# WELSH'S SYNTHESIZER COOKBOOK
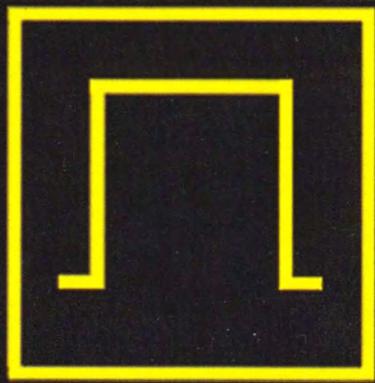
FRED WELSH

OSCILLATORS

Harmonics and oscillator waveforms

The most fundamental sound is the sine wave. Every sound whether it is natural or synthetic is made up of sine waves.
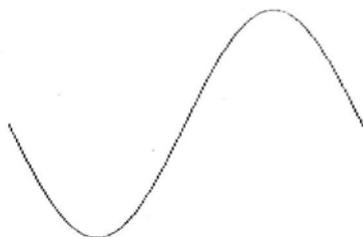

**Figure 1. All sounds, natural and
synthetic, are made up of sine waves**

The sound of a sine wave could be described as extremely mellow and not particularly interesting, but when numerous sine waves of different pitches and amplitudes are mixed together the sonic possibilities become endless.

It is possible to create *any* waveform using nothing but sine waves. Let's begin by creating a sawtooth wave. Each sine wave is going to have a different pitch and as they go higher in pitch they will decrease in amplitude. The first one will have the lowest pitch and it will in fact determine the overall pitch of the sawtooth. This is called the fundamental frequency. For the case of this example we will give it a pitch of 440 Hz corresponding to the A4 key.
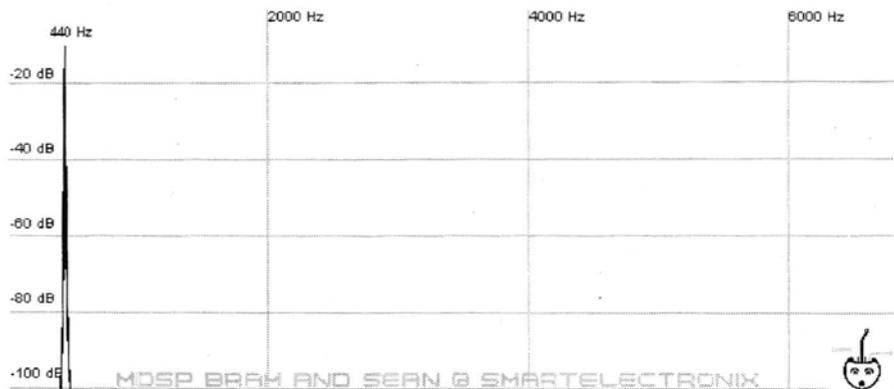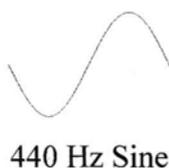
440 Hz Sine

Figure 2. A 440 Hz sine wave and a graph representing its frequency and amplitude

OSCILLATORS

The bottom portion of the figure is what is called a harmonic diagram. The horizontal axis is frequency/pitch and the vertical axis is amplitude. The spike represents the frequency and amplitude of the 440 sine wave. Let's add a second sine wave with a frequency of 880 Hz (440Hz x 2).
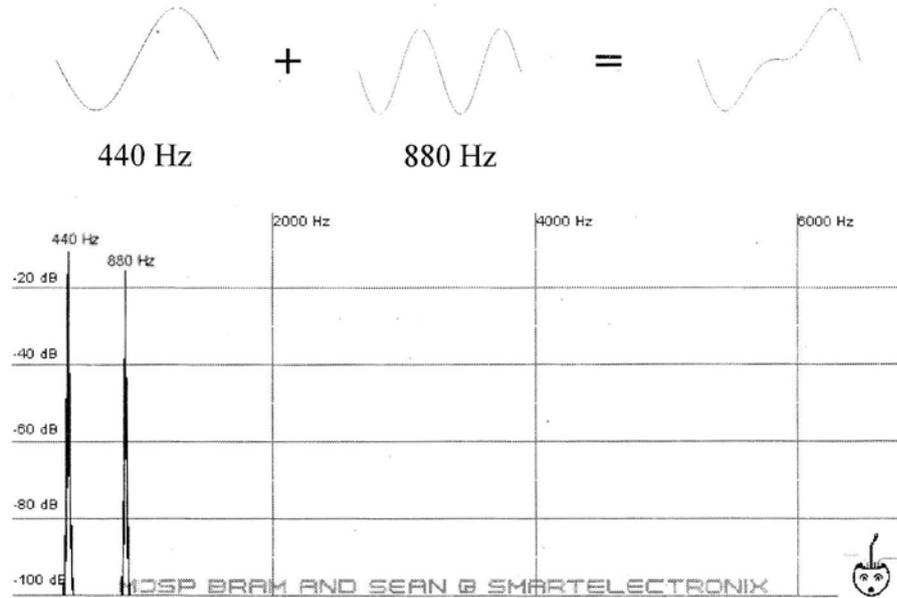

Figure 3. A 440 Hz sine wave added to a 880 Hz sine wave create a new waveform

Notice there is a second spike in the diagram representing the 880 Hz sine wave. Now add a third sine wave at a frequency of 1320 Hz (440 Hz x 3)
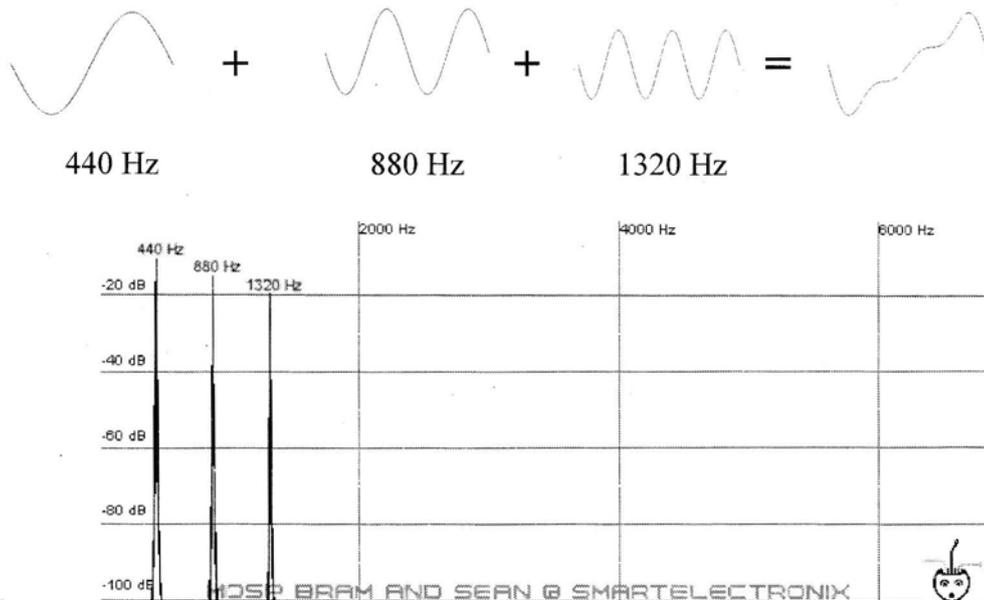

Figure 4. Three sine waves of frequency 440 Hz, 880 Hz, and 1320 Hz added together create a waveform that begins to look like a sawtooth wave.

Already the new waveform is beginning to lean to one side like a sawtooth wave. Keep adding sine waves that are frequency multiples of 440 Hz and eventually a sawtooth wave is produced.
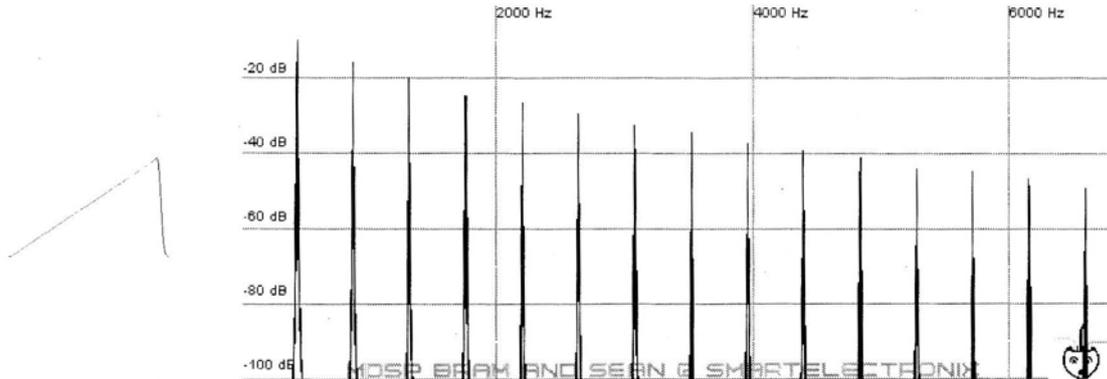
**Figure 5. A sawtooth wave and its harmonic diagram**

Each one of the harmonics(spikes) in the diagrams above are called partials. The first partial at the far left, as stated earlier, is known as the fundamental and it determines the pitch of the waveform. The other partials are all frequency multiples of the fundamental and they are known as overtones. Also notice that the fundamental is the loudest sine wave of all the harmonics and that as the overtones get higher in pitch their amplitudes decrease. There is of course nothing preventing us from mixing sine waves that all have the same amplitude it's just that in this particular case it is a requirement of creating a sawtooth that the amplitudes diminish with higher frequency. Just as a point of interest if we were to mix these harmonics all at the same amplitude it would produce something that as successively higher-pitched sine waves are added would at first look like a sawtooth wave with a bad complexion and as more harmonics are added would look nothing like a sawtooth at all. Harmonic amplitudes are important!

