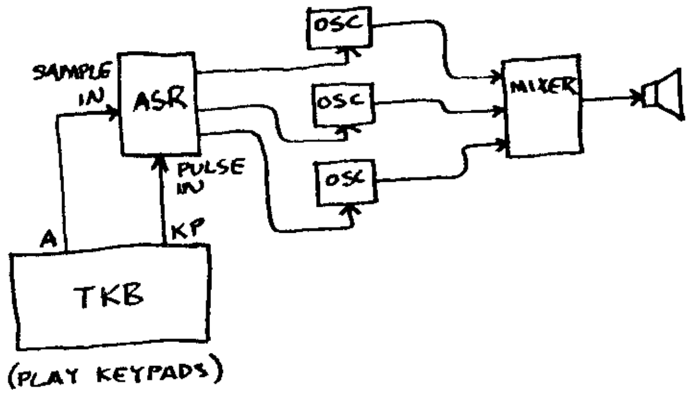


ACTION WITH A SERIES OF PULSES

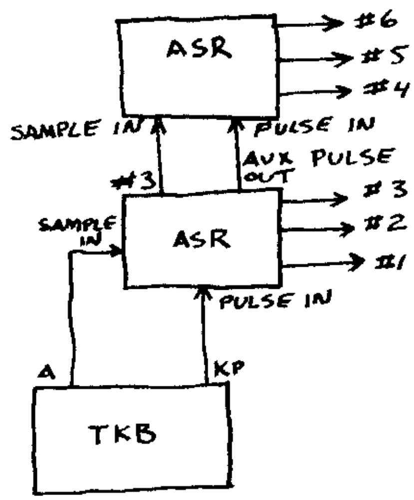


WHAT HAPPENS AT EACH PULSE

One of the many uses to which this device can be put is that of producing chords and arabesques from a mono-phonic keyboard input. The following patch illustrates this use.



Two ASRs can be joined (to produce a single ASR with 6 stages) by pulsing the second ASR with the AUX PULSE OUT from the first ASR and sending OUTPUT #3 of the first ASR to the SAMPLE IN of the second. Such a delay is useful for long time lags.



PORTAMENTO-IN on NTO. This input is a voltage controlled slewing processor. It has the effect of glissing or slewing between stepped voltages. The resultant smooth function is summed with the other VC of the NTO to determine the NTO's frequency. The rate of portamento is controlled by a pot and a voltage control in. This input is calibrated to a sensitivity of 1V/OCT.

APPENDIX NUMBER ONE

USING EXTERNAL SOUND SOURCES

The Serge Synthesizer can use any sound as an audio signal so long as it has been converted to a varying voltage of the appropriate level. These can be sounds of people talking, instruments playing, airplanes overhead, dogs barking, hummingbirds humming. The Serge can mix, filter, ring modulate, amplitude modulate, and waveshape these sounds. It can also put new envelopes on these sounds or extract envelopes from them to use elsewhere. It can use these signals to frequency modulate oscillators and, with a comparator, check for amplitude peaks to trigger anything from the TKB to envelope generators.

The Serge system has high impedance, line level inputs. This kind of input allows one module to control many other modules without losing accuracy, and is almost impervious to electrical damage. Most Control Voltage inputs are from 50K to 200K ohms. Audio inputs are from 22K to 100K.

Sound sources fall into two broad categories of impedance and voltage level. Line level, high impedance signals, which can be used directly in the Serge and low impedance, low voltage signals which must be pre-amplified before using.

Line level sources include tape recorder outputs, headphone outputs and the Line or Aux output of mixers and pre-amplifiers. There are a few microphones which are also line level, but these are not common.

Low level sources include almost all microphones, instrument pick-ups and record player cartridges. All three of these sound sources must be pre-amplified before they can be used on the Serge.

Available Pre-Amplifiers:

1. The Serge instrument and microphone pre-amplifier.

2. Most portable and studio mixers have microphone and instrument inputs and line level outputs. Often these mixers can also accept line level signals and provide the user with a switch to choose between the inputs.

3. Component stereo systems have pre-amplified outputs and turntable inputs. The line level output may be labelled TAPE OUT, LINE OUT or AUX OUT. This is the most common way to pre-amplify a record.

4. Most tape recorders have a microphone input which will pre-amplify the microphone. The signal can then be taped, and when played back, used directly. Or, the tape recorder can be switched to SOURCE and the input, or source, will appear pre-amplified directly at the output.

