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.

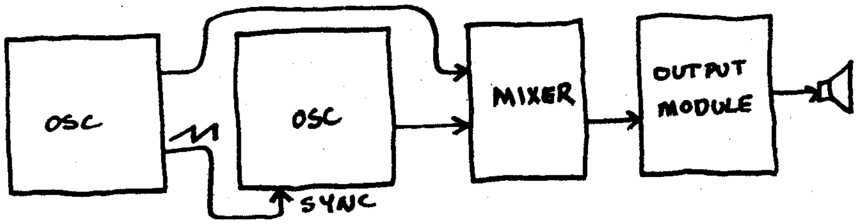
A fifth type of module that has not been dealt with yet is an Audio-to-Control-Voltage converter. One example might be a module that inputs an audio signal and outputs a control voltage representing the envelope of that sound. Such a module is called an envelope follower.

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.

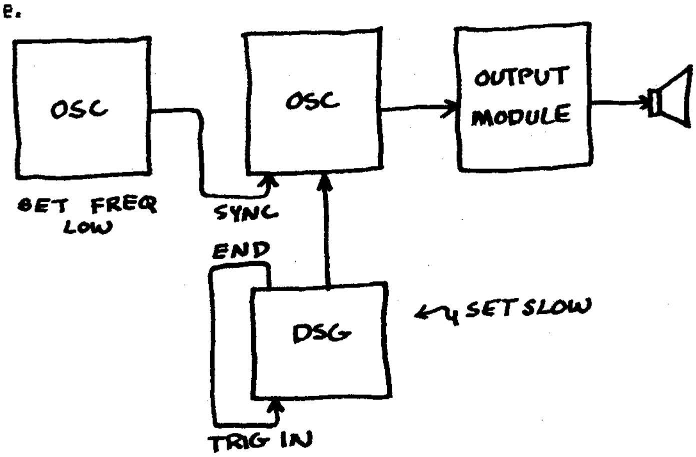
Each of the next five chapters will cover one of these five types of modules, presenting modules and functions not yet covered. It is suggested that each module be explored as it is presented by setting up patches using it with modules already understood.

SIGNAL SOURCES

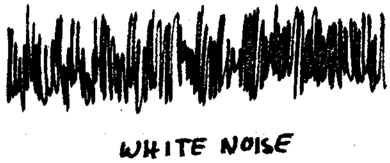
OSCILLATORS. Both the NTO (New Timbral Oscillator) and the PCO (Precision Controlled Oscillator) have been treated earlier in this manual as to their ability to generate audio frequencies. Both the NTO and the PCO have an input labelled "SYNC". This input allows two OSCs to be locked together so that 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.

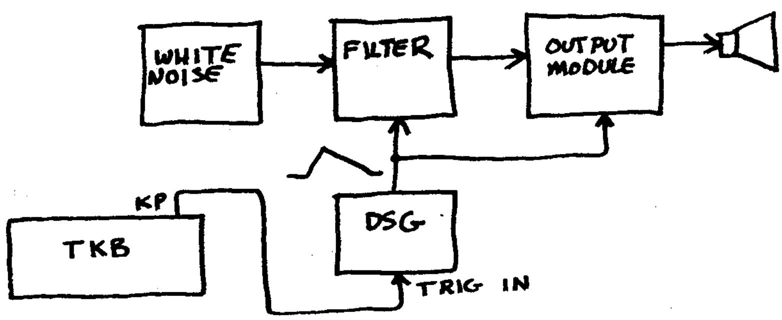


In the above patch OSC #2 is locked to OSC #1. It cannot drift. An interesting phenomenon occurs if OSC #1 is set very low and its SAW wave is used to SYNC OSC #2. If OSC #2 is now swept upwards over its range, either using its pot or a control voltage, you will hear it locking onto one overtone after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



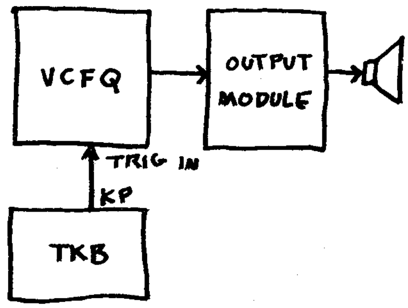
WHITE and PINK NOISE. White noise is a complex wave in which ALL frequencies appear mixed together. The sound is a sort of hissing sound. Since it contains all frequencies it can be filtered in various fashions to produce bands of sound in many different frequencies. This is the ultimate material for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

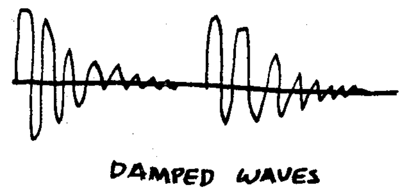




Pink noise is like White noise except that it sounds lower, more like a waterfall. The low frequencies have more amplitude.

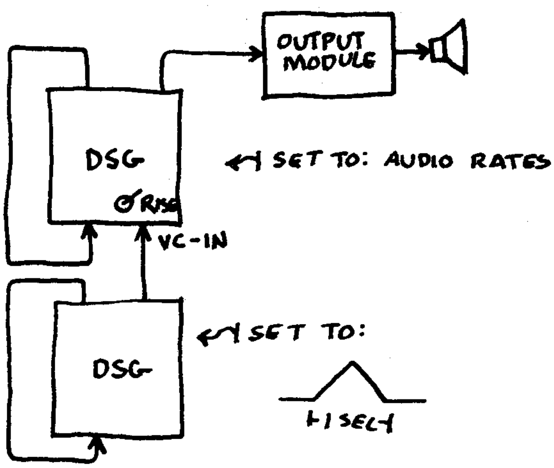
RINGING FILTER. The VCFQ can be "rung" much like a gong, with a trigger pulse to the TRIG-IN. By adjusting the FREQ, different pitches can be achieved. Adjusting the Q will alter the sound from percussive clicks to bell-like sounds. The output is a damped SINE wave.



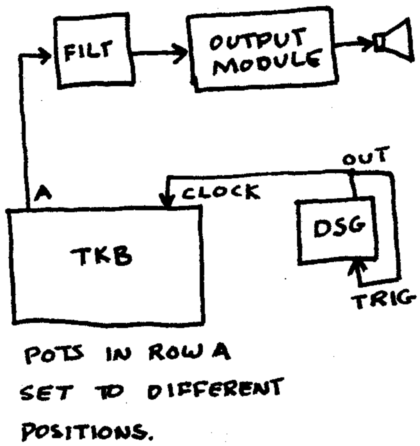


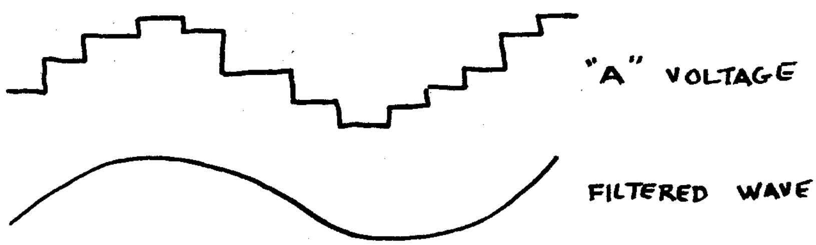
This technique can be used when other signals are applied to the INPUT of the filter to produce a wide range of interesting sounds.

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.

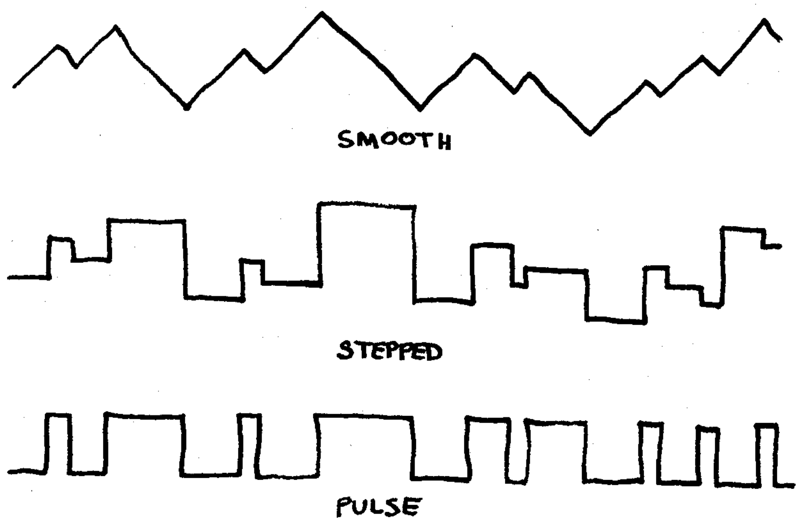




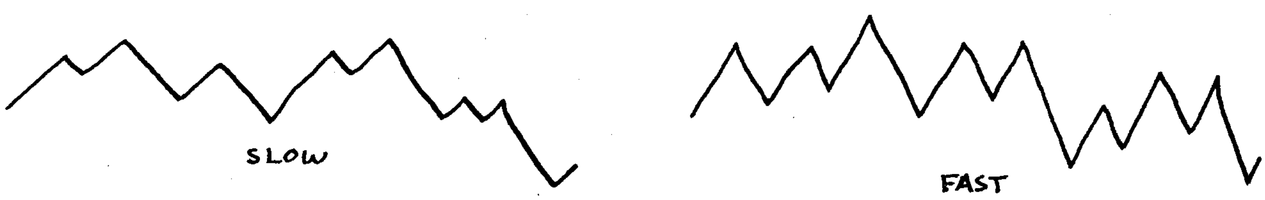
CONTROL VOLTAGE SOURCES

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.

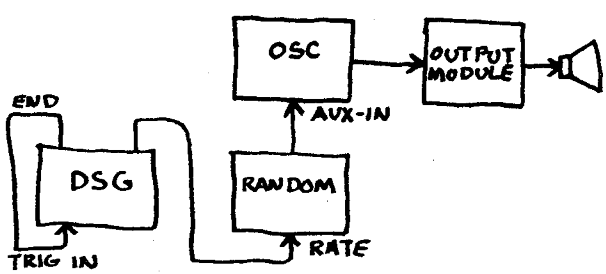
The Serge provides three kinds of random control voltage: stepped, smooth, and pulse. These are diagrammed below:



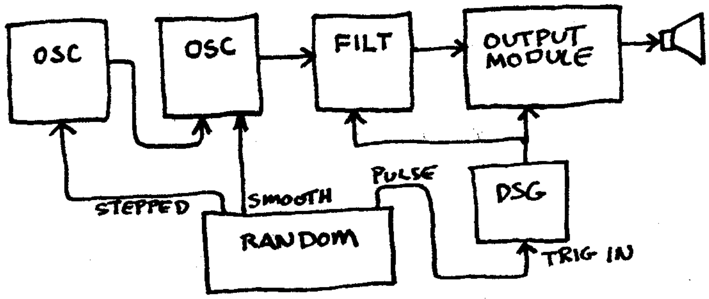
The RATE pot at the bottom of the module determines the overall rate of change, a function which can be voltage controlled.



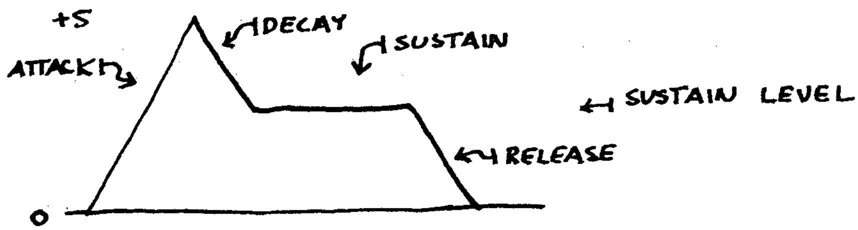
The rate of randomness can be controlled on the random module, and a Processor or a processing Input can scale the random output to any desired level. The following patch is useful for exploring the possibilities of the smooth and the stepped random output:



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:

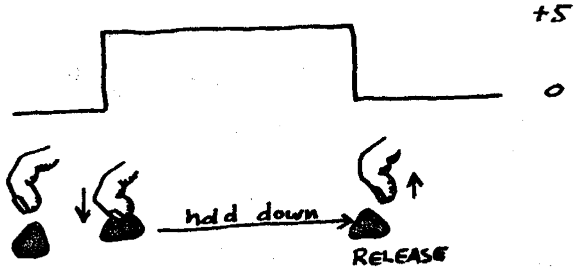


ADSR. The Extended ADSR is an envelope generator that can produce multi-segmented envelopes of a more complex variety than a single DSG is able to provide. In certain kinds of synthesis this is necessary, since few natural envelopes are a simple rise and fall. The ADSR is able to provide a four-part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



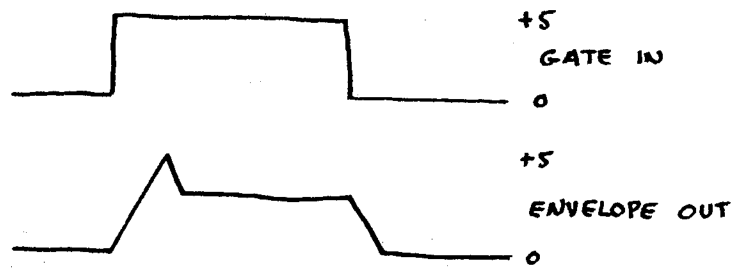
This envelope, in a general sort of way, represents the envelope of trumpet or any instrument which can sustain a note at a steady level. The sustain section is settable to different Sustain Levels by means of a pot and voltage control. In addition to these functions the ADSR also provides a delay that 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.

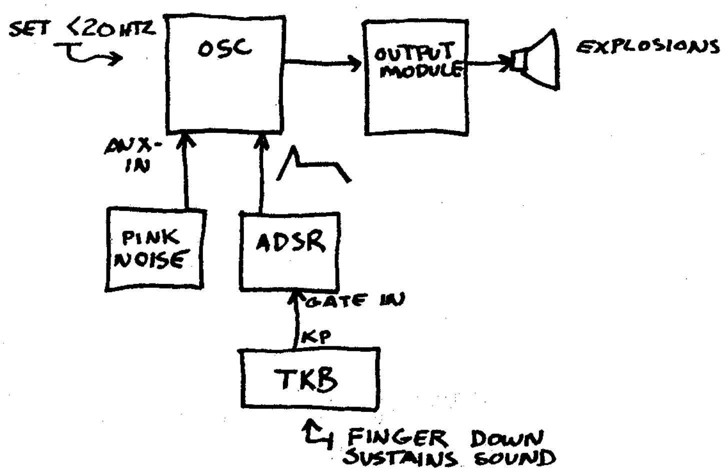
The module is triggered by a pulse to its GATE input. Usually this trigger comes from a keyboard device such as the TKB's keyboard output (KP) which is a trigger that stays at a +5 volt level as long as a finger remains on the key:



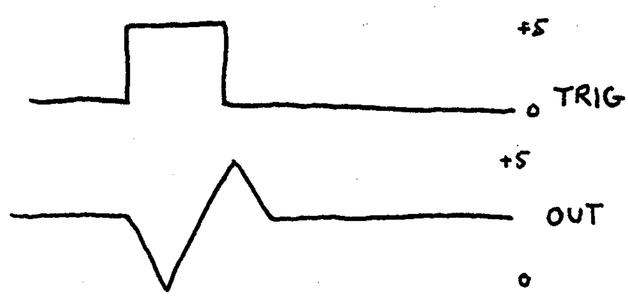
This sustained high level will be used in the timing of the ADSR's output.