**Figure 6. Waveform produced by setting
the first 16 harmonics to equal amplitudes.**

A square wave can be created using the same process. The difference is that a square wave is only made up of odd-numbered partials.
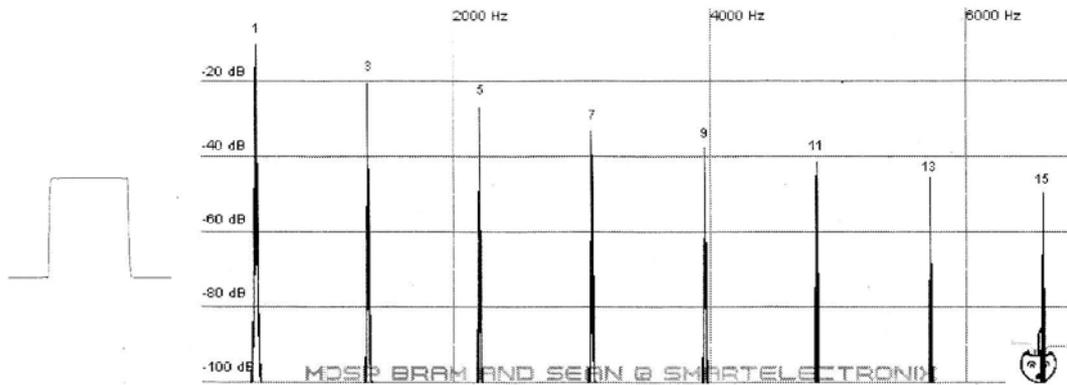
Figure 7. A square wave has only odd partials. This square wave was produced by an *actual* oscillator. Notice the slight imperfections in the waveform. No oscillator produces perfect waveforms nor do any two oscillators produce identical waveforms.

Triangle waves are also made up of odd-numbered partials but the higher harmonics are weaker than in a square wave.
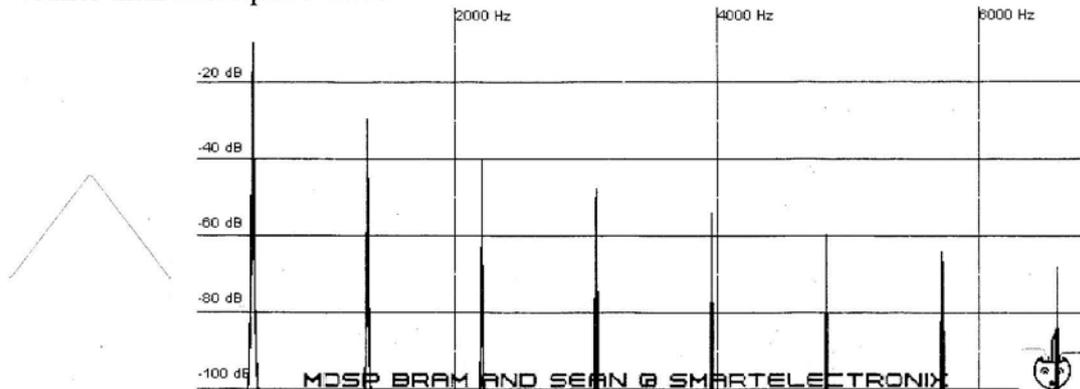
Figure 8. A triangle wave has odd harmonics that are weaker than those of a square wave.

The process of adding sine waves together to create sounds is known as *additive synthesis*. This method is used on a few digital synthesizers and soft synths and has also been used by pipe organs for hundreds of years. In a pipe organ each pipe produces a sine wave of a different pitch and by controlling the amount of air to each pipe it is possible to control the individual amplitudes of each sine wave which in turn makes it possible to produce sounds that are harmonically similar to other instruments. It could be said that pipe organs were the first synthesizers.

Analog synthesizers use a process called *subtractive synthesis* which is simply additive synthesis in reverse. Here's some synth terminonlogy for you: Sounds created by synthesizers are referred to as patches. This goes back to the early days of modular synthesizers when patch cables were used to route signals from module to module to

create a sound. Patches created using subtractive synthesis start with waveforms that are already rich in harmonics such as sawtooth, square, and triangle waves. These waveforms are then passed to a filter which removes harmonics from the waveforms in order to produce the desired sound. The harmonics are *subtracted* out hence the process is known as subtractive synthesis.

The most common waveforms on analog synthesizers are the sawtooth, square/pulse, and triangle wave. There are three main reasons these are used almost universally rather than other waveforms. First is the fact that they all have lots of harmonics which can be chiseled away by the filter. Secondly they are relatively easy to produce using analog circuits. Thirdly they are each harmonically similar to broad families of acoustic sounds even without any filtering. This almost sounds as if to imply that acoustic imitation is the goal of synthesis which it most certainly is not. It is a practical criteria to start with if nothing else. Sawtooth waves are similar to brass and string instruments, square waves are similar to woodwinds, and triangle waves with their diminished higher harmonics are good for mixing together to produce inharmonic sounds (bells, chimes, etc) as well as adding the occasional rogue harmonic to saw and pulse waveforms. This may be surprising but even though these waveforms have similar harmonics and sound similar to certain families of instruments the waveforms of acoustic instruments look nothing like squares/pulses, sawtooths, or triangles. A waveform's usefulness lies with the harmonics that make it up and is NOT inherent to the shape of the waveform. Two sounds can have very similar harmonics yet have waveforms that look nothing alike. Adding waveforms together in an attempt to match the waveshape of an acoustic instrument won't produce desirable results most of the time.

While some synthesizers only have one oscillator and a few others have many, the majority have two oscillators that can be independently set to different waveforms. Two-oscillator configurations are so common because they are cheaper to produce than multi-oscillator synths yet they are much more capable of producing rich sounds than single-oscillator instruments. They are a good balance of economy and ability. By using two oscillators it is possible to not only create more interesting strings, brass, and woodwinds but it also becomes possible to create vocals, inharmonic sounds such as bells, and countless other acoustic and synthetic sounds. One simple fact is that a single oscillator will have harmonics where each successive partial has less amplitude. While the higher harmonics of acoustic instruments do diminish in amplitude they do not follow this progression so rigidly. Below are harmonic diagrams of some acoustic instruments.
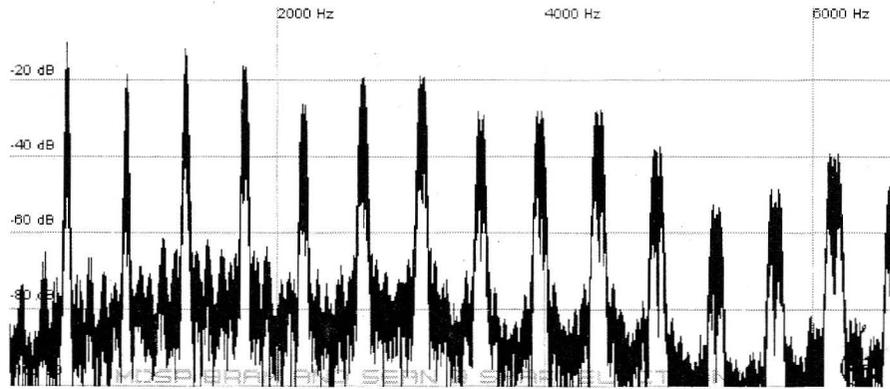
**Figure 9. Violin harmonics**
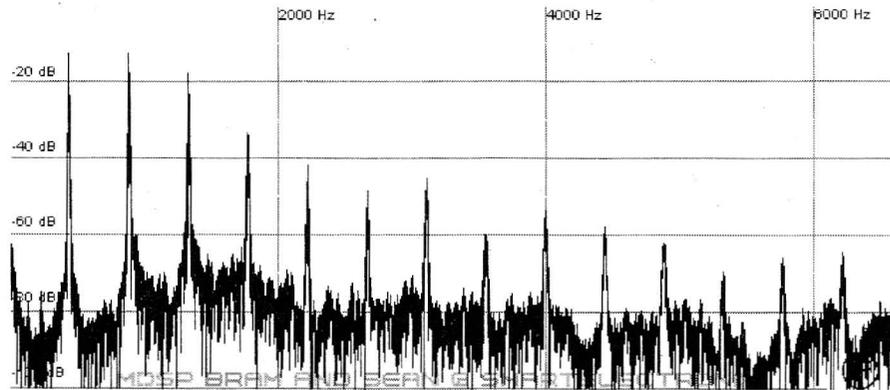
Figure 10. Flute harmonics
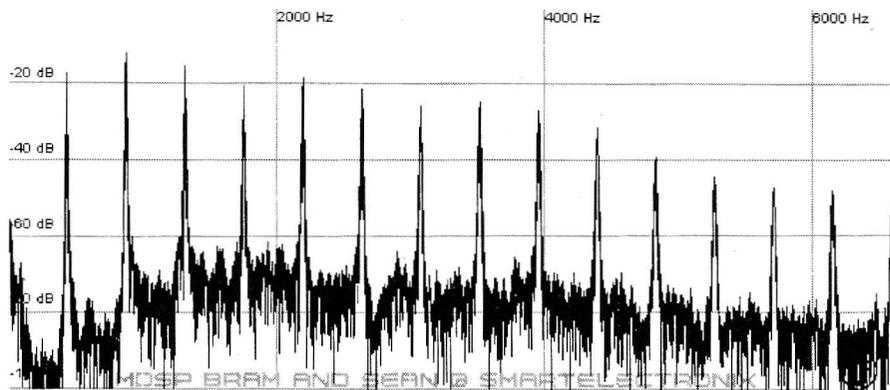
Figure 11. Trumpet harmonics