5. Small, moderate quality pre-amplifiers can be purchased at electronic stores.

6. Electric guitars and other instruments with pick-ups need to be pre-amplified. Most stage type amplifier "heads" have a LINE OUT which can be used directly to the Serge. Otherwise, small instrument pre-amps are required.

APPENDIX NUMBER TWO

SENDING SERGE SIGNALS OUT

The output levels of all Serge Modules are line level signals and can be sent directly to the LINE, or AUX in of any electronic sound device including pre-amps, amps and mixers. These levels are also appropriate for sending signals to such devices as Reverb units or graphic equalizers. The output impedance of most Serge modules is about 300 ohms.

For tape recording purposes the Serge signal can be sent directly to the Line Inputs of a tape recorder, or to the Line level input of a mixer and then to a tape recorder.

Auxiliary mini-jack inputs and outputs can be found on most Serge mixers and should be used to inter-connect to other equipment to prevent hum, crosstalk and static.

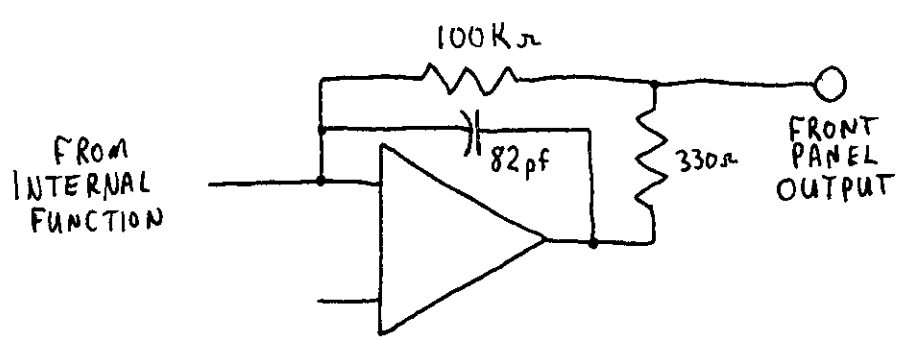
Some inputs and outputs of external devices have "balanced" lines with two signal lines and one ground line. Typically these lines use Cannon Connectors. Many microphones have balanced lines which permit much longer cables to be used before hum becomes noticeable. These lines must be unbalanced before being connected to mini-jacks, unbalanced phone jacks or RCA jacks on the Serge. There are three ways of unbalancing a balanced line.

1. Connecting one of the hot lines to ground.

2. Using a balanced-in, grounded-out transformer.

3. Using a mixer that balances and unblances signals.

To send Serge or Line level signals any distance it is a good idea to use shielded wire. To send low level signals long distances it is advisable to use balanced line. Because it is possible to send low impedance signals extremely far (say 1000 feet) without hum building up, it is often wise when sending line level signals such distances, to use a transformer to convert it to a low impedance balanced signal, with a second transformer at the far end of the line reconverting it back to an unbalanced signal.