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.



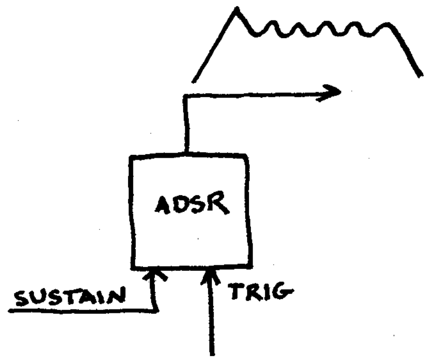


The ADSR functions in a slightly different fashion if the trigger pulse is applied to the TRIG input and not the GATE input. When there is no input to the GATE, the output of the ADSR remains at the voltage level set by the Sustain pot (pot #4). When the ADSR receives a trigger pulse the voltage drops 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.



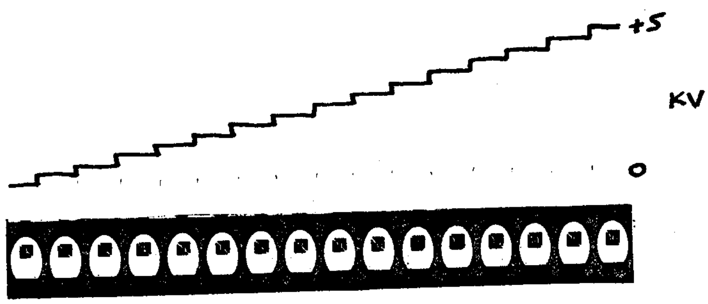
Using both inputs, more complex envelopes are possible. For example, during the sustain time set up by a GATE pulse, a trigger received at the TRIG will cause a new attack to start.

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.

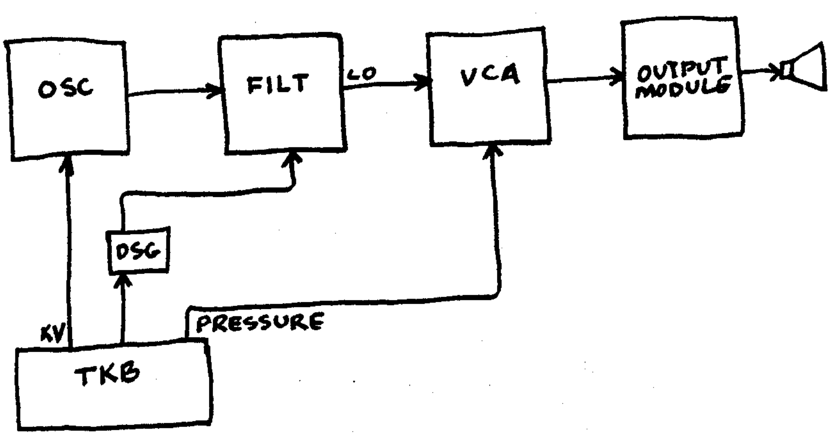


TOUCH PAD KEYBOARD on the TKB. The TKB can be thought of as two separate but interconnected modules: the Sequencer and the Keyboard. The two modules can be partially separated from each other by switching the KEYS switch to OFF.

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.

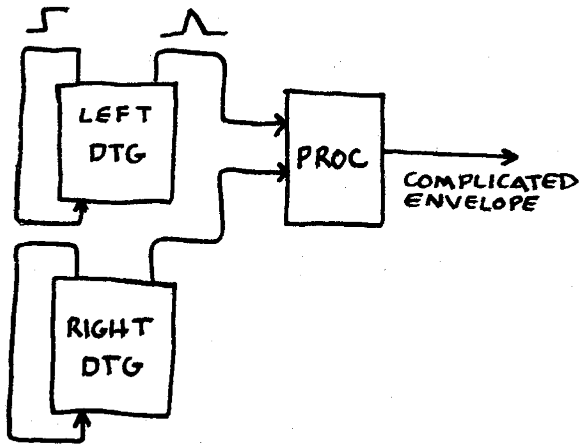


The Pressure output is a voltage proportional to the amount of pressure applied to the keyboard with the finger. When used to control a VCA, it can act much like an expressive envelope generator simulating the "piano-forte" (soft-loud).

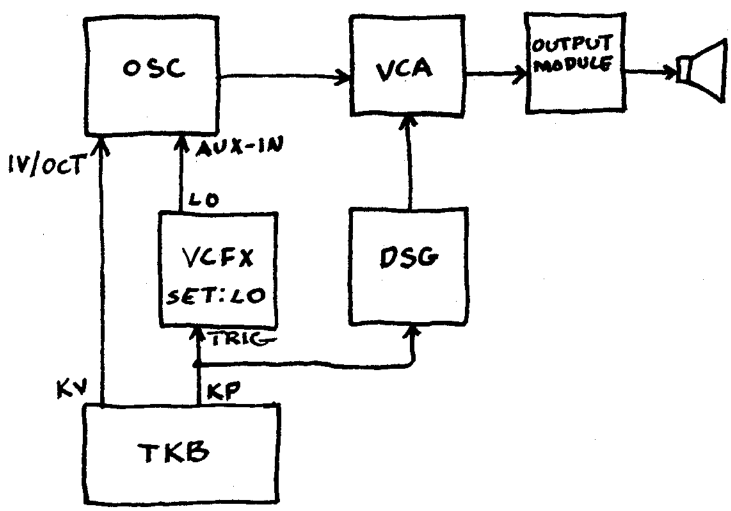


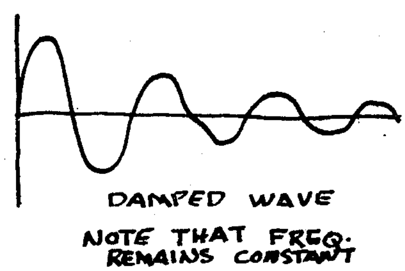
The pressure output works by using a capacitance detector. You will find that the position of your other hand in relation to the faceplate of the synthesizer affects its output.

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.

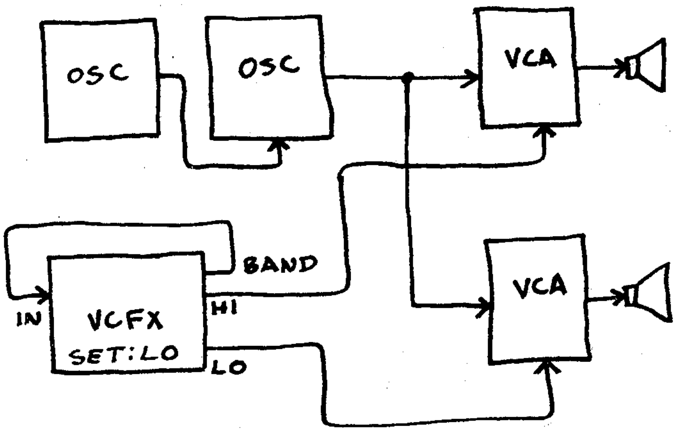


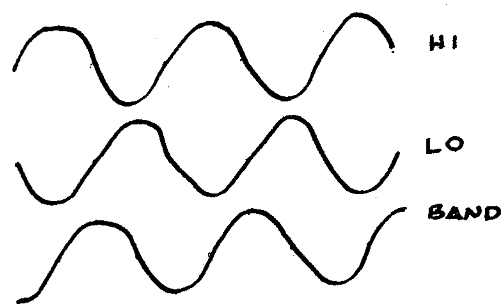
EXTENDED RANGE VCF. A surprising source of control voltages is the Extended Range VCF (VCFX) which, like the Variable Q filter, can be made to "ring" -- in this case, to ring at sub-audio or control voltage frequencies. The VCFX should be set to its LO setting. A trigger applied to its TRIG in, when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



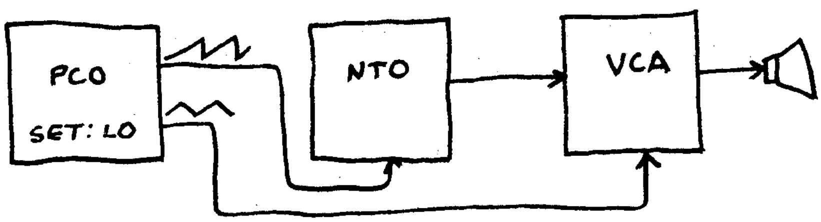


When the BAND output of the VCF is patched back into its own input to produce a feedback loop, the VCFX will oscillate. Sine waves are then available at its HI, BAND, and LO outputs. these sine waves will be 90 degrees out of phase with each other and are useful for creating stereo and quad panning patterns.





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.

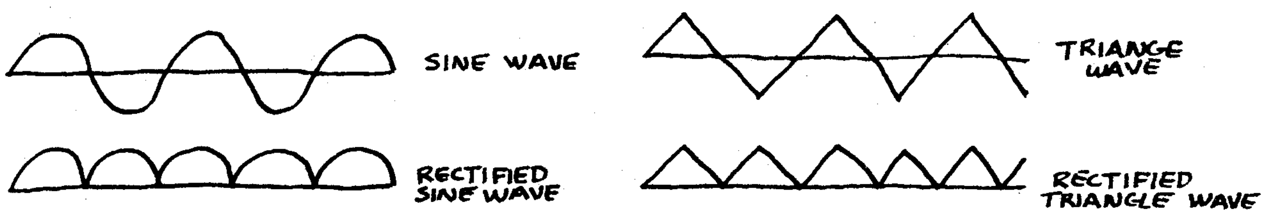
There are four major paths to analog electronic music synthesis.

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.

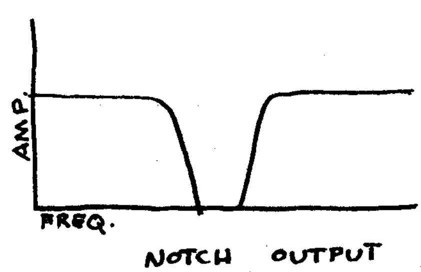
WAVESHAPING: Waveshaping is a technique where a given wave is input into a device and a related but different wave is output. For instance, a simple waveshaper is a "Rectifier" which outputs the absolute value of its input wave.



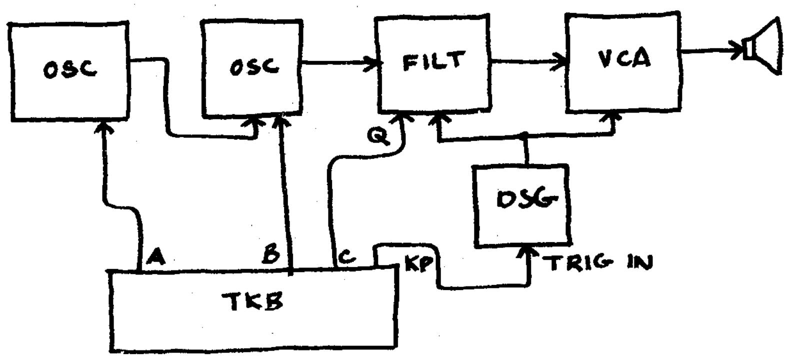
Modules that waveshape signals, add signals together, Subtract parts of signals, or that Modulate signals are called Signal Processors. The Serge system is a Signal Processor-rich synthesizer and includes many processors that are not found on any other synthesizer. So far in this manual the processors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



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.

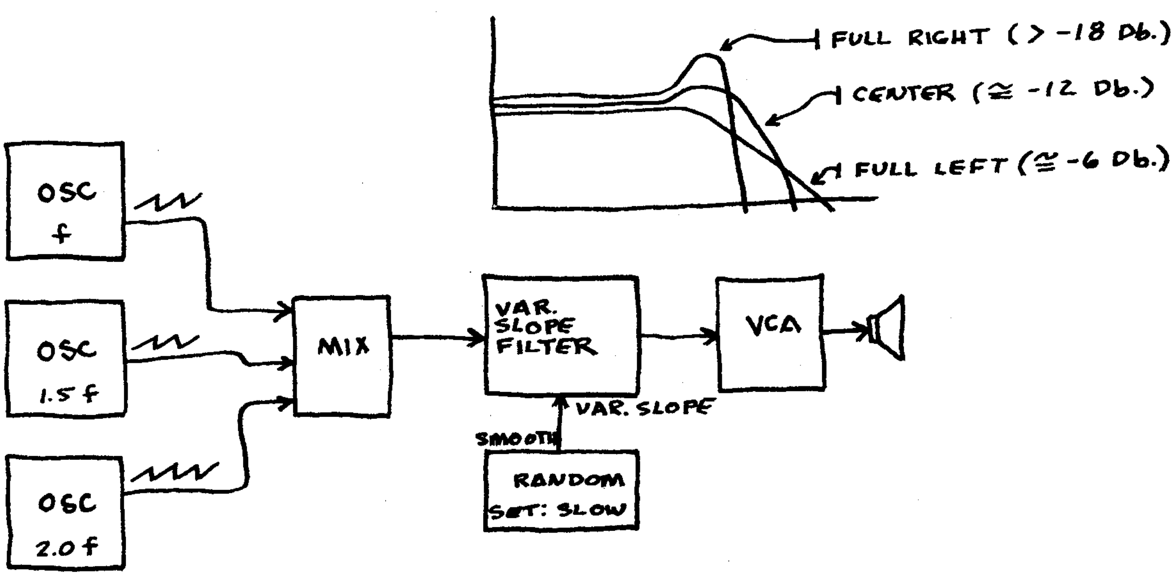


AUTOMATIC GAIN CONTROL. One of the problems with a Hi-Gain Q filter is that if the cut-off frequency hits an overtone that is higher in amplitude than expected, it can overload the filter and/or the speaker system and cause distortion. The AGC keeps the output gain constant at the cut-off frequency and 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.

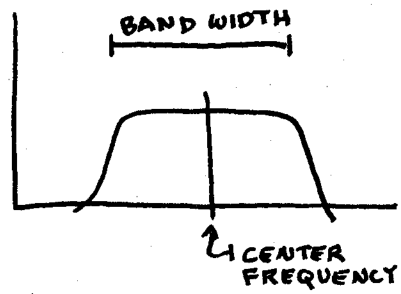
VARIABLE SLOPE VOLTAGE VCF (VCFS): Other than having the regular features of a voltage controlled filter, the VCFS has the following features:

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.