The noise at the bottom of the graphs is from resonance, reverberation, and recording noise.

Notice that while the higher harmonics do die away they do not do so in a uniform fashion like synthetic waveforms. These slight harmonic deviations help separate one acoustic instrument from another in the same family and give what could be considered a "natural" sound. Again, imitation is a possibility of synthesis but is not necessarily the goal, however lessons learned from studying and recreating acoustic sounds can help to produce more interesting synthetic patches that have character and life. By detuning oscillators it is possible to come up with much more interesting sounds. To produce the harmonics of a marimba for example take a triangle wave and add a second triangle wave at slightly lower amplitude and tuned up two octaves.
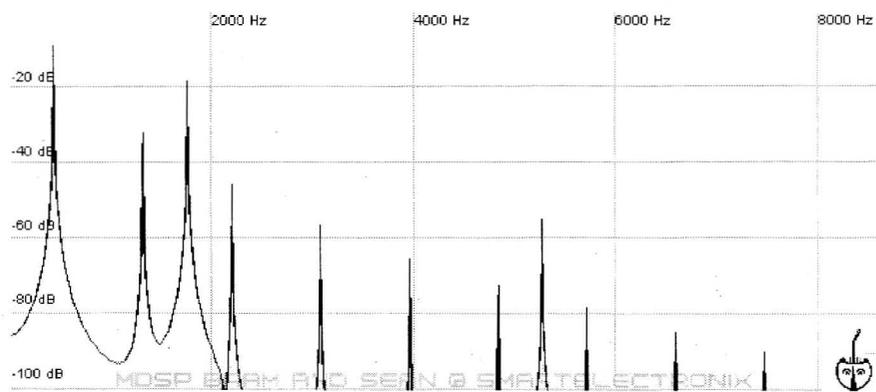
Figure 12. Using two oscillators to synthesize an instrument such as the marimba allows for a more complex harmonic structure than using single waveforms alone.

By simply mixing two humble triangle waves with different tunings a more harmonically sophisticated sound has been produced than could be done with just a single waveform. The harmonics of a single triangle may die out uniformly but the harmonics of two detuned triangles do not. While this sound is literally the sum of two triangle waves, to the ear it sounds far richer than the sum of its parts. The important point here is that using two oscillators gives more control over the harmonic make up of the final sound. As we shall see there are many other tools at our disposal that will allow us to further fine tune harmonic structure.

PW = 50%
(square wave)
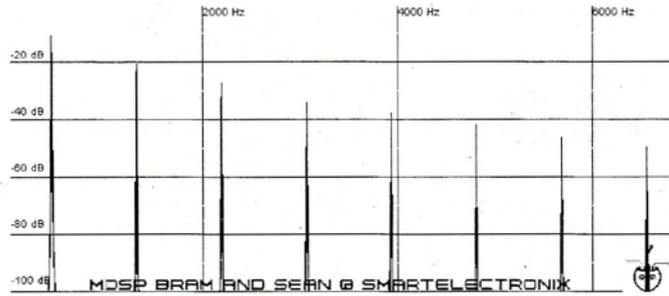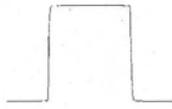
PW = 40%
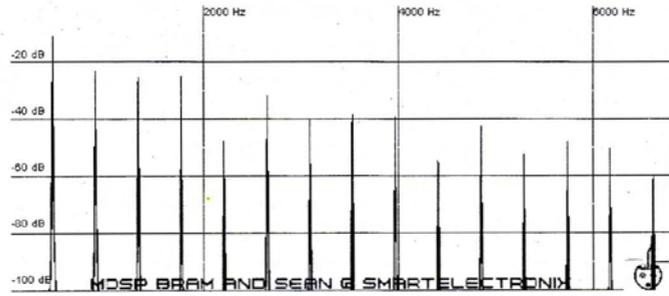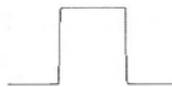
PW = 30%
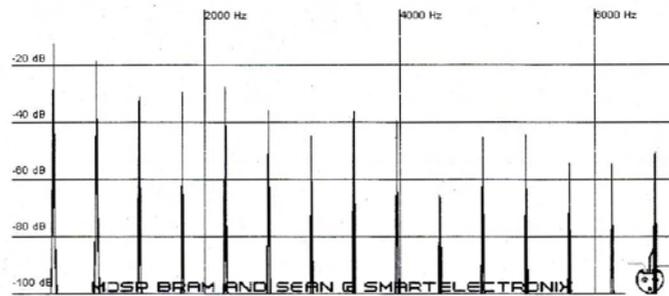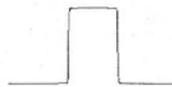
PW = 20%
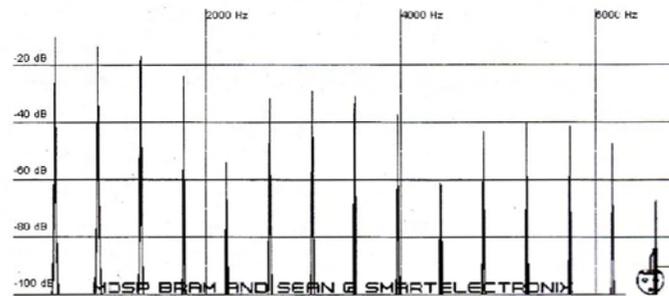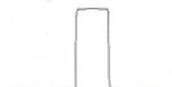
PW = 10%
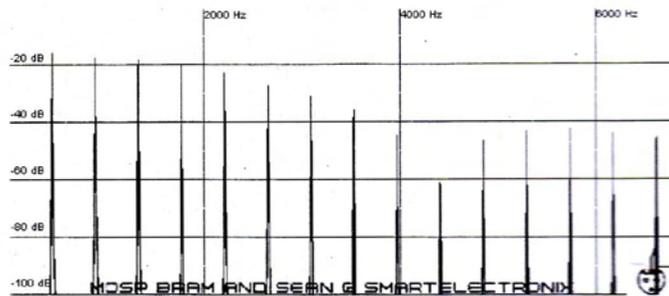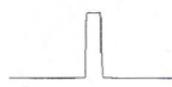
Figure 13. Pulse widths. Narrowing the pulse width creates a brighter, edgier sound similar to a sawtooth.

As a pulse becomes narrower it goes from a square wave with only odd partials to having both even and odd partials and therefore sounding more like a sawtooth wave. Look at the figures above and you'll notice that even though narrow pulses have harmonic spectra that are similar to a sawtooth there are still some important differences. The harmonics do not die out as quickly or with the same uniformity as those of a sawtooth. Narrow pulses have more of a crisp, metallic sound. By narrowing the pulse width it is also possible to give the harmonics an "undulating" pattern similar to the harmonic behavior of many acoustic instruments. The advantage of a pulse with variable width is that it allows a higher degree of control over the harmonics than using just waveform mixing and filtering alone. Another cool thing about variable-width pulses is that the width can usually be modulated to produce thick string-like patches. More about that in the section on LFOs. Some synths don't provide the ability to change the width of the pulse, but these instruments often provide a few pulses that are hardwired to different widths. This is common on ROMpler, sample-based instruments and even a few analog synths.