TYPICAL OUTPUT BUFFER ON SERGE MODULES

APPENDIX NUMBER THREE

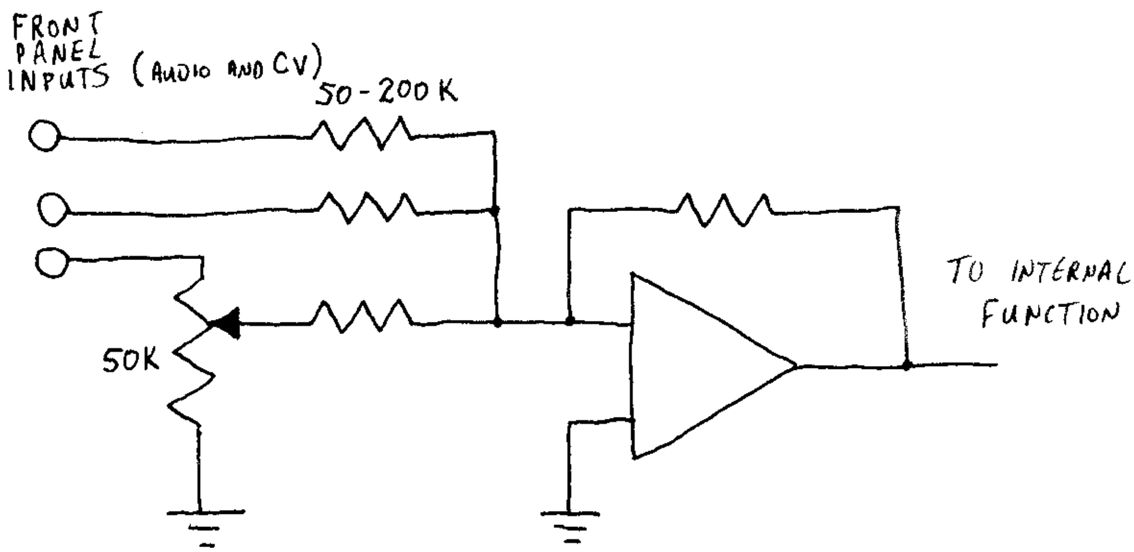
EXTERNAL CONTROL VOLTAGES

Almost all control voltage sources on the Serge such as the envelope generators, the TKB, the Random Control Voltage sources and the non-Sine wave oscillator outputs have a voltage range of about 0 to 5 volts. Using the Processor Module or Processing Inputs this range can be effectly increased to -10 to +10 volts. The great majority of Serge VC inputs respond to voltages in this range, though some only respond to positive voltages. These VC inputs can accept voltages in this range from ANY source including other synthesizers, home built circuits, foot pedals and/or voltage-out keyboard units. In fact, because of the extremely high impedance of the inputs on the Serge, DC voltages greater than 12 volts (up to 25 volts) can be used without damage to the Serge, though this will generally drive the module out of its effective range.

Voltages that are too low can be amplified slightly by using a Processor module and sending the voltage to all three inputs. (This technique also works with audio voltages using either a processor or mixer).

All Frequency dependent modules have VC inputs which operate on a 1 volt to 1 octave ratio. Other voltages that are not of this range can be scaled using Processor Modules or the processing inputs on the modules and then "tuned" by ear.

There are some synthesizers on the market that have exponential control voltages with linearly responding modules. The Serge, and most synthesizers, operate in the opposite fashion. Because of these differences, control voltage generators such as keyboards and sequencers from these other synthesizers cannot be used with the Serge in a meaningful manner.



TYPICAL INPUT NETWORK ON SERGE MODULES

APPENDIX NUMBER FOUR

COMPUTER INTERFACING

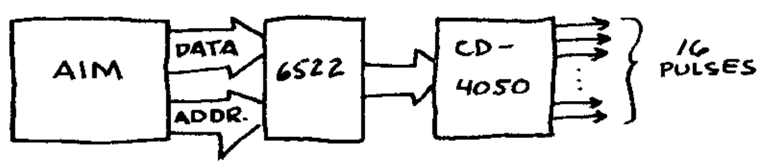
Connecting the Serge system to a small computer requires suitable buffers to prevent the higher voltages of the synthesizer from being applied directly to the computer or its associated circuitry. Most micro-computers have a single +5 volt power supply, and if negative voltages or voltages greater than +5 are received, permanent damage to the computer might result. It is therefore advisable that you be fairly confident of your technical abilities before attemping such an interface. A wide variety of books and magazines is available on computer/synthesizer interface which provide detailed schematics and procedures (see bibliography). There are commercially available interface boards and cards which incorporate these safeguards.

An inexpensive home computer such as the AIM, APPLE, KIM, SYM, RADIO SHACK or OSI is a perfect complement to the Serge System, expanding the possibilities of both. The following list of possible interfaces gives an overview of the most common applications of small computers with the Serge synthesizer.

TRIGGER GENERATOR:

Triggers for the Serge are easy to generate with a computer since they are voltages of only two levels: HI (+5 volts) and LOW (0 volts). These are the values used to represent "1" and "0" on almost all computer hardware. A HI voltage is able to open a VCA fully and sustain any sustainable function on the Serge.

The following schematic is of a simple circuit that will provide up to eight programmable pulses to the Serge. It uses a 6522 Programmable Interface Adaptor which can be used with all 6502 microprocessor-based systems such as the APPLE, KIM, SYM and AIM (and is in fact included on the board of many of these systems.) Note that this circuit uses a CD4050 buffer to protect the 6522 and 6502 from any inadvertent damage from the higher voltages of the synthesizer.