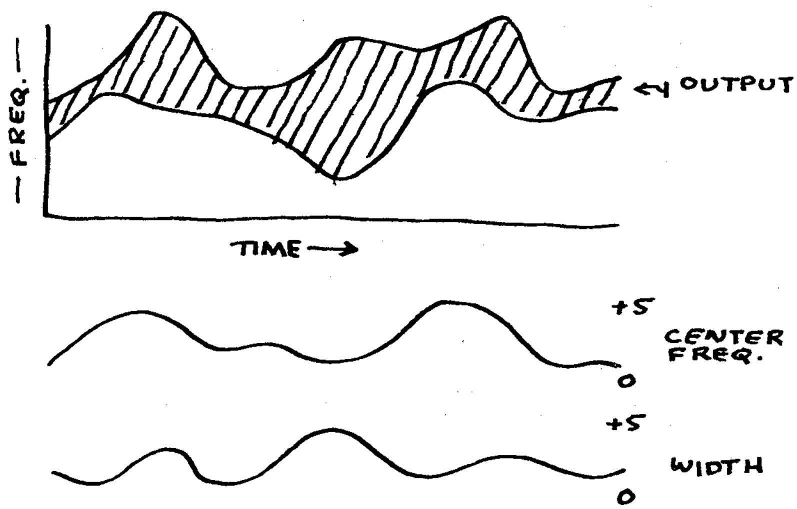
VOLTAGE CONTROLLABLE SLOPE. The Slope of a filter is the amount of attenuation per octave the input signal is attenuated beyond the cut-off frequency. Though an "ideal" filter has a 60 db/oct cut-off (or even greater) slope, this is not a particularly useful slope musically. In fact, different slopes 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.



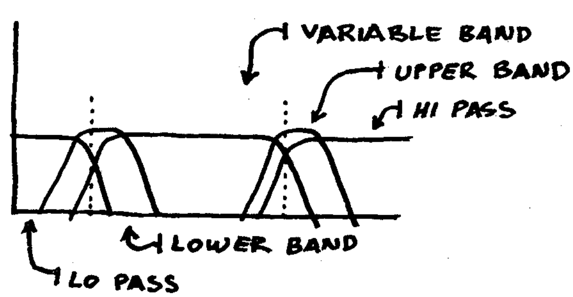
The VCF2 also has four other outputs:

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.

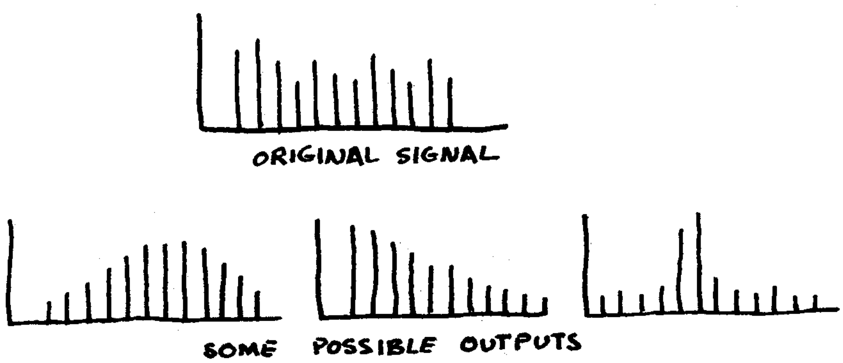


The VCF2's output on all five bands is flat, that is there is no resonance or Q, and therefore is ideal for processing concrete sounds, or for studio use since the filter does not change the quality of the sound that it is passing.

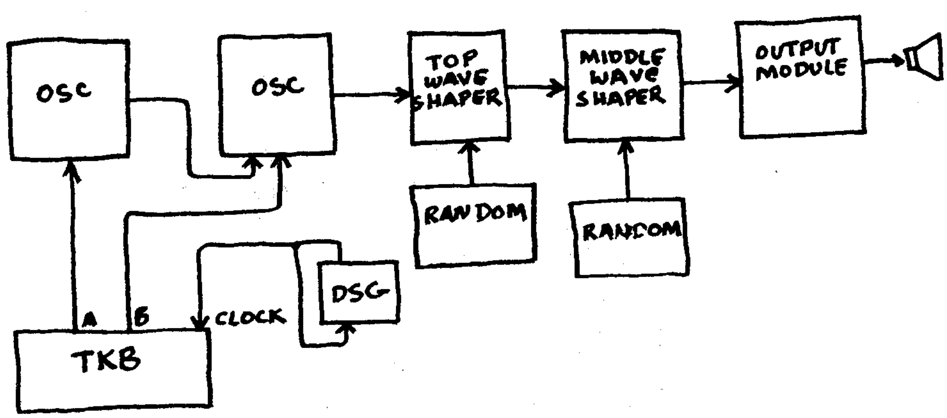
RESONANT EQUALIZER (EQ). An Equalizer or Comb filter is a bank of band pass filters that covers the entire spectrum and whose outputs are mixed together such that the amplitudes of each filter can be controlled. With this device the sound as a whole can be adjusted and balanced to suit. The Resonant 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).

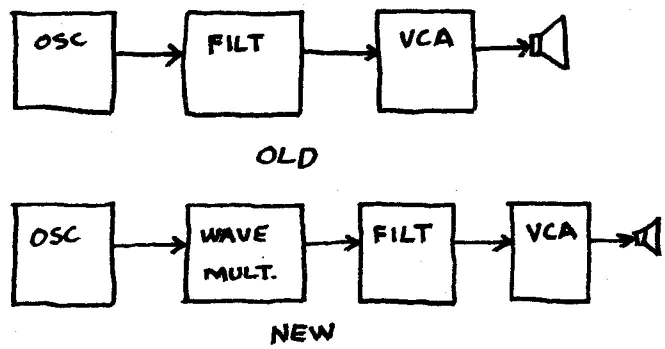
The bands are arranged in sevenths so that a false tonic does not develop. A Level pot at the bottom adjusts the over-all gain of the output and prevents overload when the resonance is set high. These fixed resonant bands are common in almost all timbres produced by musical instruments, and it is the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



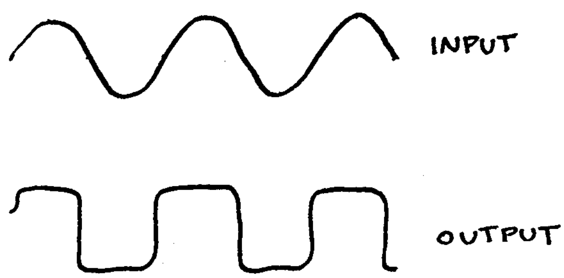
TRIPLE WAVESHAPER. (TWS). The Triple Waveshaper module contains three identical devices which can be used to convert sawtooth waves into sine waves and can provide a wide range of other forms of sound and timbre modification. The timbre can be affected by a manual pot and two different VC inputs which 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.



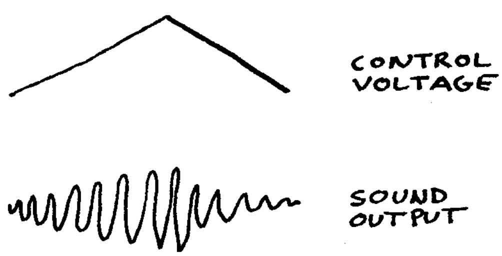
WAVE MULTIPLIERS (VCM). The Wave Multiplier Module is a triple module that, unlike most other multi-modules on the Serge, contains three DIFFERENT modules. Each of the three modules operates on its input in a unique fashion, transforming simple sounds into musically complex and interesting ones. They 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.

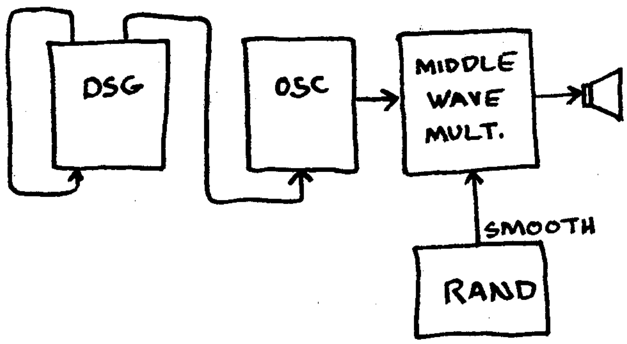


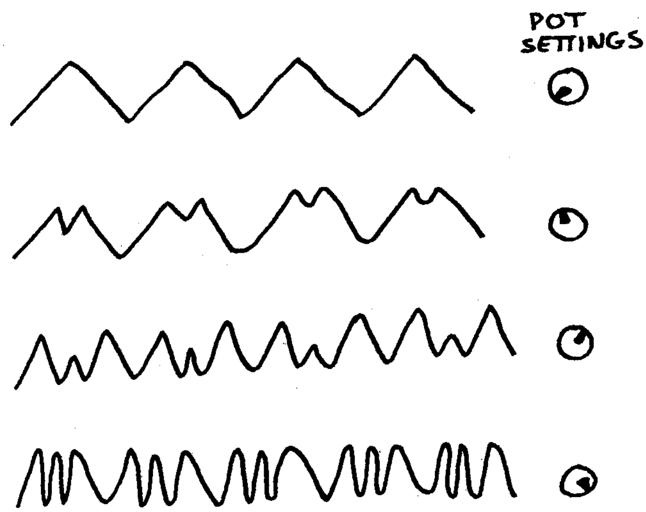
In the LO setting it acts like a linear VCA, a device useful for producing different types of AM sounds.



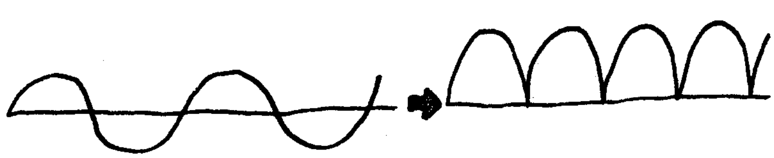
In both settings the module can be controlled either manually or with a VC (control voltage.)

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.

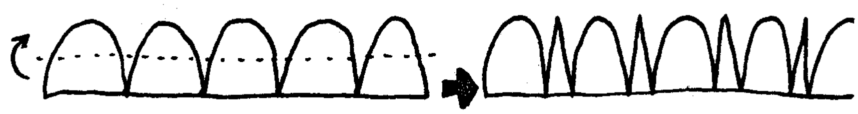




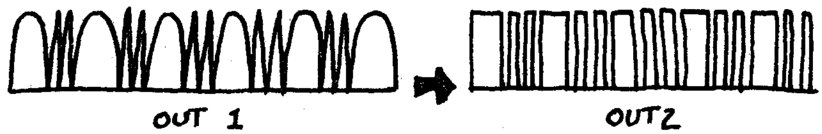
BOTTOM WAVE MULTIPLIER: Like the Middle Multiplier, the Bottom one also has two independent (but identical) inputs. Both inputs are AC coupled. The general effect of the module is that of a full-wave rectifier for audio signals, which means that negative voltages are "flipped" up into the positive.



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.

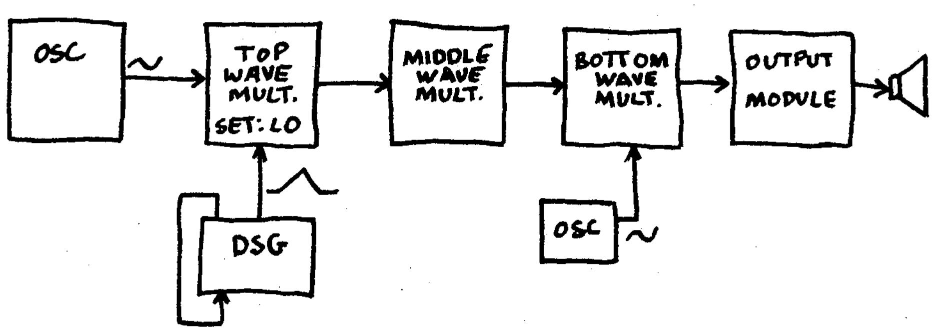


Like the upper two wave multipliers the module provides both manual and voltage control over the output. Unlike its companion modules, however, there are two distinct outputs, OUT 1 and OUT 2. OUT 2 provides a "squared up" version of OUT 1.

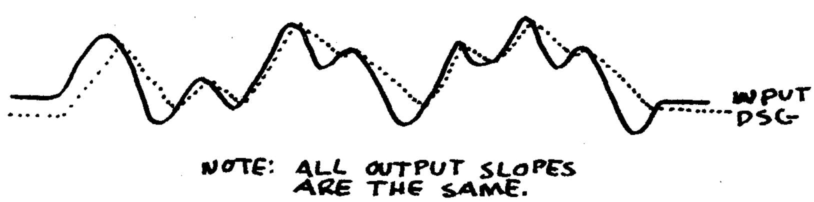


One important feature of this lower module is that, unlike simple rectifiers, the amplitude of the output does not decrease through succesive rectifications.

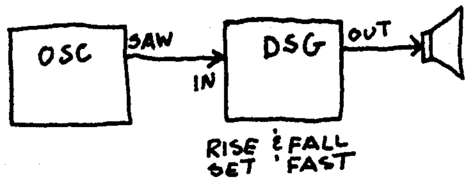
Overall, these three Wave Multipliers provide a method of producing timbres as rich and as varied as acoustic sounds and yet having the precision and repeatablity of analog synthesis.



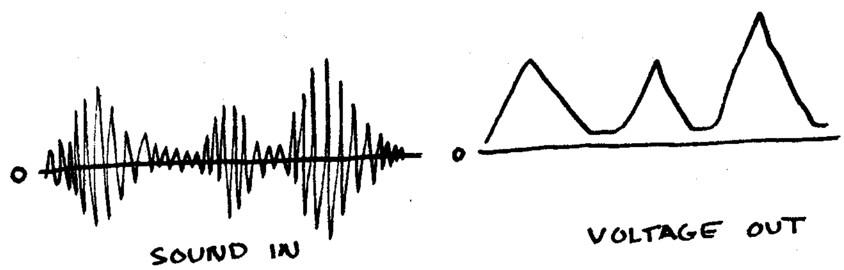
DUAL UNIVERSAL SLOPE GENERATOR (DSG). The DSG can function as a non-linear lo-pass filter essentially by softening the slopes of the IN signal. Generally speaking, the less steep the slope of a waveform the fewer high frequencies it contains. To accomplish this the RISE and FALL times must be set quite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



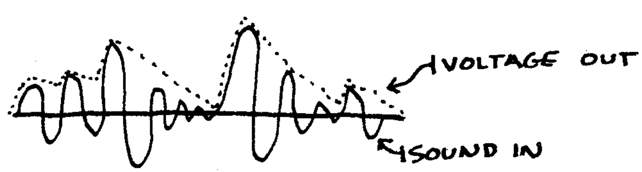
Because the RISE and FALL time on the DSG is voltage controllable, when used in this fashion, the module becomes a voltage controlled filter.



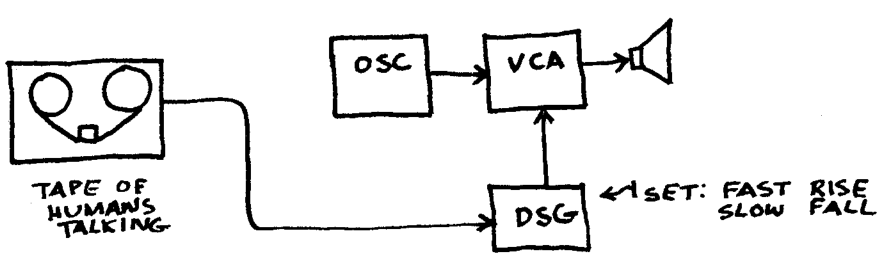
Closely related to this patch is the use of the DSG as an ENVELOPE FOLLOWER. An envelope follower is a device or module that inputs a complex sound and outputs a control voltage proportional to the envelope of the input.



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.