Syncing

Another common feature with most dual-oscillator analog synths is the ability to sync one oscillator to the other. This is useful for producing, amongst other things, plucked and hammered strings such as pianos and guitars. Syncing forces the wavecycle of one oscillator to restart when the wavecycle of the other oscillator restarts. The oscillator that is being synced is called the slave while the oscillator it is synced to is called the master. Usually, but not always, a synthesizer's second oscillator is the slave while oscillator one is the master. You may want to check your synth's manual to be sure since it's often not clear on the front panel. For syncing to have any effect on the sound the synced oscillator must be tuned *higher* than the master oscilltator.

When the slave oscillator is tuned higher than the master oscillator its waveform will restart each time the waveform of the master restarts. The resulting waveform produced by the slave using a sawtooth would look similar to that shown at the bottom of the following figure. When synced, the tuning of the master also controls the *actual* tuning of the slave while any changes that are made using the slave's tuning/pitch control will instead change the tonal qualities of the oscillator.

cycle set by master

MASTER

**SLAVE
UNSYNCED**

**SLAVE'S
SYNCED
OUTPUT**
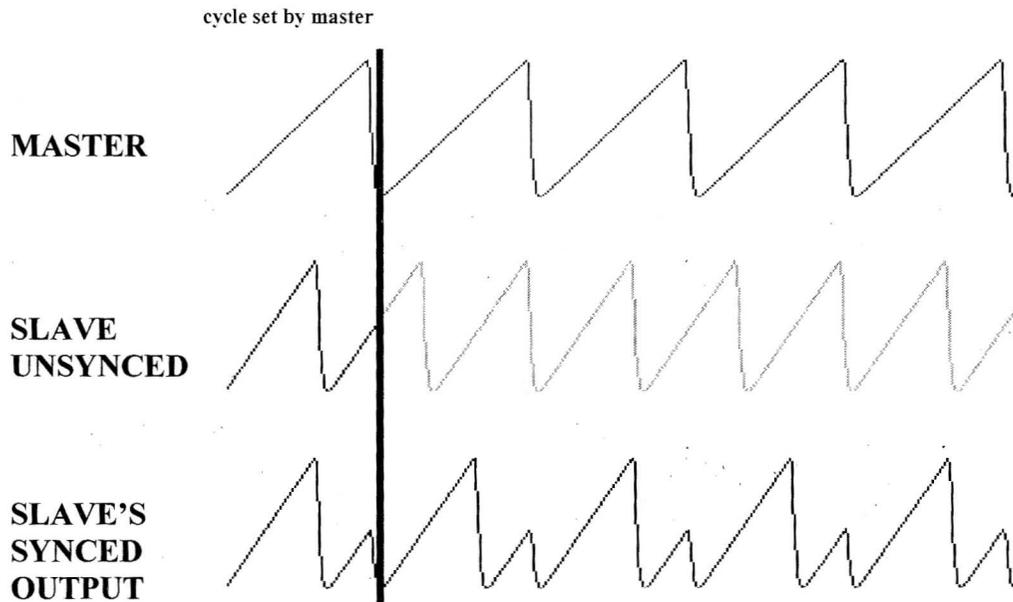
Figure 14. When a slave oscillator is synced its waveform will restart every time the master oscillator restarts.

Here's an example that you may want to try for yourself. Start by setting the slave oscillator to produce a sawtooth waveform. The waveform used by the master doesn't matter. Only its pitch matters since it controls the wave cycle time. Turn the volume of the master oscillator all the way down but make sure it is turned on. We want the master to be available for syncing the slave but we don't want to hear the master's output. Set both oscillators to the same pitch. If you now turn syncing on and off you will notice that the output of the slave sounds the same either way. This is because the master is making the slave's waveform restart at the same location that it would even if the two weren't synced since the pitch settings are identical.

Next turn syncing on and slowly increase the pitch of the slave without going as high as one octave above the pitch set by the master. As it increases the output of the slave will take on a thin, brittle timbre, however the overall pitch stays the same.

Turn the slave's pitch control up to the point where it corresponds to exactly one octave higher than the pitch setting of the master. If we now turn syncing off we will hear no change. The brittle timbre is gone and you are now hearing a plain sawtooth only it is an octave higher in pitch. Even though the slave is indeed tuned higher than the master it's pitch is now exactly double that of the master and therefore the master is causing the slave to once again restart at a point where it would restart even if it weren't synced. A further increase of the slave's pitch will continue to produce metallic, harmonically-rich sound until the slave is pitched two octaves above the master, and so on. To summarize: The slave must be tuned higher but if it is tuned at octave intervals syncing will have no effect. Look at the figures below. As the slave's pitch setting is increased the harmonics change, but when the slave's pitch is set to one octave above the master the output simply becomes that of a plain sawtooth (or square) tuned one octave higher in pitch.

SYNCED SAWTOOTH

Slave's pitch = Master's pitch
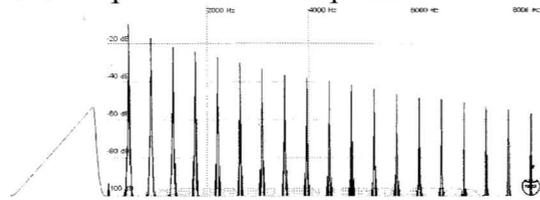
SYNCED SQUARE
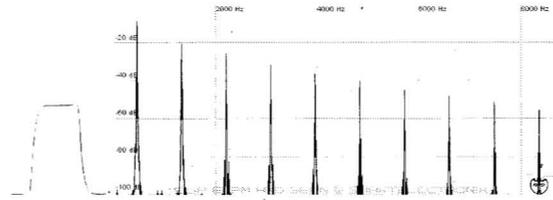
Slave = Master + 4 semitones
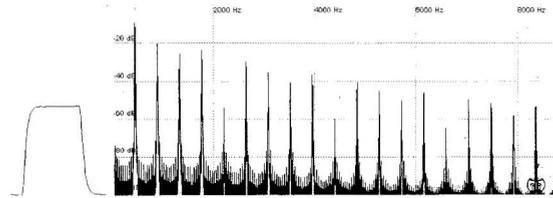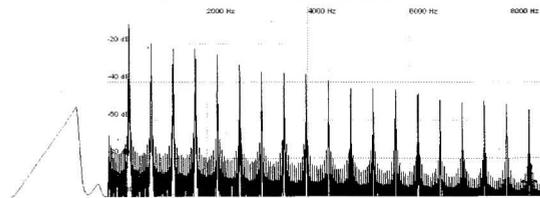
Slave = Master + 8 semitones
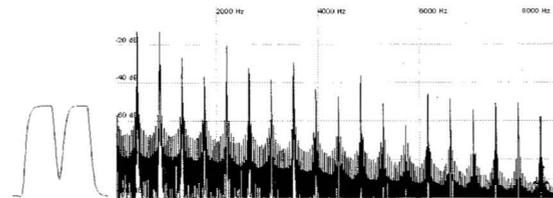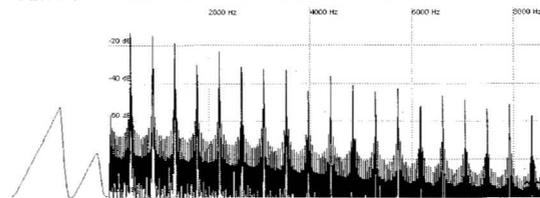
Slave = Master + 12 semitones (1 octave)
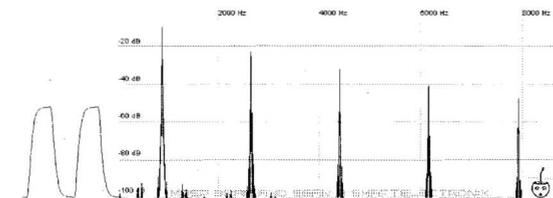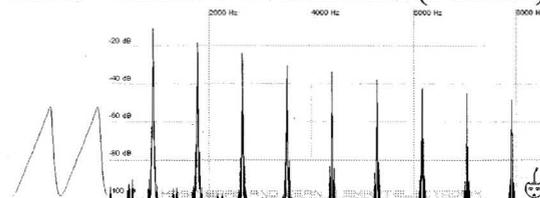
Slave = Master + 16 semitones
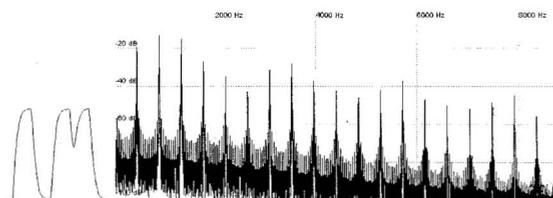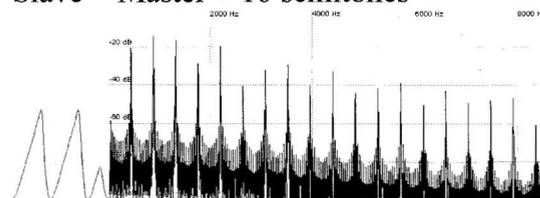
Slave = Master + 20 semitones
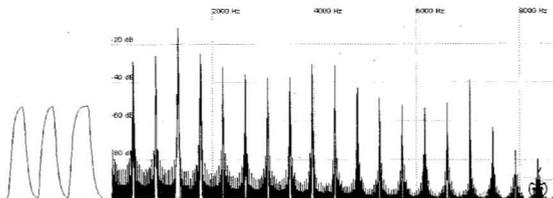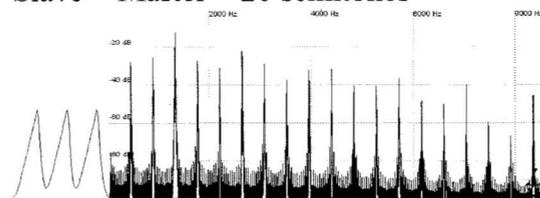
Figure 15. Harmonics of synced sawtooth and square waveforms are much more complex than unsynced waveforms. Detuning the slave from the master by octave intervals negates the syncing effect since the slave is forced to restart at a location where it would restart even if it weren't synced.