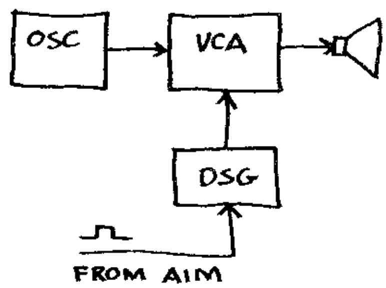
To turn on one output you can use the following 6502 Assembly Language routine:

```
LDA MASK
ORA PORT
STA PORT
```

where PORT is the address of the 6522 port; MASK is an 8-bit word which has one of its bits set to a digital "1" or high value and the rest set to "0". To turn the same output off:

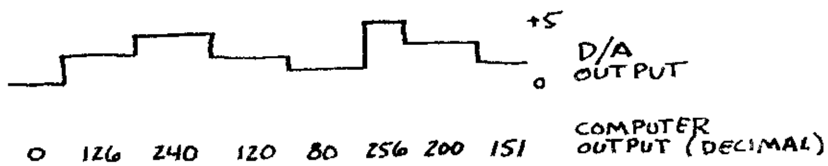
```
LDA MASK
EOR $FF
AND PORT
STA PORT
```

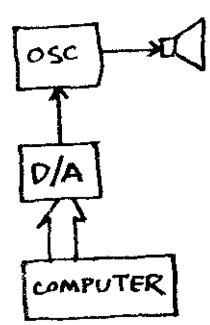
For one of these pulse outs to trigger a Serge device it must rise from a LO level to a HI level (since the Serge is rising-edge-triggered). To trigger an envelope generator repeatedly, the pulse out must return to a "0" level first.



CONTROL VOLTAGES:

A Digital-to-Analog converter is a device that takes a digital number, and converts that number into a voltage level. A sequence of numbers from the computer to a D/A will result in a series of voltages.

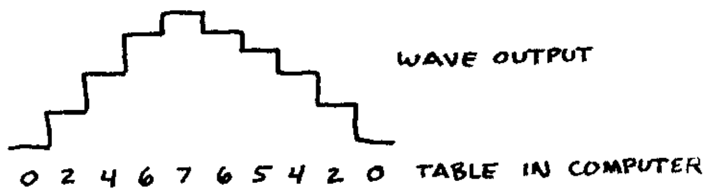




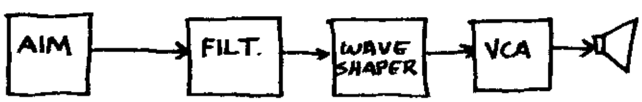
A D/A that can convert numbers from 0 to 255 into a voltage is fairly inexpensive. Such a D-to-A is perfect for creating presets and doing some kinds of voltage control. A D/A that can accurately control the frequency of an oscillator generally requires greater precision and is somewhat more costly. In "computer time" most of the control voltages that a synthesizer sequencer or envelope generator might produce are very slow, thus a computer, with the appropriate hardware, can handle many channels of control voltages.

AUDIO VOLTAGES:

It is possible for a computer, using a D/A converter to output actual waveforms. Usually such a waveform is stored in a table and "read" out to the D/A. The computer can manipulate these waveforms in certain interesting ways, but in general, the programmer/synthesist will be up against the speed of the computer. With 8-bit computers it is still fairly hard to duplicate the timbral possiblities of the filters, waveshapers, and modulators of the synthesizer, especially if these are to change in fast, musically useful ways.



It is fairly easy to produce square waves using a one-bit output of the computer and to control its frequency accurately. This pulse wave can then be "timbre" modified by the synthesizer. The Serge system makes a perfect companion for such digital waveform generation. These waveforms almost invariably need filtering and can be further modulated using Wave-Multipliers and VCAs. These constructed waves can be mixed with analog waves.



PULSE CONTROL OF COMPUTER:

In much the same way that the computer sends a pulse to the synthesizer, the synthesizer can send a pulse to the computer. This pulse can tell the computer that some effect has commenced or that one is over. Knowing this, the computer can then respond in some appropriate way. The computer can "see" the pulse in one of two ways: The pulse can interrupt the computer, sending it to a new program; or, the computer can continually "poll" the bit , acting only when it finds a pulse there.

ANALOG TO DIGITAL CONVERSION:

The functional opposite of a Digital to Analog converter is an Analog to Digital converter which inputs an analog voltage and outputs a series of numbers that corresponds to the input voltage.

Using an A/D converter any control voltage can be converted into a series of numbers and stored in the computer. This table then can be altered and read back out to the Serge. If the A/D converter is fast enough it is possible to "read" audio waveforms which can then be manipulated and sent back to the Serge using a D/A converter.