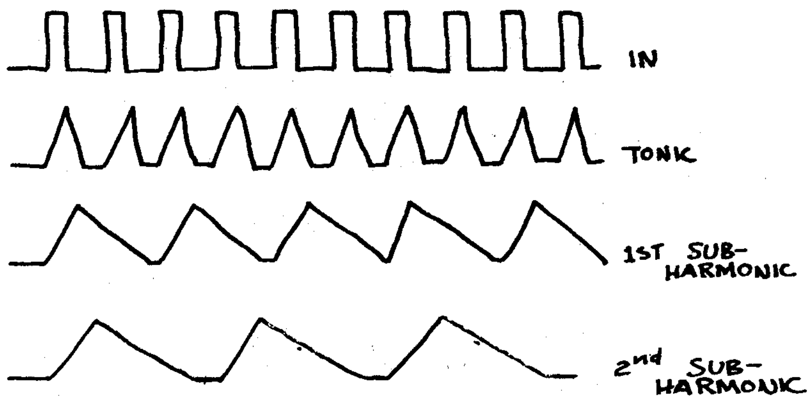


On the DSG this is easily accomplished by using a very fast RISE time and a slow FALL time.



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.

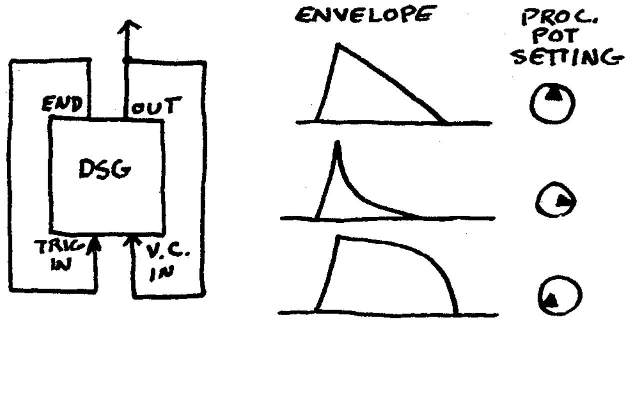
Another use of the DSG is that of an SUB-HARMONIC or UNDERTONE GENERATOR. This is accomplished by applying a very fast pulse train to the TRIG IN and by having the DSG set to audio frequencies. The DSG will not respond to a second trigger until its envelope is complete. If the duration of the envelope 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.



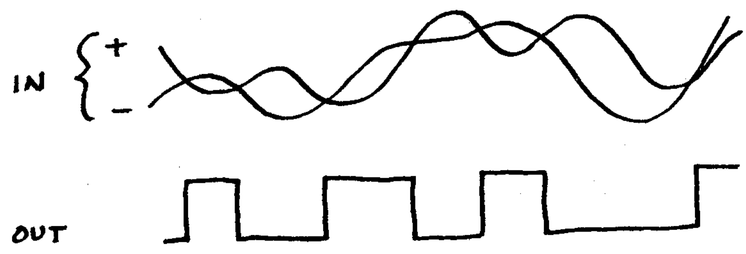
It is sometimes desirable to shape the envelope so that it has a NON-LINEAR SLOPE. this is often the case when producing long sustained sounds or sounds that gradually change in loudness over a long duration. The DSG, when patched to the VCA, will seem to have little effect on the loudness of sound at 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.

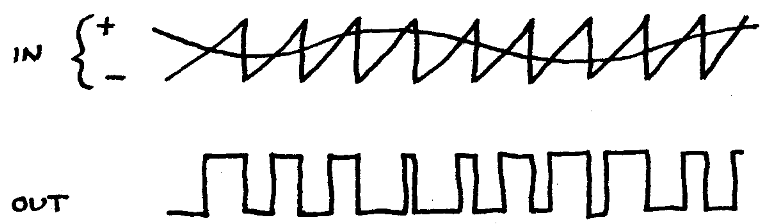
If the processing pot of the DSG is turned to the right, the feedback has the opposite effect: the envelope becomes a sharper and sharper spike, useful for creating short percussive envelopes.



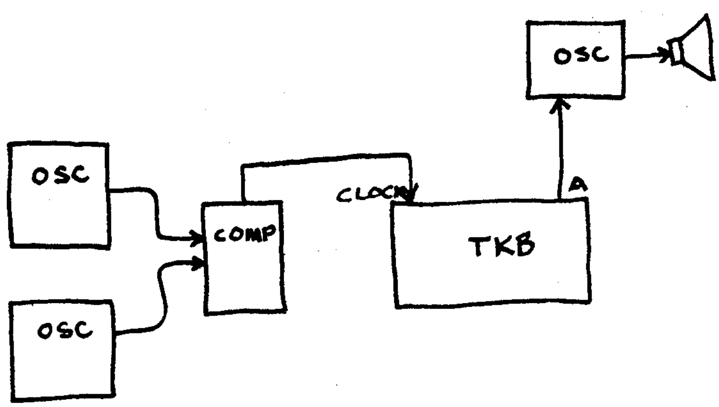
COMPARATOR (COM): The action of this module is this: When the voltage of the "-" input is greater than the "+" input, the OUT goes high to +5 volts. Otherwise the OUT is 0 volts. The pot sets a threshold voltage which is added to the "+" input. If there is no "+" input, this pot alone sets the voltage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



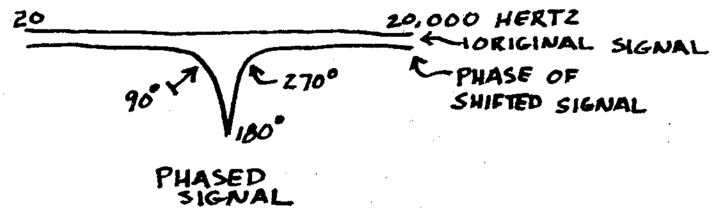
This device can be used to create rectangular waves of various duty cycles. By inputting a sawtooth wave into the "-" input (which is labelled with a "sawtooth" wave to indicate this) and inputting a control voltage into the "+" input, a voltage controlled pulse width generator can be created. These 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.

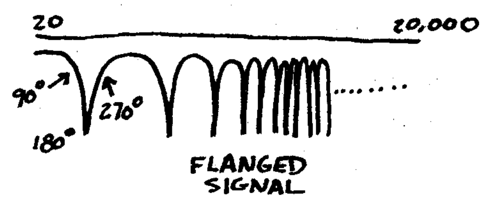


The pulse outputs of the COM can be used to trigger any device on the Serge that requires a trigger pulse.

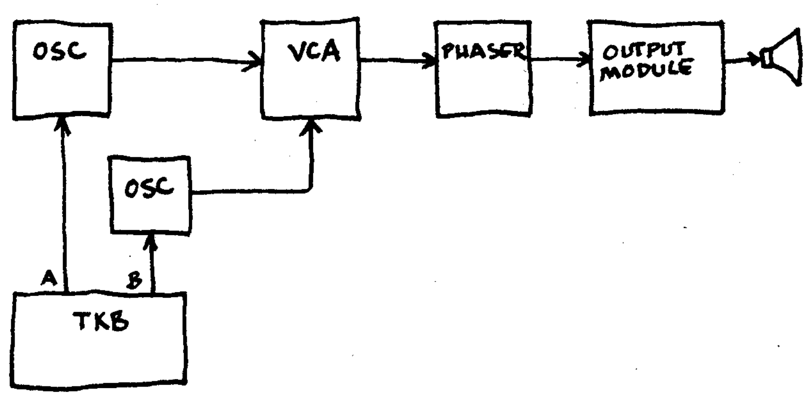


PHASER (PHA). When two identical signals of different phase are mixed together, various frequencies of the signal are cancelled out or attentuated. Which frequencies are cancelled is dependent on the phase relationship between the two signals. "Phasing" is often created when a single sound arrives at 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.

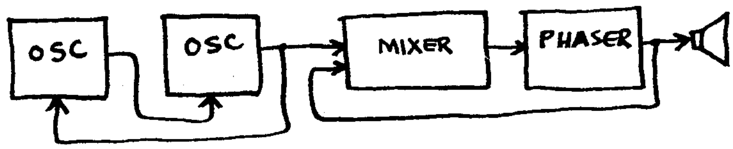




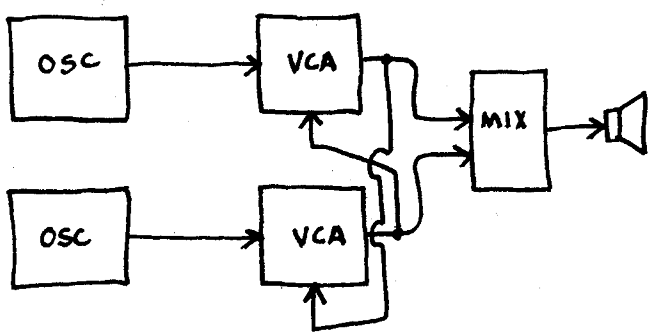
The Serge Phaser is a three-stage phaser providing outputs at each stage (360, 720, and 1080 degrees), however, the characteristic sound of phasing only occurs when these phase-shifted signals are mixed back into the original signal. (The shifted signals by themselves are useful for certain spatial effects, 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.



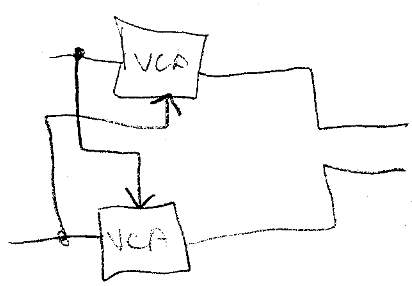
This module also comes in a DUAL PHASER Module (2PHA). The upper phaser has 720 degree phased output as well as the mixed out, while the lower phaser has a 360 degree out and the mixed out. If desired, these two modules can be strung together to produce a single, deeper phase shift. It has been found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

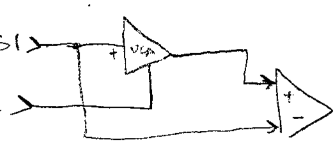


QUAD VCA. (QCA). The Quad VCA contains four independent VCAs each with an audio input, a voltage control input, an output and a GAIN pot. These are logarithmic VCAs and are useful for putting envelopes on sounds, and for Amplitude Modulation.



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.

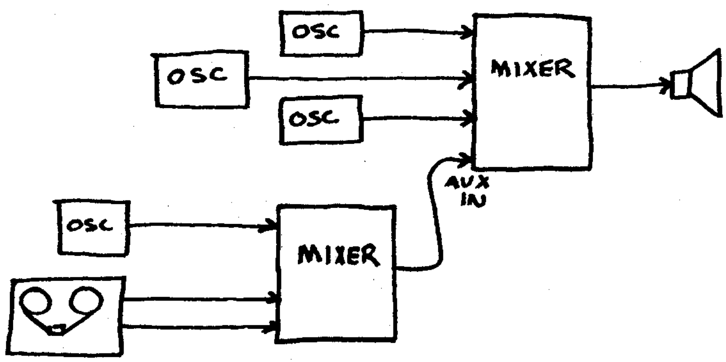
A MIXER is a module that adds together a number of different sounds and sends the mixture to one or more outputs. A MIXER can be categorized by the number of inputs and outputs it has.

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.

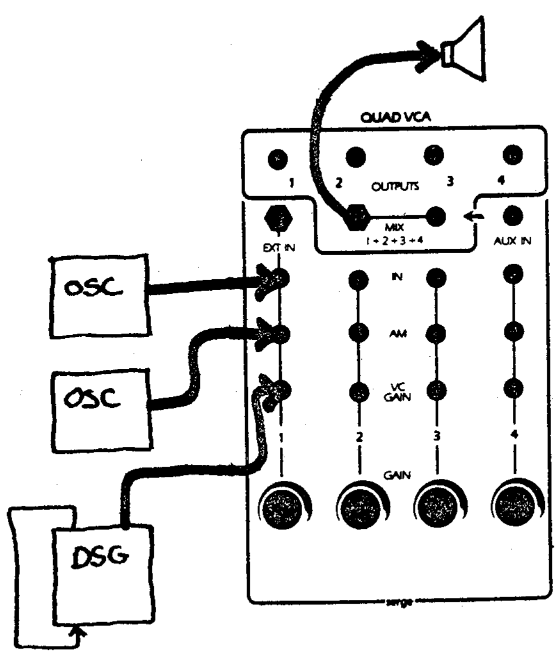
The Serge system offers a wide variety of voltage-controlled Output Mixers, many of which also contain Panners, Crossfaders and voltage-controlled Amplifiers. These mixers also have at least one grounded output jack that can be directly patched to an amplifier. Typically this jack is a Mini-phone, but 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.

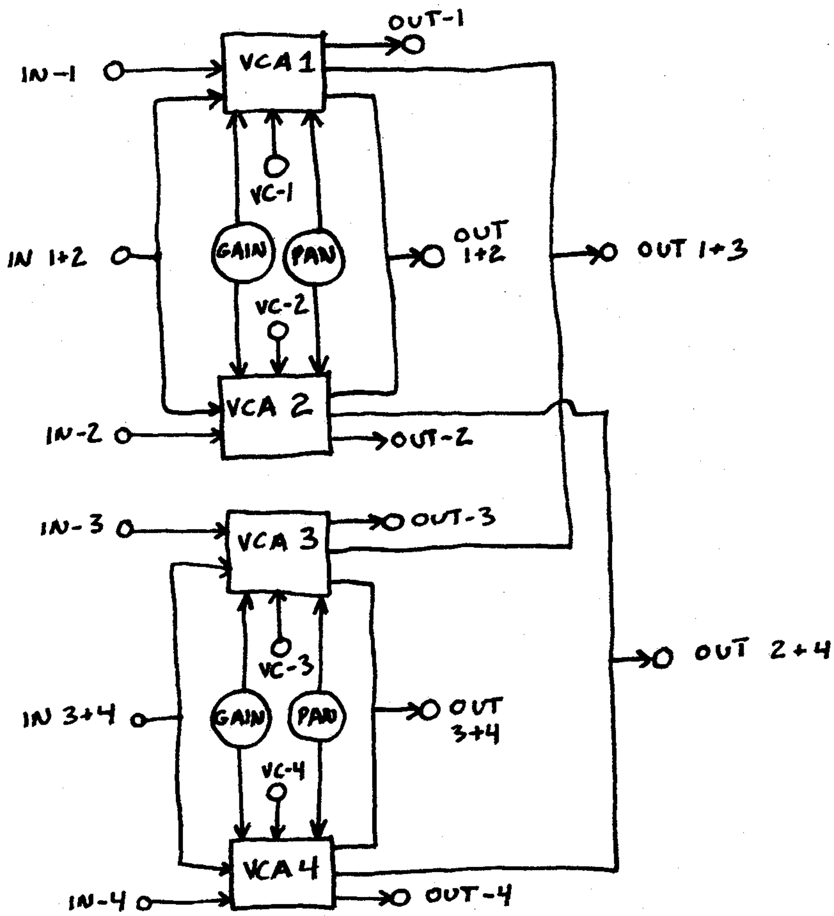


QUAD VCA (QCA). The Quad VCA contains four separate, independent VCAs and provides an additional output Mix of all four VCA outputs. These VCAs are arranged as parallel columns with the outputs at the top of the module. The gain of each VCA can be controlled by a GAIN pot at the bottom of the module 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.