Noise

Subtractive synthesizers generally have two types of sound sources. One of these sources are the oscillators while another is noise. On some models the noise source is actually provided by one of the oscillators while on others it is provided seperately. Noise is useful for creating many percussion sounds, creating sound effects such as ocean waves and wind, and driving filter resonance. The most common form is white noise which maintains uniform power across all frequencies. Television static is white noise.
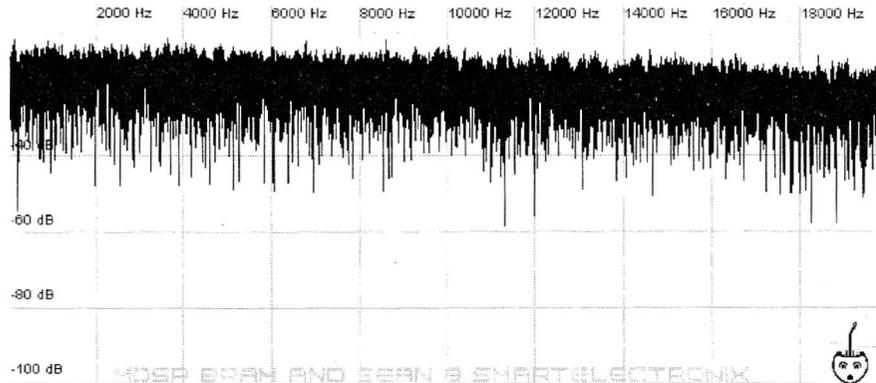
Figure 16. White noise

There is pink noise which diminishes in power by 3 dB per octave.
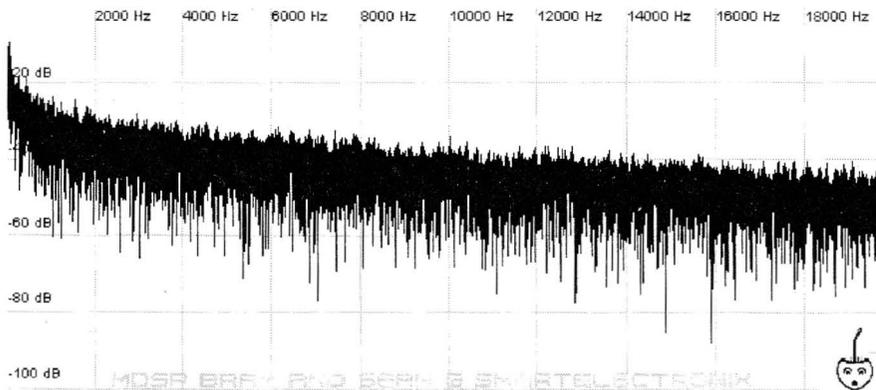
Figure 17. Pink noise

There is also brown noise which diminishes in power by 6 dB per octave. Put simply, white noise sounds brighter than pink noise, and pink noise sounds brighter than brown.
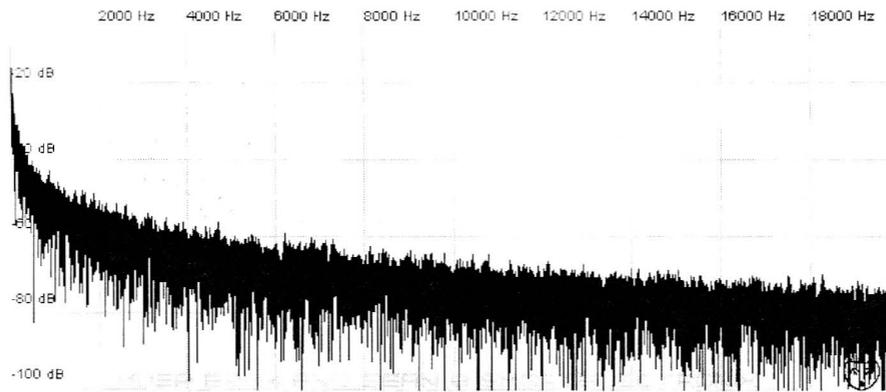
Figure 18. Brown noise

What distinguishes noise from a typical waveform is that it has no harmonic partials and the sound is spread solidly across the audio spectrum. True noise has no pitch. When noise is used to create percussive patches such as snare drums or cymbals it may use filter resonance or sometimes a triangle wave will be mixed in to impart a sense of pitch.

With a regular oscillator waveform pitch, pulse width, and syncing can all be controlled to manipulate the sound before it reaches the filter. Noise has none of these controls and therefore is highly dependent upon the filter for sculpting. Noise when used along with the filter, and the filter's resonance and envelope is far more useful than it may at first seem.

Keyboard Tracking

Oscillator keyboard tracking, sometimes called keyboard pitch tracking, allows the user to select whether or not the pitch of an oscillator will change as different keys are played. Turning this off will cause the same pitch to be sounded by every key. This can be particularly useful when creating bagpipes, sitars, and other drone instruments. Bagpipes and sitars are similar in the respect that they both play a set of notes of varying pitch over top a drone of unchanging pitch. Use one oscillator with tracking turned on to play the notes of varying pitch and use a second oscillator with tracking turned off to imitate the drone. The pitch control sets the single pitch that is played on all keys. It is quite common for an untracked oscillator pitch to correspond with the same pitch that would be played by either the C4 or A4 keys if tracking were turned on.

Keyboard tracking is sometimes offered as a control on other parameters as well in addition to pitch. After pitch tracking, the most common is filter cutoff keyboard tracking. This can be used to either gradually increase or decrease the amount of filtering that takes place as keys are played either up or down the keyboard. For example, if the high notes in a patch sound too bright keyboard tracking can be used to give them more filtering while letting the lower notes pass through unfiltered. Keyboard tracking is also occasionally used to modify envelope times and amounts. It can be used to give higher notes faster envelope times which can be useful for creating piano patches where high notes decay faster than lower ones.

Polyphony

Polyphony refers to the number of keys that can be played and still produce sound. For example, a synthesizer with five-note polyphony will produce sound for a maximum of five notes at a time. What would happen if more than five keys were played? This depends on the synthesizer. Some synths will "steal" notes that have been held the longest in order to play newer notes while other synths may steal the lowest-pitched, or highest-pitched notes. Many synths allow the user to choose the method of note stealing that they prefer.

Synthesizers that only allow one note to be played at a time are called *monophonic* while synthesizers that allow multiple notes are *polyphonic*. Most early synthesizers were strictly monophonic. Monophonic playing isn't always a limitation. In fact most modern synths give the user the option to choose between monophonic and polyphonic playing. Monophonic playing can be particularly useful for some leads and bass patches. It helps to keep a single patch from over-dominating a mix and sounds particularly interesting when used in conjunction with portamento.

Portamento

Portamento, also known as glide, will cause the pitch to change gradually between two consecutively struck keys. This is controlled by a time parameter which determines how long it takes for the pitch to change. There is often a setting that will determine whether glide is only present when the keys are played overlapping (legato) or glide is automatic whether the keys overlap or not.

Unison

Unison is used to create a chorusing-like effect. Historically this has been achieved by using all of a polyphonic synth's oscillators to play the same note and would effectively make the instrument monophonic when in unison mode. The fact that true analog oscillators cannot keep *exactly* the same tuning becomes an asset here. Because all of the oscillators are slightly detuned from each other over the range of a few cents the sound is very thick. The sound is also somewhat similar to that produced by pulse-width modulation. The graph below shows a sawtooth wave being played in unison. The higher harmonics are broadened similar to the way they would be in pulse-width modulation. Unison unlike PWM, as we will discuss later, doesn't flatten out the harmonic levels and it also adds some low-amplitude, inharmonic content. Notice how the harmonics are broadened similar to those of the violin graph at the top of page 6. Unison is often used to create bowed-string sounds.
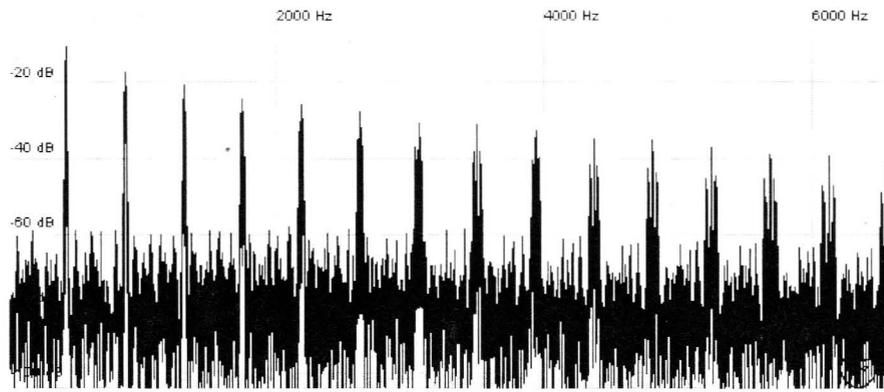
Figure 19. Unison broadens the harmonics of a waveform creating a chorus-like effect.

Unison is also particularly good for use with filtered pads as well as beastly, bone-crunching leads. If your synth allows you to set the amount of unison try decreasing it to keep it from becoming too overbearing. Except of course when you want it to sound beastly!

Ring Modulation

Even though the synthesizer provided with this book and the included patches do not use ring modulation it is nonetheless common enough that it deserves some explanation. Ring modulation works by multiplying two signals together which produces an output that consists of frequencies that are the sum and difference of all the original harmonic frequencies. For example take two harmonics which each belong to separate signal inputs on a ring modulator. If one has a frequency of 50 hz and the other has a frequency of 150 hz then the signal put out by the ring modulator will have harmonics at 100 hz and 200 hz. This particular example illustrates the process but it may not seem to be particularly useful since we are just taking two harmonics and producing two *different* harmonics. The important point to consider is that in practical usage *every* harmonic in one signal gets multiplied with *every* harmonic in the other. If the two signals have tunings where many of their harmonics coincide then the end result will be a harmonic sound with rich overtones.

If the oscillators are set to inharmonic intervals the ring modulator will produce a rich inharmonic sound that is harsh and metallic and is quite similar to inharmonic sounds produced with FM (frequency modulation). Ring modulation was used quite a bit in old sci-fi flicks and early avant garde electronic music. As a point of interest, the Dalek voices on Dr. Who were originally created with a ring modulator by multiplying a 30 Hz sine wave with mid-range boosted speech input.

FILTERS

Filter Types

The incorporation of filters was the fist great revolution in early electronic music after the oscillator. There are only a few distinct sounds that can be created using unfiltered waveforms. Synthesizers were born with the introduction of the filter. Of all the components on a synthesizer it is the most responsible for shaping the sound. Filters are used to change the timbre or tone color of these basic waveforms.

The most common type is the low-pass filter. A low-pass filter works by removing or attenuating high frequencies and passing low frequencies. A signal that has been subjected to low-pass filtering will not sound as bright. This type of filter is the most common because our ears are accustomed to hearing sounds with attenuated high frequencies. As sound travels through air it not only loses amplitude overall but the higher frequencies also diminish more than the low frequencies. Bass frequencies travel further than high frequencies. Higher frequencies also experience more absorption by instrument bodies and obstructions. Figure 21 shows white noise that has been passed through a low-pass filter.
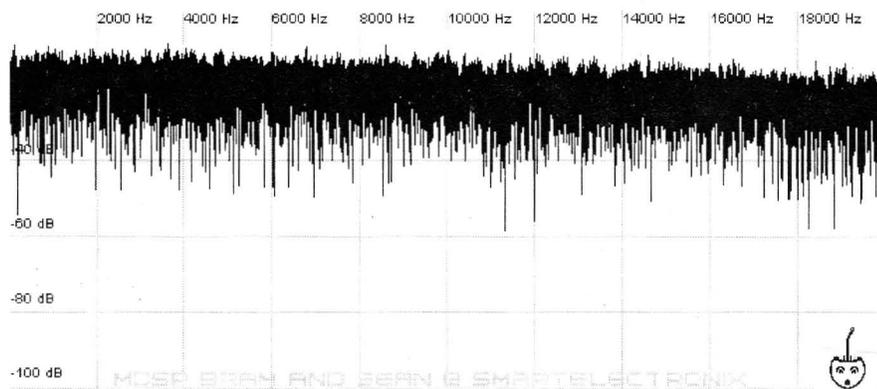
Figure 20. Unfiltered white noise
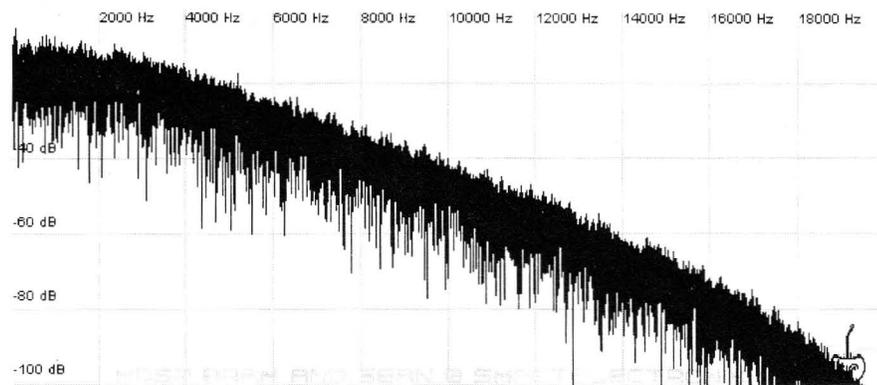
Figure 21. White noise routed through a low-pass filter. Higher frequencies have been attenuated while lower frequencies pass through.

Another common filter type is the high-pass filter. This filter as the name implies works by attenuating low frequencies and passing the high ones. Some natural sounds have diminished or outright lack lower-pitched harmonics. Isn't this impossible given that the pitch of a note is determined by the frequency of its lowest harmonic the fundamental? Remember that the frequencies of overtones are related to the frequency of the fundamental. If the fundamental and some of the lower overtones are missing our brains are able to relate the frequencies of the existing higher overtones back to the missing fundamental and perceive the correct pitch. This type of filter can be useful for recreating cymbals, handclaps and other instruments that sometimes lack lower frequencies.
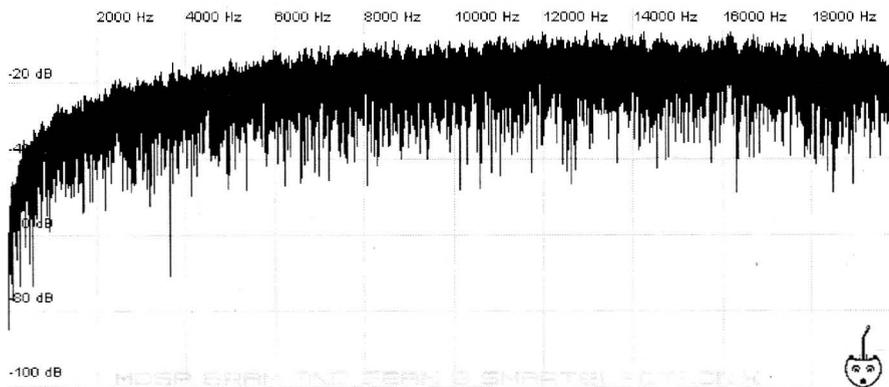
Figure 22. White noise routed through a high-pass filter. Lower frequencies are attenuated while higher frequencies pass through.

Two other filter types that are related to one another are the band-pass and band-reject filter. A bandpass filter attenuates both high and low frequencies and only lets a narrow *band* of frequencies through.
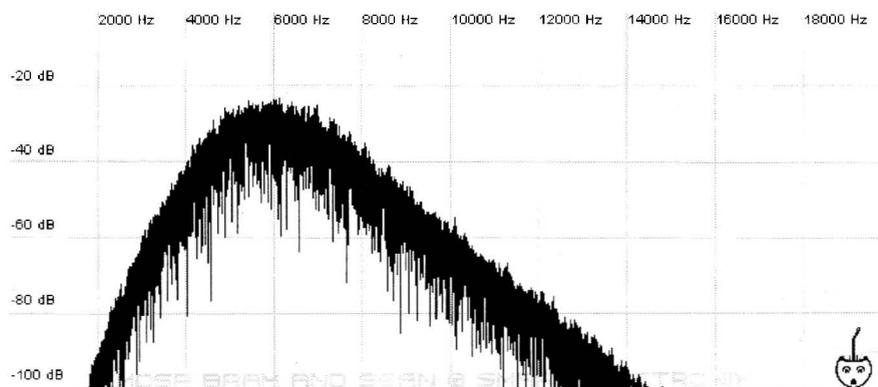
Figure 23. A band-pass filter only allows frequencies within a narrow band to pass.