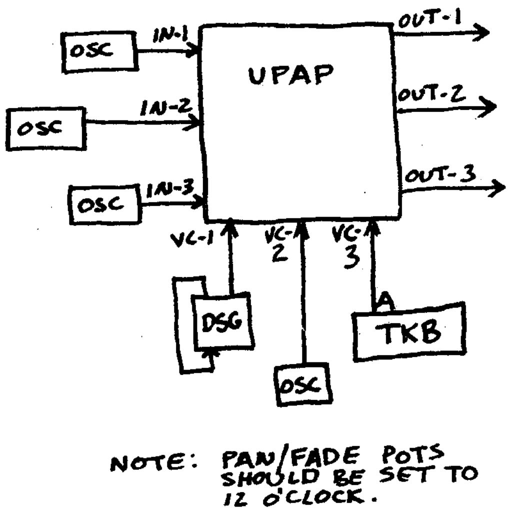
The output of each VCA appears individually at outputs 1,2,3 and 4 respectively. There are four mixed outputs: 1+2, 3+4, 1+3 and 2+4. Outputs 1+3 and 2+4 have, in addition to their banana jacks, mini-phone jacks to be used to send the output to external equipment. These are the usual left and right stereo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

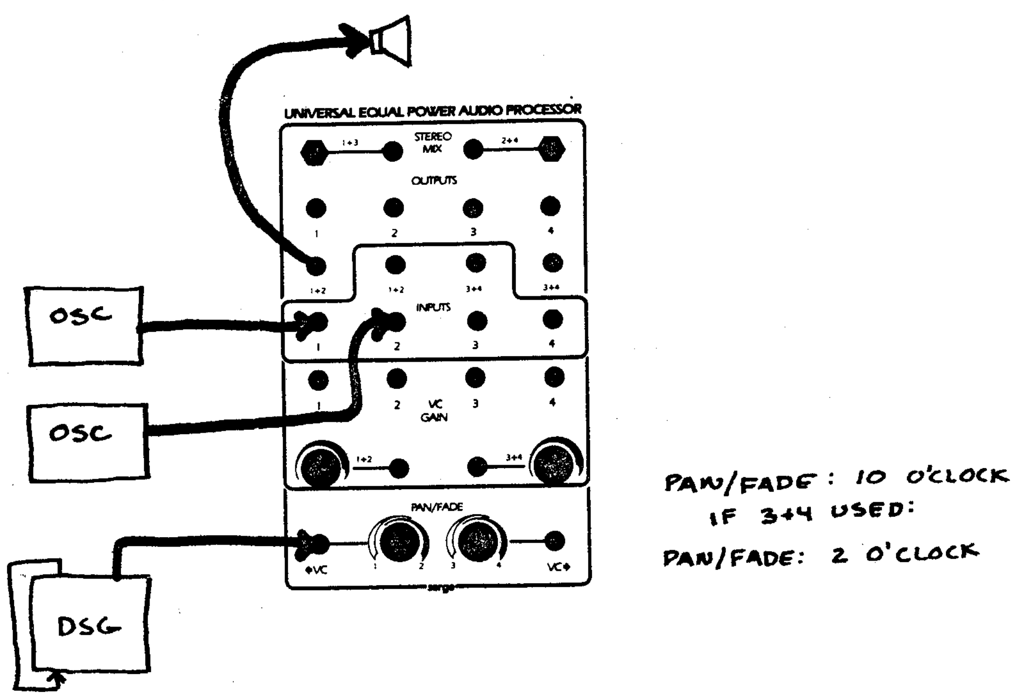
Lastly, the gain of the two pairs of VCAs (1 & 2 and 3 & 4) can be controlled by the Pan/Fade pots and their associated VC inputs such that, if set properly, the combined gain of the two remains the same during a cross-fade between them.

PATCHES:

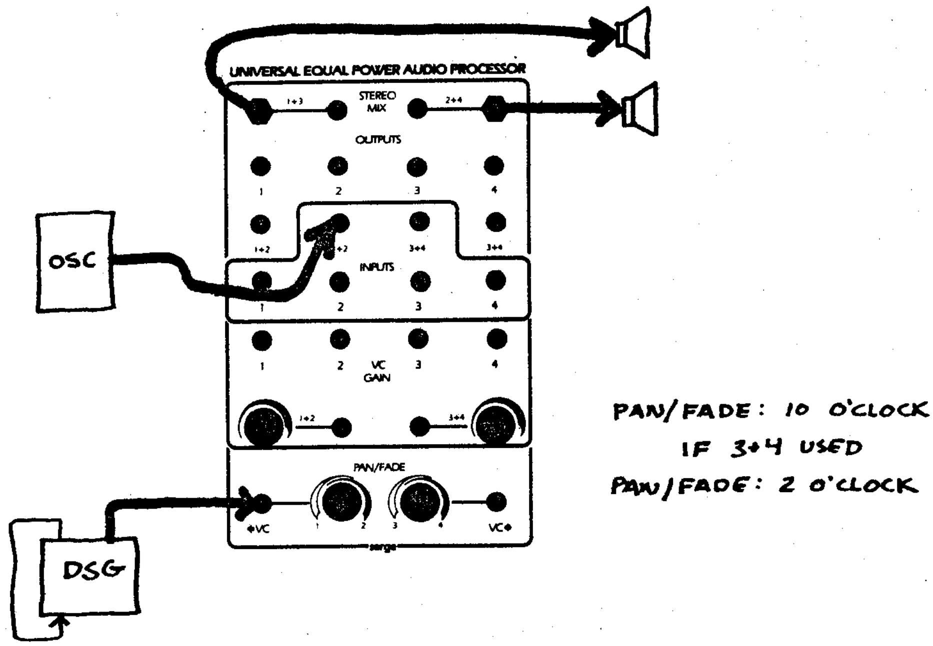
1. Four VCAs:



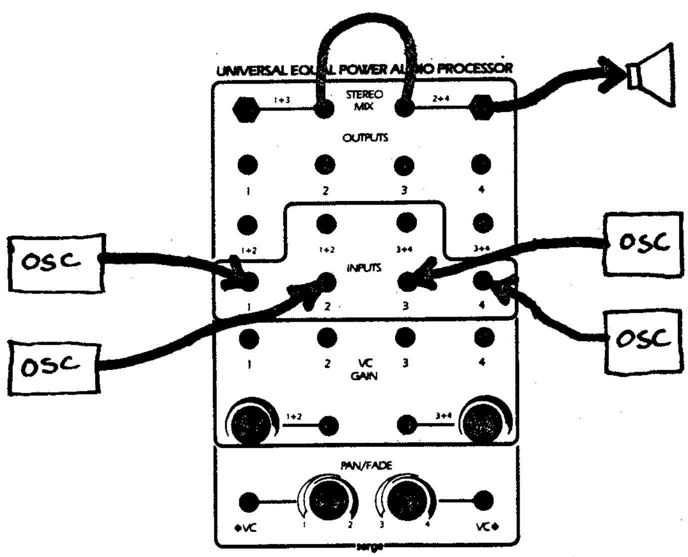
2. Two Crossfaders:



3. Two Stereo Panners:

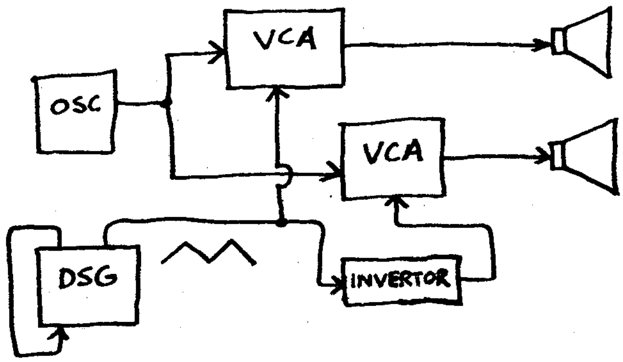


4. Four out mixer:

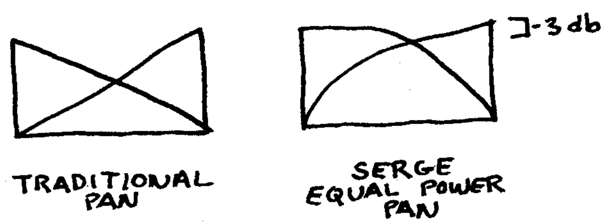


A Note on the PAN/FADE:

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:



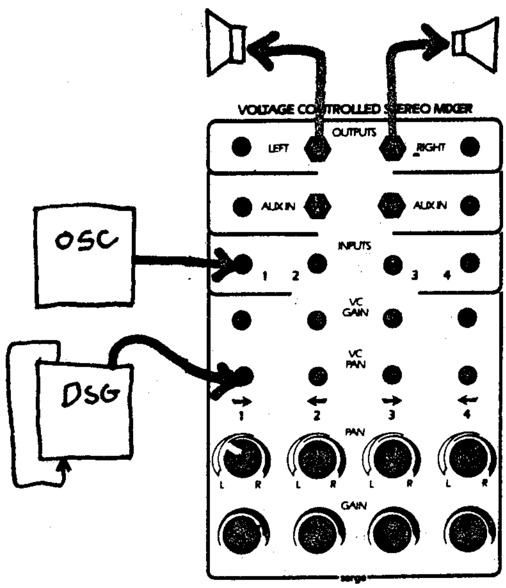
This technique caused a severe dip in overall loudness of the sound at the half-way point. The PAN/FADE control on the Serge corrects for this problem by keeping the total power (and perceived loudness) equal as the sound moves from speaker to speaker. This technique uses sophisticated circuits to compute 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".

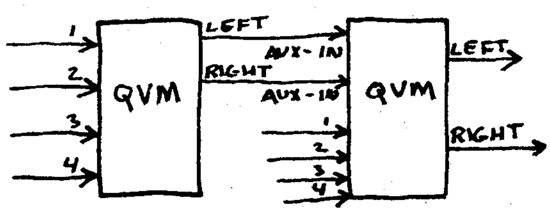
If the pan/fade pot is set beyond these cut-off positions, it will increasingly override the control voltage action. For example, if pan/fade pot 1-2 is turned full left, nothing will be heard from output #2 regardless of other VC inputs.

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.

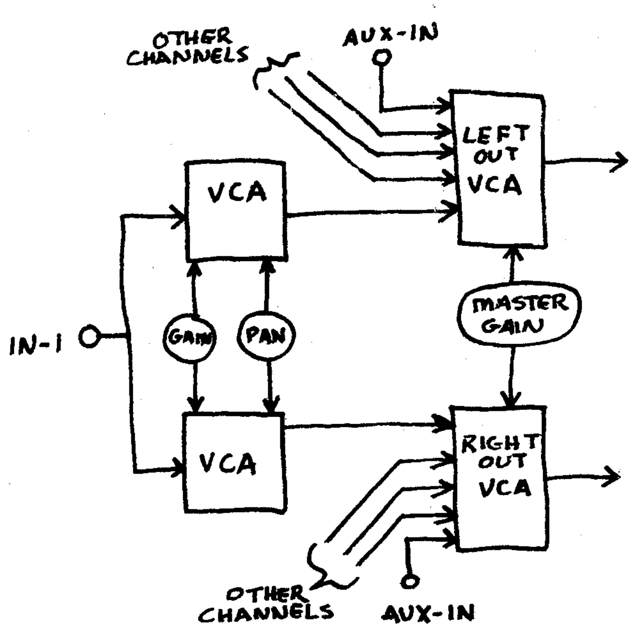
The QVM has auxiliary unity gain inputs for each of its two output channels. These can be used to create an 8-in 2-out mixer by patching the outputs of one QVM into the AUX-INs of another QVM. The AUX-INs also have a mini-jack input which can be used to bring external signals into the Serge.



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.

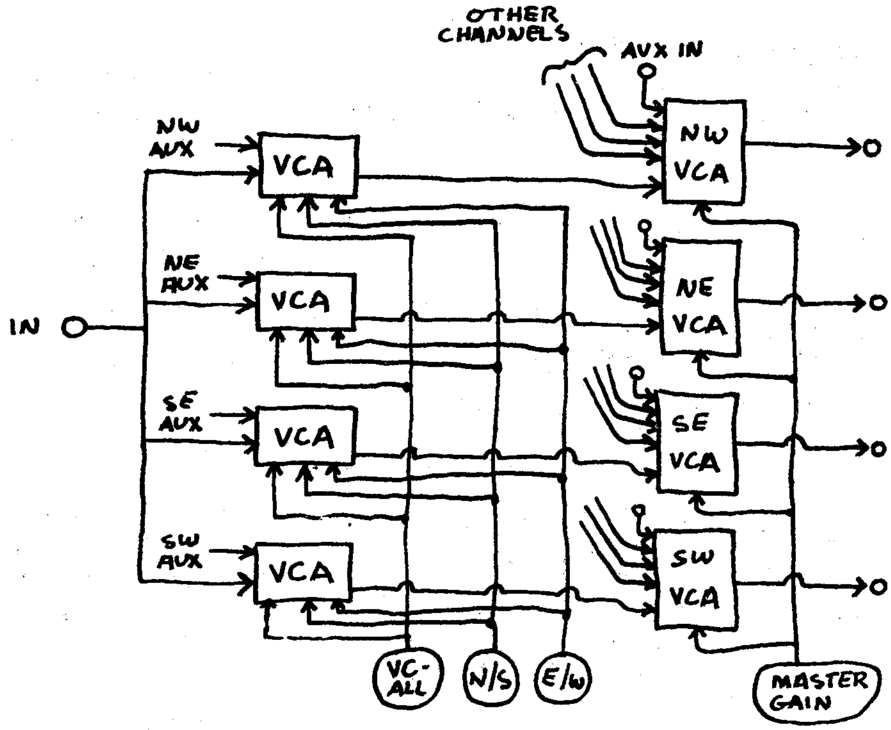
MULTI-CHANNEL STEREO MIXER (SMX).The SMX is a 2X input, 2-out mixer where X is the number of Dual Stereo Panner modules. Typically there will be three or four of these input modules. Each input channel is identical to the input channels of the QVM with individual gain control pot and VC input; and pan 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

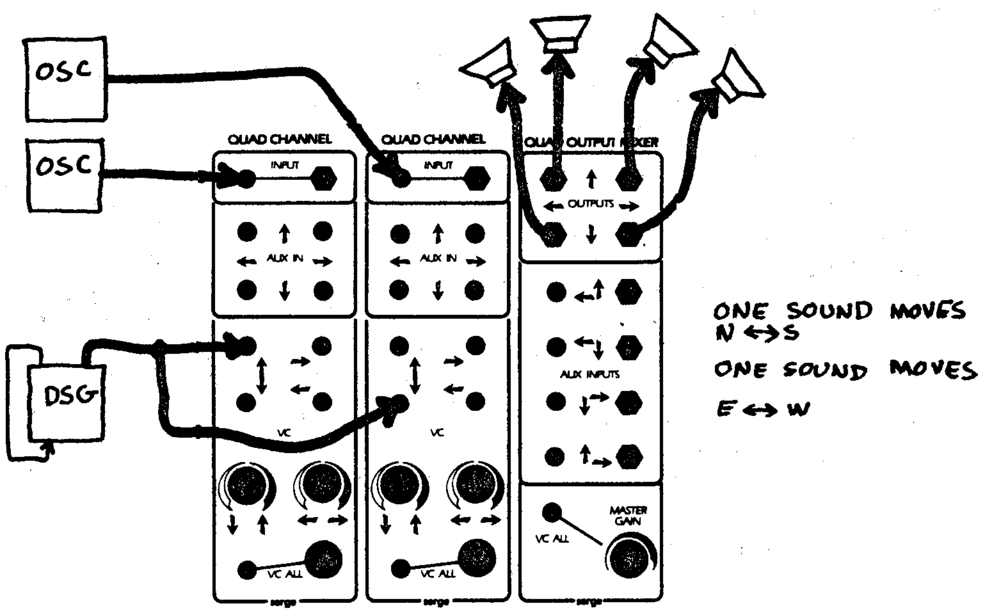
MULTI-CHANNEL QUADRAPHONIC MIXER (QMX). The QMX is an X-input, 4-output mixer where X is the number of Quad Channel inputs (from 2 to 7). Each input channel can be sent to any of the four output channels, either with manual or voltage control. Each input channel has four AUX-INs which are dedicated inputs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

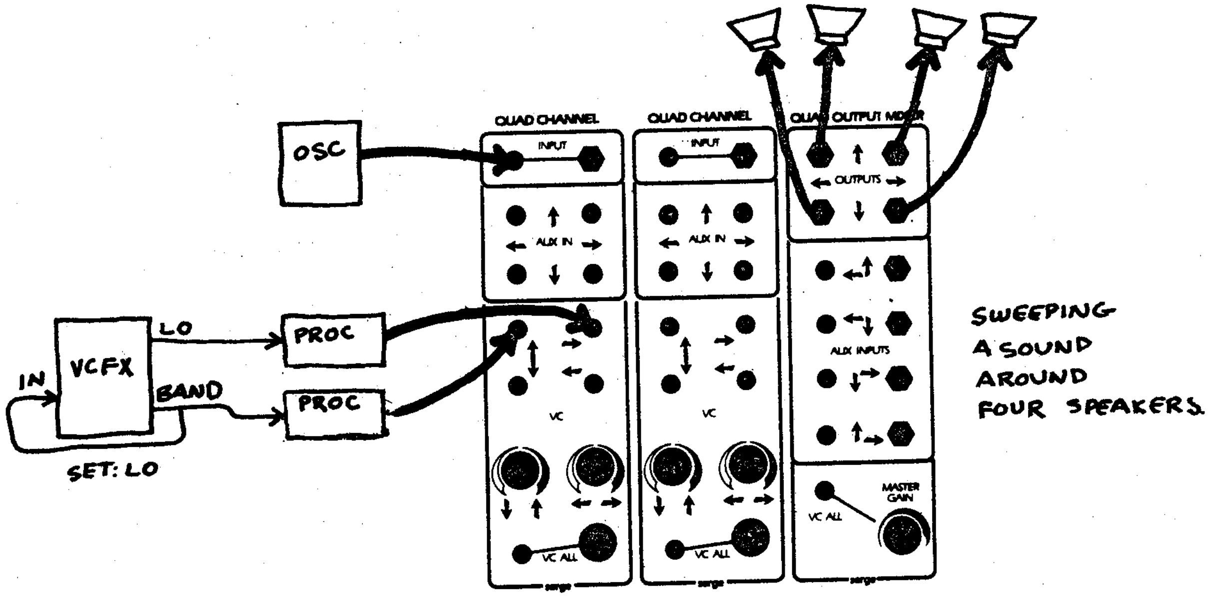


1 INPUT CHANNEL OF QMX

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.

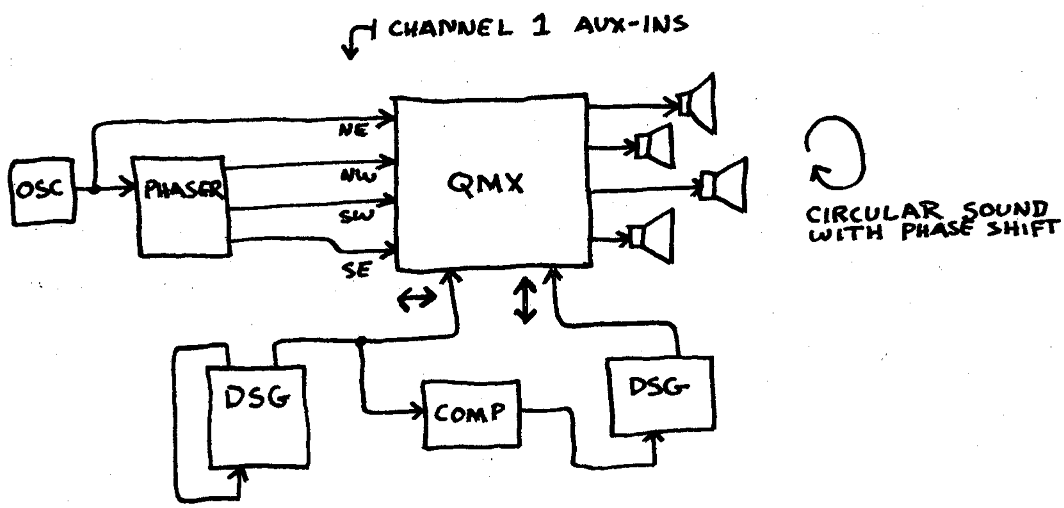


To achieve a perfect rotation of a single sound, two sine waves of identical frequency, 90 degrees out of phase are required. These can be obtained using the VCFX filter by patching the Band output to the IN and setting the filter to its low range. Sine waves 90 degrees out of phase are available at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



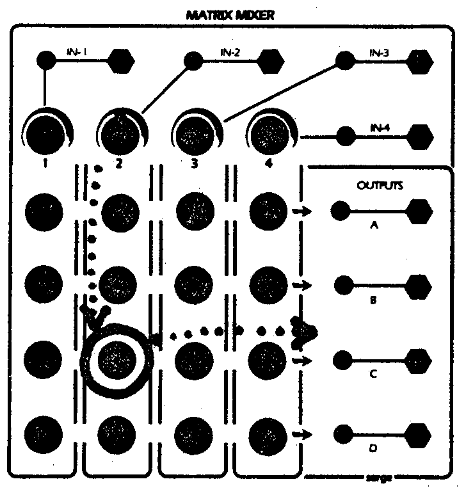
The high quality of the QMX allows rotation of a sound at audio frequencies providing a "spatial modulation" (SM).

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.



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.

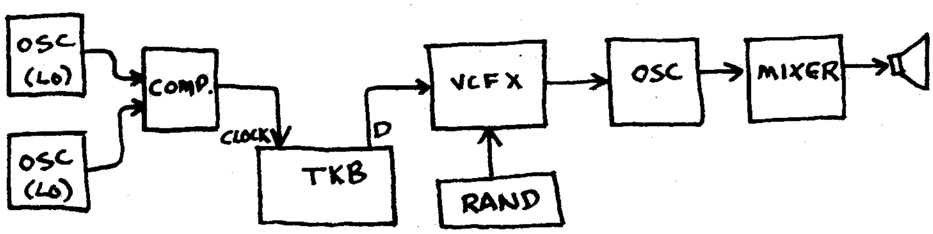


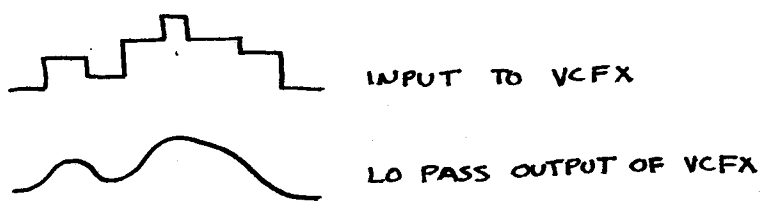
Each input has an overall gain control so that that input can be attenuated equally throughout the mix.

CONTROL VOLTAGE PROCESSORS

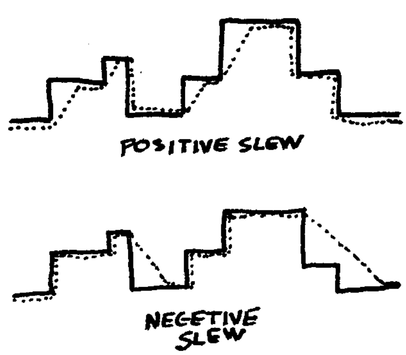
Because there is no difference, other than frequency, between audio voltages (AC) and control voltages (DC) it should not seem too strange that it is possible to have modules which process control voltages in much the same way that there are modules which process audio voltages. Nor should it seem odd 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.



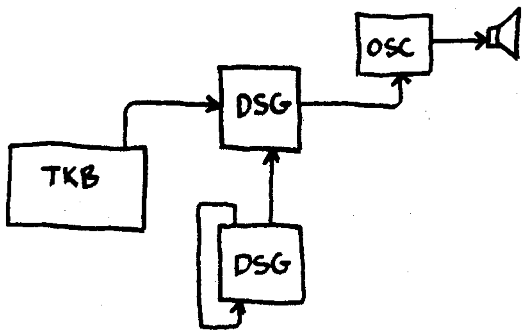


DUAL UNIVERSAL SLOPE GENERATOR (DSG). The DSG can act as a Positive or Negative or Positive/Negative SLEW or Portamento device. A Slew is a device which slides from one voltage to another voltage; a Positive slew affects positive-going voltages (not just positive voltages, but changes in a positive direction),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

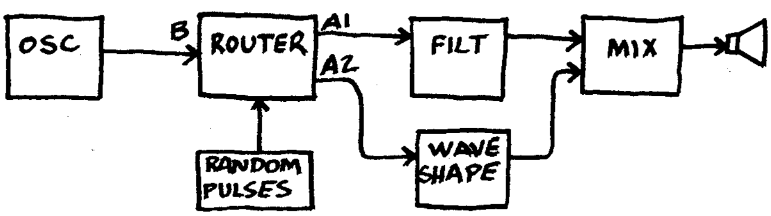


BI-DIRECTIONAL ROUTER. (ROU). This unique triple module can be used in one of two ways.

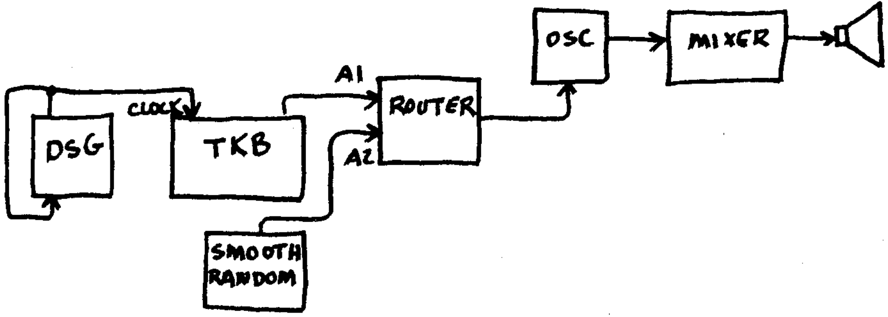
ONE-IN TWO-OUT SWITCH. An input at B can be sent to either output A1 or A2 depending on the state of input A1-A2. If A1-A2 is HI (+5 volts) then B appears at A1, otherwise it appears at A2.

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.

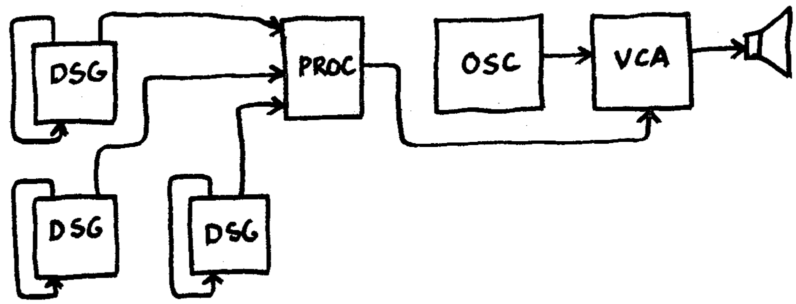
Note that if there is an INPUT at B there CANNOT be an input at either A1 or A2. This would, in effect, short outputs together.



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.



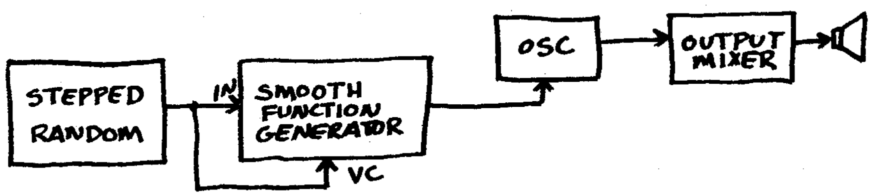
DUAL CONTROL VOLTAGE PROCESSOR (PRC). The Processor is to control voltages what a Mixer is to audio voltages. It can be used to sum together up to three control voltages. Each of its three inputs can independently be attenuated, amplified and/or inverted. Furthermore a manual offset pot sets a fixed 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.



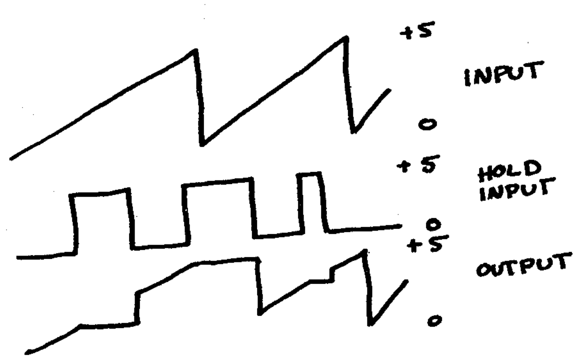
SMOOTH AND STEPPED FUNCTION GENERATOR. (SSG). The SSG is a dual module that contains two different and independent modules (though an internal coupler is provided). These are very versatile modules that can be patched to process control voltages in a wide variety of ways.

THE SMOOTH FUNCTION GENERATOR

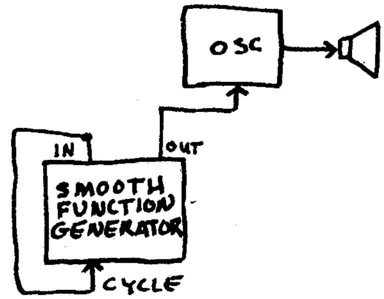
SLEW LIMITER. The Smooth Function Generator serves as a voltage controlled Slew Limiter on its Input. The slope of the slew (both positive and negative) is determined by the manual RATE pot and a VC input with associated attenuation pot.



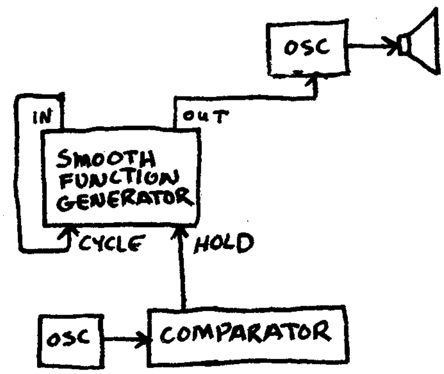
TRACK AND HOLD. A Hold input is provided on the Smooth Function Generator. When this input receives a HI voltage it has the effect of HOLDING the present output voltage level until the HOLD input goes low. If the RATE is set very fast so that the Smooth Function Generator follows its input closely (that 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.