A band-reject filter, or notch filter, is essentially the inverse of the bandpass filter. It lets high and low frequencies pass and attenuates frequencies within a narrow band.
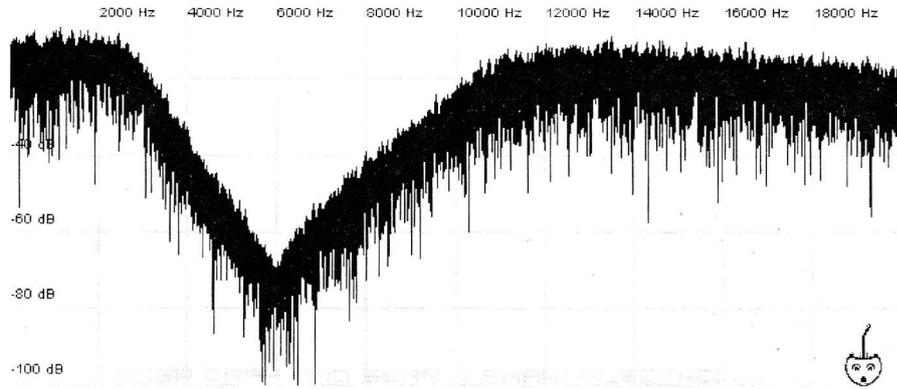
Figure 24. A band reject filter removes frequencies within a narrow band but allows all others to pass.

Cutoff Frequency

The cutoff frequency sets the frequency point where filtering begins to take place. If for example a low-pass filter has its cutoff frequency set to 6 khz then the output will pass all frequencies below 6 khz and all higher frequencies will be diminished. In actuality some filtering actually takes place at frequencies before the cutoff frequency. To be a bit more strict with our definition the cutoff frequency is usually taken to be the point at which the frequencies have been attenuated by -3 db, but this can be slightly different from one instrument to another.
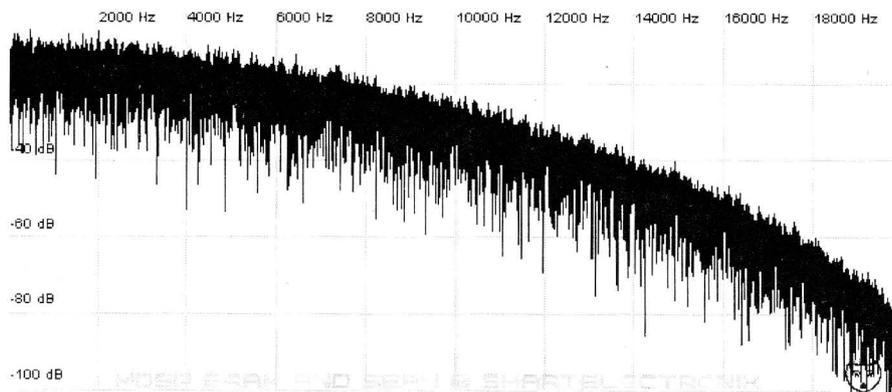
Figure 25. Here cutoff frequency has been set to 6 kHz (6000 Hz). Notice that frequencies less than 6 kHz are also filtered but to a lesser extent than those above 6 kHz

Filter Slope

A common misconception with filters is that *all* frequencies beyond the cutoff frequency are removed from the signal. The filtering beyond the cutoff frequency actually takes place gradually. The rate or degree of filtering that takes place beyond the cutoff frequency is determined by the slope of the filter. The most common types are 12 db

and 24 db per octave slopes and occasionally 6 db, 36 db and other values are used. A larger decibel value means more attenuation. A 12 db low-pass filter will sound slightly brighter than a 24 db low-pass when set to the same cutoff frequency. What these values mean is that for every octave the frequencies will be attenuated by the given decibel amount. For example let's say we have a 24 db low-pass filter and the cutoff frequency is set to some fairly low value like 1.5 kHz. Let's compare the filtering at 4 kHz and 8 kHz since these values are both in the filtered region. Frequencies at 8 kHz will be 24 db "quieter" than those at 4 kHz since these frequencies are one octave apart.
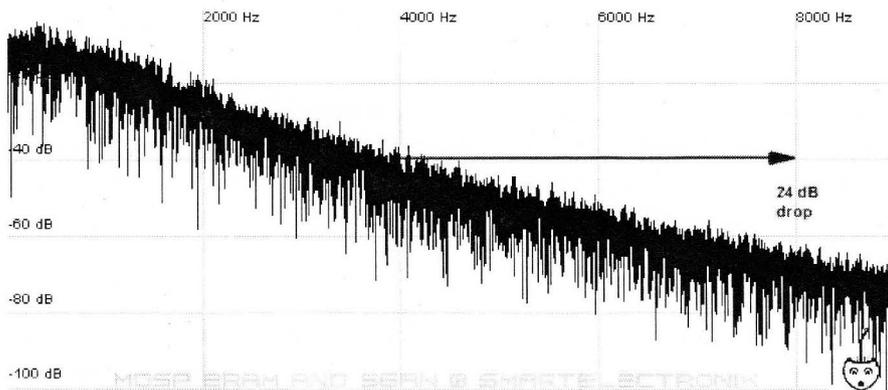
Figure 26. Filter response of a 24 dB per octave low-pass filter. Material at 8 kHz is attenuated 24 dB more than material at 4 kHz since these frequencies are one octave apart

Filters are also referred to by their number of "poles." One pole equals 6 dB of filtering. A two-pole filter is the same thing as a 12 db filter. A four-pole filter is the same thing as a 24 db filter.
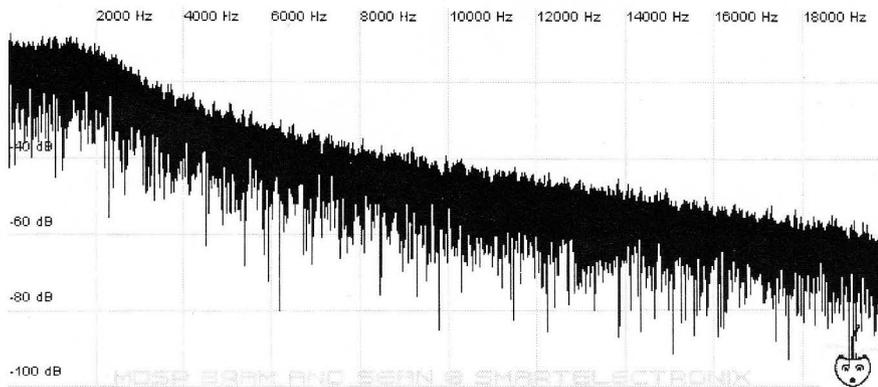
Figure 27. Filter slope of a 2-pole, 12 dB per octave filter
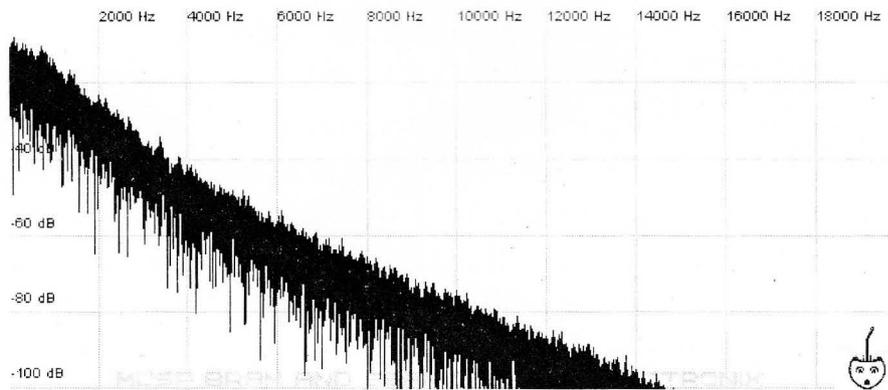
Figure 28. Filter slope of a 4-pole, 24 dB per octave filter

Some synthesizers allow the user to select various filter types. No type is inherently better than another and the most preferable type depends upon the application.