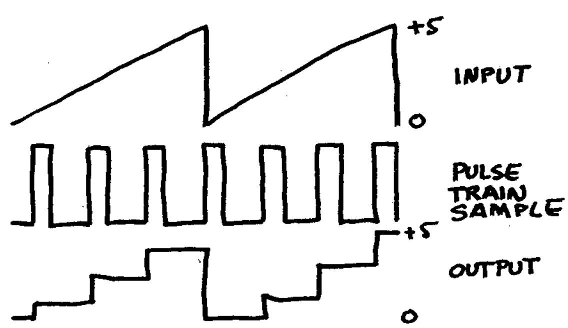


When the Smooth Function Generator is patched to cycle,the HOLD function remains operative to be able to produce up and down staircase-like voltages.

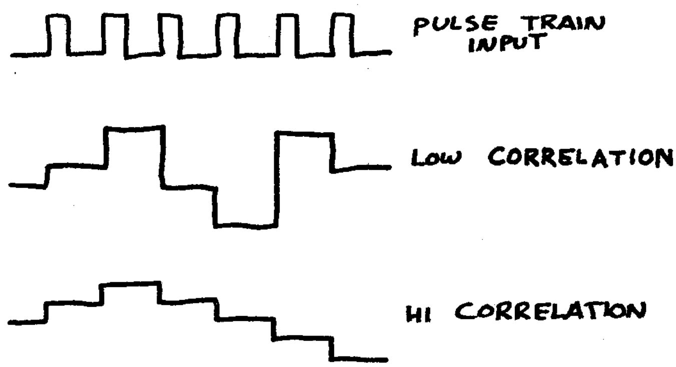


STEPPED FUNCTION GENERATOR

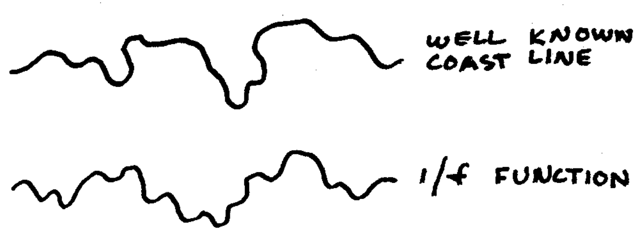
SAMPLE AND HOLD. A Sample and Hold is a device which produces a discrete stepped waveform from a changing input voltage. When a pulse is received at the SAMPLE input, the voltage appearing at that instant at IN appears at STEPPED OUT and is HELD there until another pulse is received at SAMPLE, when this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



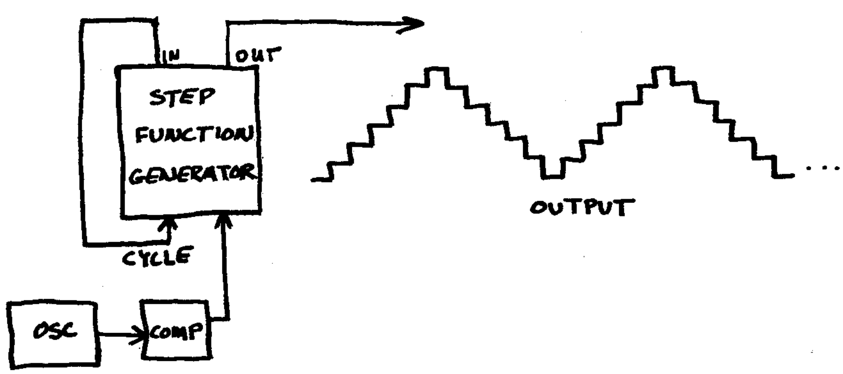
CORRELATION. The RATE pot and the VC input with its associated attenuation pot control the "correlation" of one voltage output level to the previous voltage output level. In the stepped voltage as correlation increases, each step must be closer and closer to the previous step.



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.



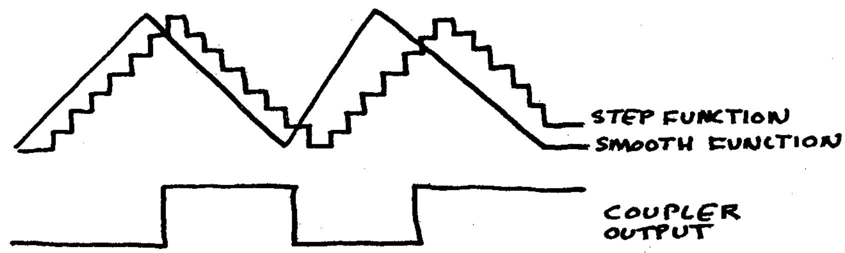
STAIRCASE GENERATOR. When the input is patched to the CYCLE jack and pulses applied to the SAMPLE, a complex staircase wave is generated at STEPPED OUT, determined by the pulse frequency, the position of the RATE pot, and a VC.



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.

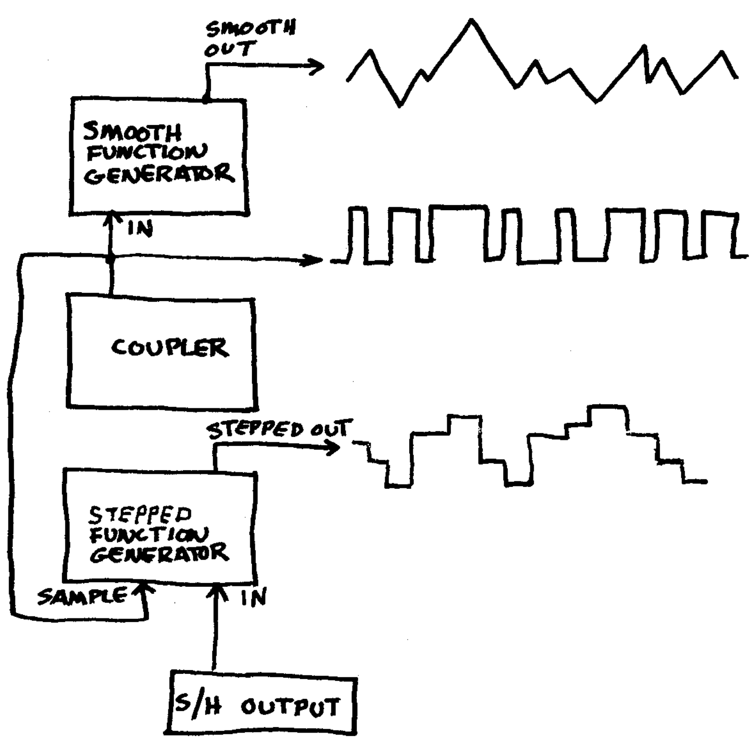
COUPLER

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.



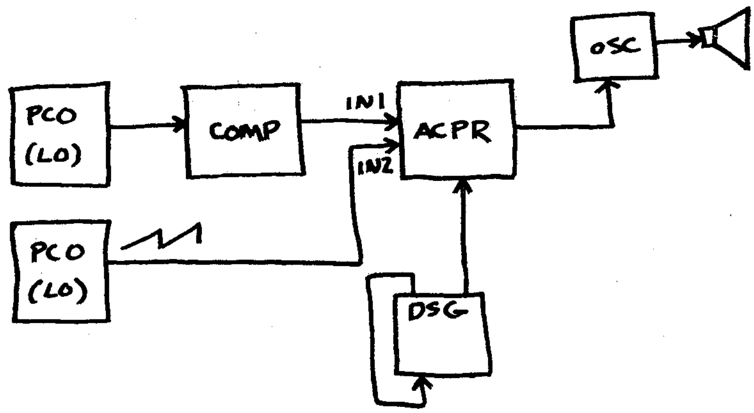
The S/H Output of the Noise Source is a randomly modulated sawtooth wave. This form of wave is good for generating random voltages with sample-and-holds. Since this wave is always going from 0 volts to +5 volts, there is an equal probability for any voltage to be selected. Since the frequency is random,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

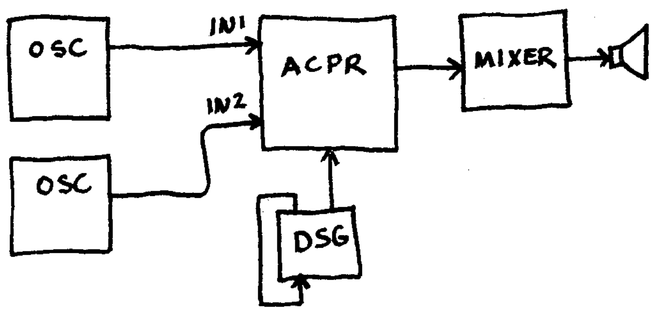


ACTIVE PROCESSOR. (ACPR). This module contains two separate and different modules. The lower module is a simple processor that allows the user to invert a control voltage and apply an offset. This module is useful for processing control voltages sent to modules without processing inputs.

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.

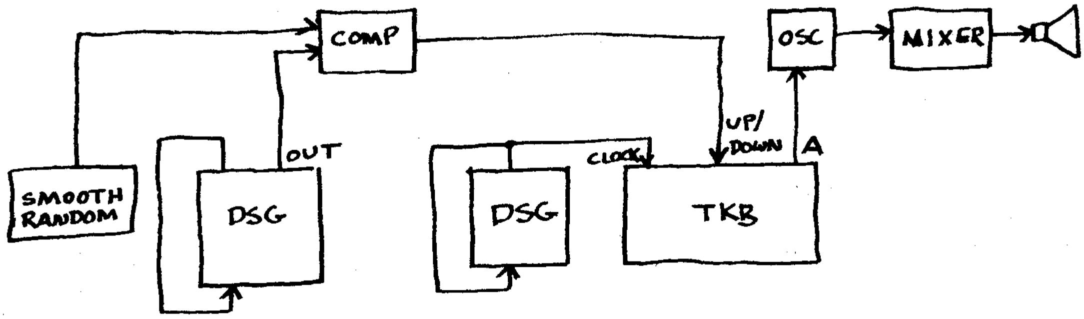


TOUCH ACTIVATED KEYBOARD SEQUENCER:(TKB). It is useful to think of the TKB as composed of two separate parts: The Sequencer and the Touch Pad Controller.

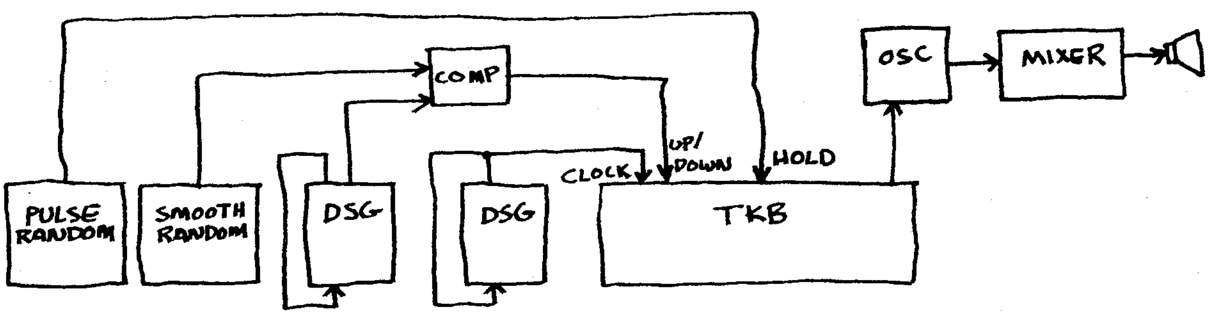
16-STAGE Sequencer. Only one stage can be on at any given instant. Each stage controls a specific column of pots, so that if stage #7 of the Sequencer is on, it activates all four pots in stage #7.

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.

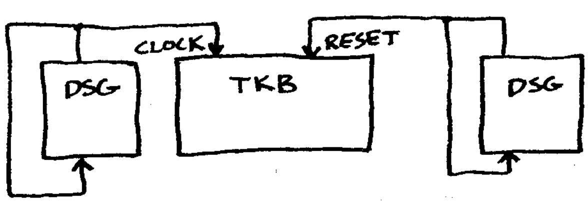
UP/DOWN. If a HI voltage is applied to the UP/DOWN input, the Sequencer will step DOWNWARDS instead of upward when it receives a CLOCK pulse. If it is on stage #1, it will wrap-up to stage #16.



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.

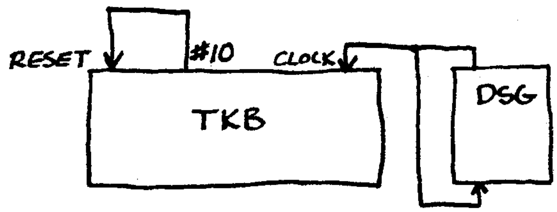


VARIABLE LENGTH SEQUENCES. It is often desirable to have sequences shorter than sixteen stages, or to have variable length sequences. For these purposes two RESET inputs are available at the top of the TKB. The RESET is triggered by a pulse from other pulse outputs on the TKB or by pulses from other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



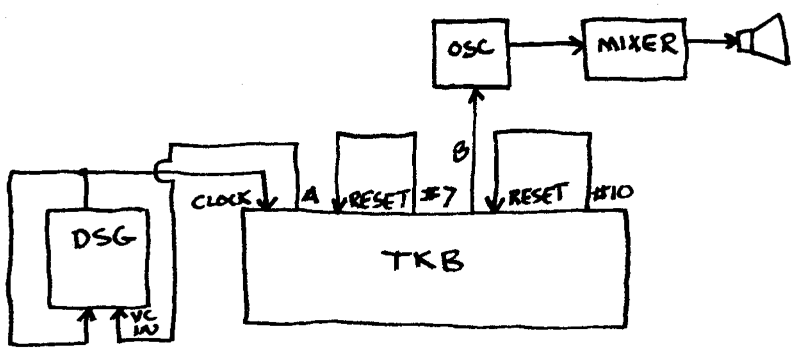
In the above patch, to get the TKB to RESET to Stage #1 you must first touch Keypad #1. The number of stages clocked advanced before resetting is determined by the Transient generator which is Triggering the RESET. Set this Transient generator so that about four stages are clocked through before RESETting 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.

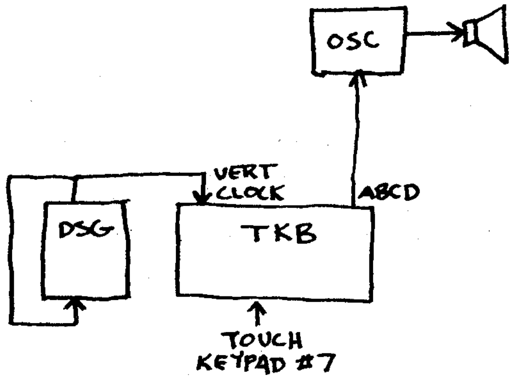


In this patch, just like in the previous one, the sequence will return to stage #1 only if that was the last keypad touched, otherwise, it will reset to whatever pad was last touched. In the above patch you can get the sequence to activate stages 5,6,7,8,9 and then reset to 5 by tapping stage 5's keypad.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

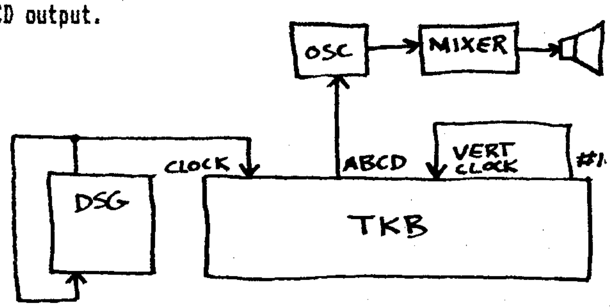


VERTICAL SEQUENCER. The TKB contains a second Sequencer, this one in 4 stages, which is clocked independently of the main 16 stage sequencer. The clock input for this sequencer is labelled VERT CLOCK. The VERT Sequencer's output is labelled ABCD. Every time a Trigger is received by the Vertical Sequencer 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.



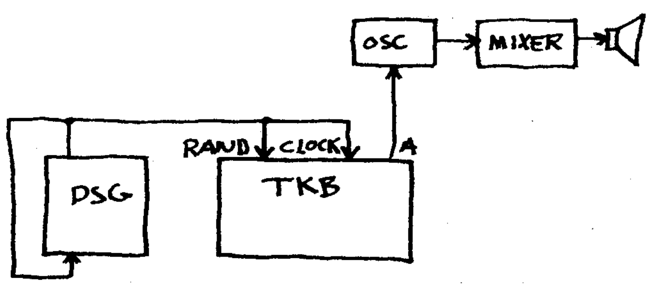
If in the above patch, stage #7 is now activated by touching its associated keypad, the four pots in stage #7 will determine ABCD's output.

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.



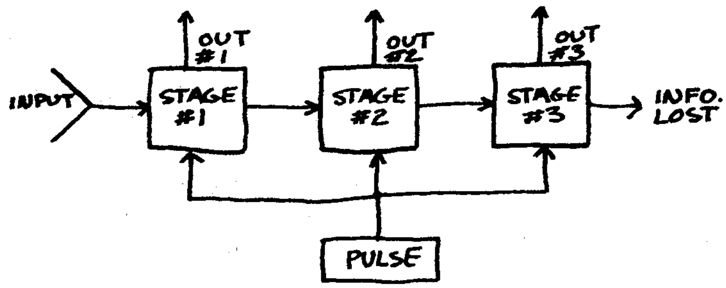
Consider what happens when the vertical sequence starts on Row A. Since the main sequencer clocks through all sixteen stages, all of Row A appears at output ABCD. When the main Sequencer gets to Stage #1, its pulse out, steps the Vertical Sequencer down one row, to Row B. The Main (horizontl) Sequencer 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.

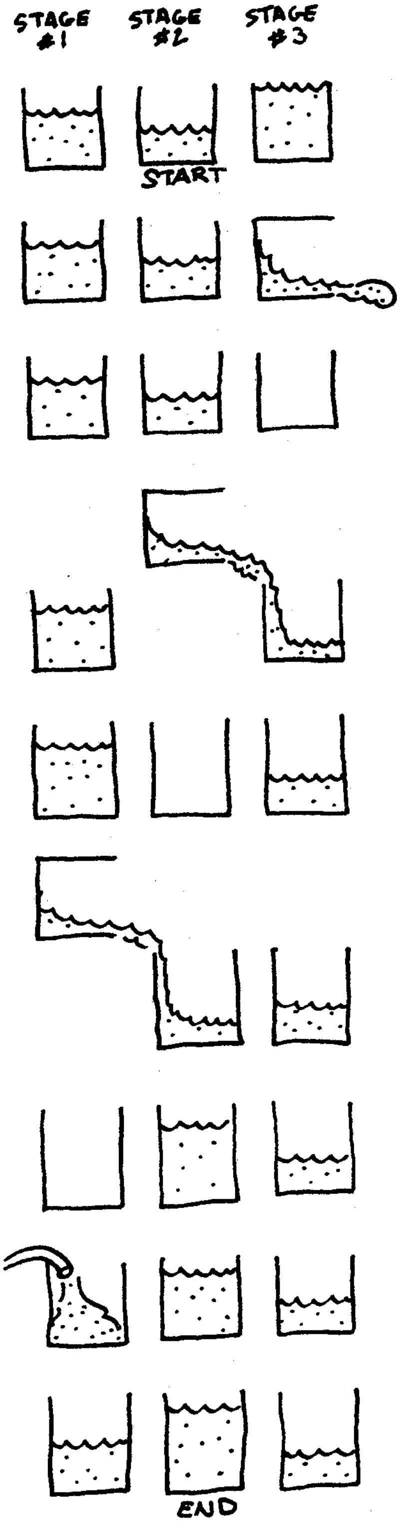


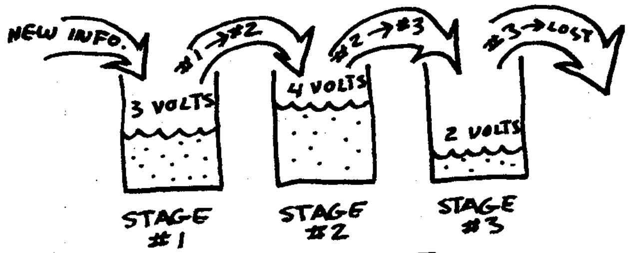
TOUCH PADS. The 16 Touch Pads can be used to interact with the 16-stage Sequencer, or can be used independently. Some of its interactive functions have been discussed already. It is important to note that the KV, KP, and PRESSURE outputs are always selected by the touch pads and not the Sequencer except 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.

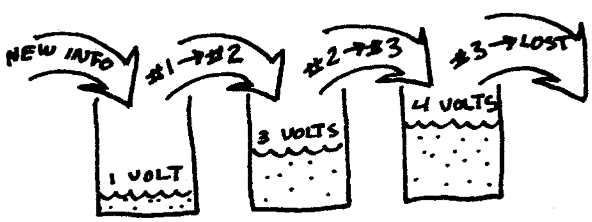


OVER-ALL ACTION

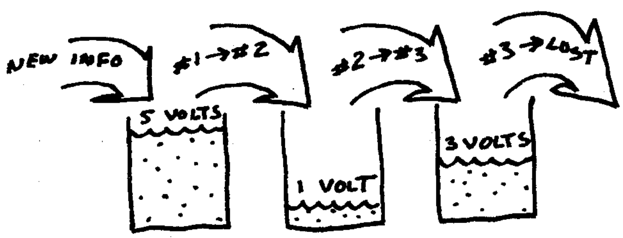




PULSE ONE



PULSE TWO



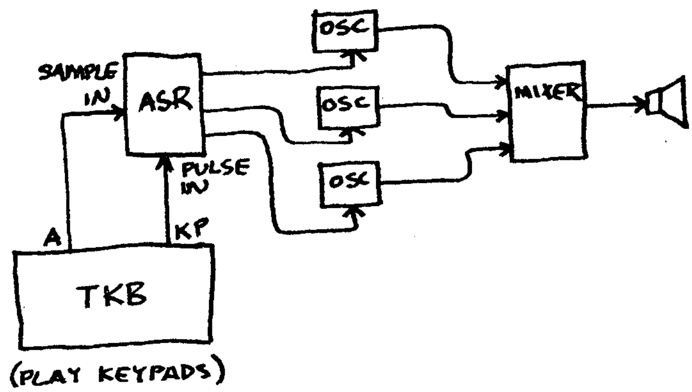
PULSE THREE

(ETC.)

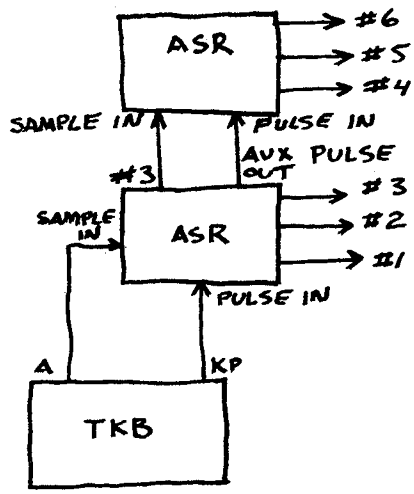
ACTION WITH A SERIES OF PULSES

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.

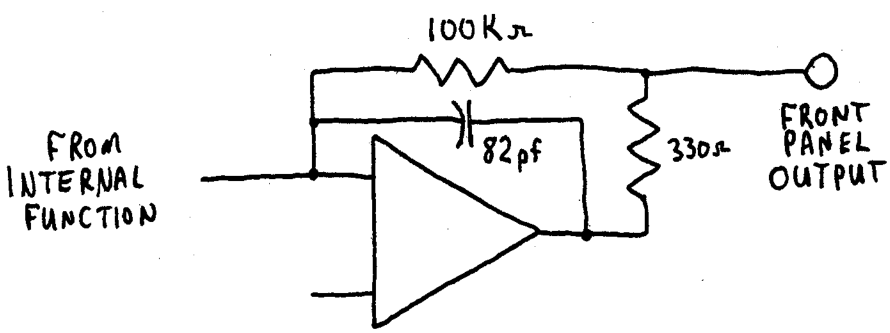
Some inputs and outputs of external devices have "balanced" lines with two signal lines and one ground line. Typically these lines use Cannon Connectors. Many microphones have balanced lines which permit much longer cables to be used before hum becomes noticeable. These lines must be unbalanced before 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

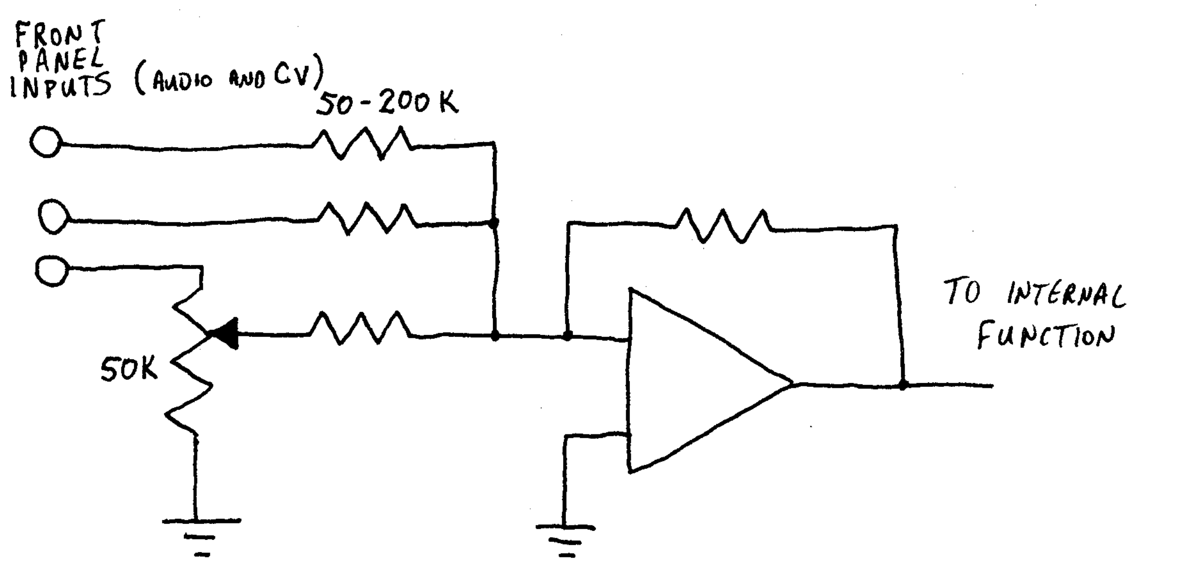
EXTERNAL CONTROL VOLTAGES

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

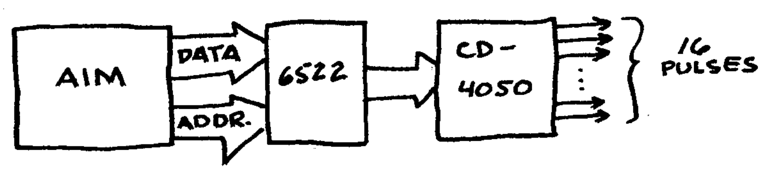
Connecting the Serge system to a small computer requires suitable buffers to prevent the higher voltages of the synthesizer from being applied directly to the computer or its associated circuitry. Most micro-computers have a single +5 volt power supply, and if negative voltages or voltages greater than +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.



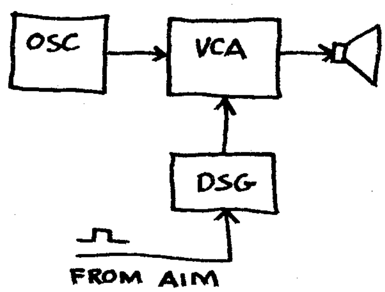
To turn on one output you can use the following 6502 Assembly Language routine:

```
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:

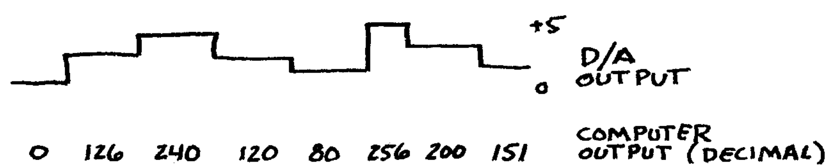
```
LDA MASK
EOR $FF
AND PORT
STA PORT
```

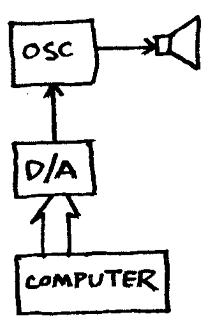
For one of these pulse outs to trigger a Serge device it must rise from a LO level to a HI level (since the Serge is rising-edge-triggered). To trigger an envelope generator repeatedly, the pulse out must return to a "0" level first.



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.

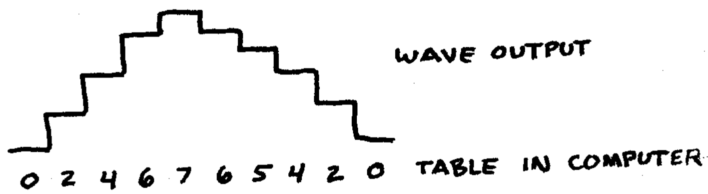




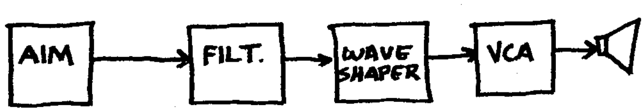
A D/A that can convert numbers from 0 to 255 into a voltage is fairly inexpensive. Such a D-to-A is perfect for creating presets and doing some kinds of voltage control. A D/A that can accurately control the frequency of an oscillator generally requires greater precision and is somewhat more costly. 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.

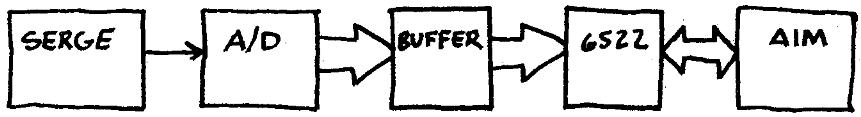


PULSE CONTROL OF COMPUTER:

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.