Resonance

The function of the resonance setting on a filter depends upon the filter type. Its purpose is completely different in low and high pass filters than it is in band-pass and band-reject filters. When used with either a low-pass or a high-pass filter, resonance amplifies those frequencies closest to the cutoff frequency. When used with band-pass and band-reject filters the resonance setting acts as a "*Q*" control and sets the width of the band.
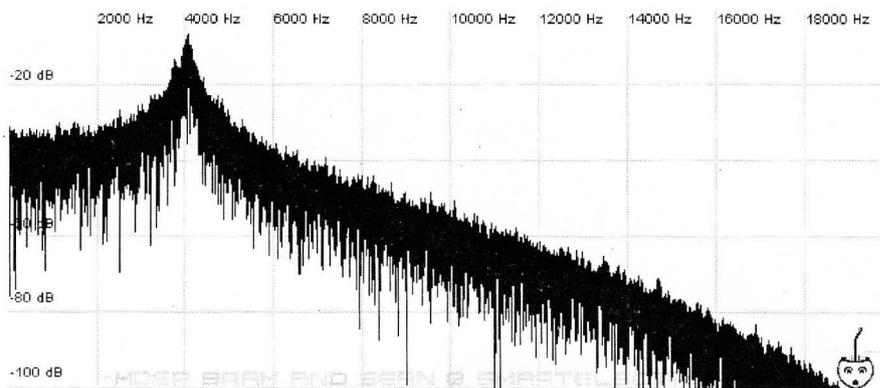
Figure 29. Filter resonance at a 4 kHz cutoff frequency

It should be made clear that resonance on a synthesizer is not the equivalent of resonance of an acoustic instrument nor is it *necessarily* used to replicate the resonance of an acoustic instrument. In situations where a particular frequency or set of nearby frequencies need to be amplified resonance becomes a useful tool.

Most synthesizers have gain compensation which will automatically lower the overall output of the signal as resonance is increased. This helps to prevent the resonant spike from going through the roof but it comes at the expense of lowering the rest of the signal.
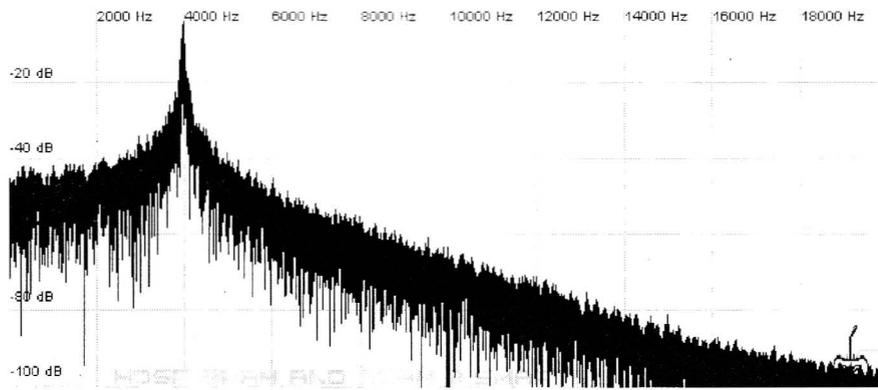
**Figure 30. Increasing the resonance results in gain compensation
automatically lowering the amplitude of surrounding frequencies**

Some filters are self-resonating meaning that when the resonance is turned up high the
filter will resonate even if no signal is input into the filter. This effectively turns the filter
into a sine-wave oscillator. If keyboard tracking is available on the filter then this sine
wave can be played as if it were a waveform from one of the oscillators. In cases where a
synth does not have a self-resonating filter it is still possible to "play" the filter's
resonance with keyboard tracking by feeding a noise source into the filter and using this
to drive the filter's resonance.

Another way to play a resonant filter directly is through a method employed in acid
electronica. The acid genre got its name not as a drug reference but rather from what are
known as acid lines or acid tracks. Acid lines in turn got the name due to the way they
sound. The acid line is one of those great stories in electronic music where a
technology, in this case the resonant filter, is used in a fashion not originally intended.
More specifically it was born out of the Roland TB-303 in the mid 80s. The "303" was
originally marketed as a bass accompaniment for guitarists. It is essentially a sequencer
that plays a single oscillator and was never intended to be played in a realtime or
improvisational sense. The unexpected thing that happened was that electronic artists
learned to use the TB-303 as a realtime instrument by turning up the resonance and
alternating the filter cutoff frequency manually in realtime thus changing the pitch of the
resonance while the unit's onboard sequencer cycled thorough note patterns.

Here's an example of how to create a very basic acid line without having to sell vital
organs to buy a TB-303. Program a sequencer to loop through a rhythm that consists of
only one note. Most acid tracks don't use just one note but we'll do it here just to better
illustrate the process. The synth's oscillators can be set to any waveform but I'd suggest
using a sawtooth. Turn the resonance up to about 3/4 of its full position. Now as the
sequencer plays the synth alternate the filter cutoff frequency by hand. This will cause
the pitch of the resonance to change. You may have to confine the extent over which you
twiddle the cutoff frequency to the lower part of its range as higher frequencies may be
inaudible. Twiddle the cutoff frequency knob in time to the rhythm and there you have
an acid line albeit a simple one.

ENVELOPES

Envelopes are used to control how various parameters change over time once a key is played. Envelopes can be used to control just about any parameter, but the most obvious use of an envelope is to control the way a patch's amplitude changes. If we want a patch to start loud and die out slowly like a cymbal or start out quiet and slowly become louder like a violin an envelope could be used to control the amplitude.

The most common type is the ADSR envelope which breaks the time over which a key is struck and held into four distinct periods called stages. ADSRs are hence known as four-stage envelopes. "A" stands for *attack* and it controls the rate of the envelope's initial increase. "D" is for *decay* and it controls the rate at which the envelope decreases after the attack. "S" is for *sustain* and this controls the level at which the envelope will remain until the key is released. "R" is for release and it controls the amount of time that it takes for the envelope to die out after the key is released. Attack, decay, and release all control lengths of time and settings are commonly given in seconds and milliseconds. Sustain on the other hand sets the level after decay has completed and has no set time associated with it. It begins as the decay ends and will last indefinitely until the key is released. It's sometimes given as a percentage where 100% would be the maximum setting.
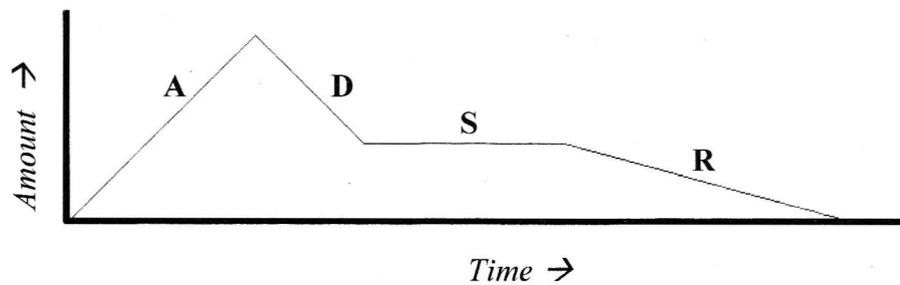
Figure 31. Representation of an ADSR envelope. ADSR envelopes can be applied to many different parameters but are most commonly used to control amplitude and filter cutoff frequency.

The following graph shows the effect that an ADSR envelope can have on amplitude.
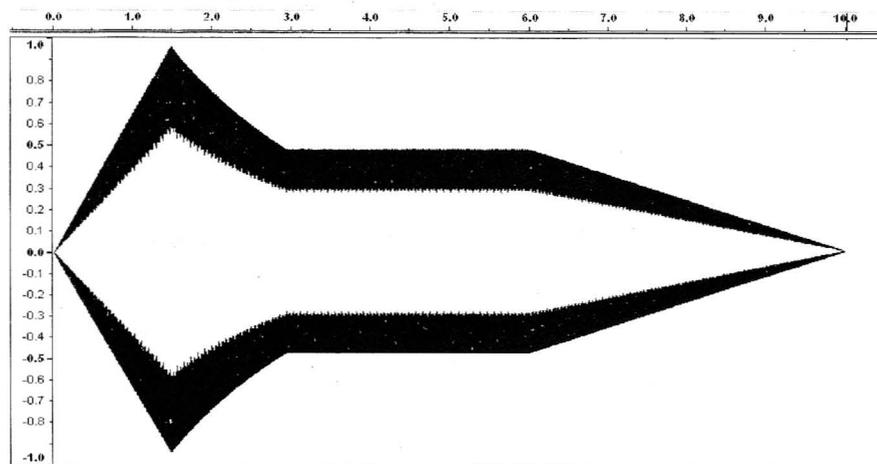
Figure 32. The results of an ADSR envelope controlling amplitude as viewed in the sound editor Audacity

Incidentally, if sustain is set at 100% then decay really serves no purpose as it will not "decay" down. There are also variations on the ADSR configuration. The original MiniMoog used an ADS envelope where the "Decay" setting also doubled as a release setting. The ARP Odyssey and ARP 2600 had AR envelopes as well as the conventional ADSR type. There are also envelope types that use more than four stages but the ADSR is the most common.

Another use for envelopes is to control the filter's cutoff frequency. If you listen closely to woodwinds or brass instruments you may notice that as the player begins to blow into the instrument that not only does the amplitude increase but the sound also becomes brighter. As pressure builds in the instrument higher frequencies are added to the sound. These higher frequencies are also the first to die out when the player stops blowing. This behavior can be replicated by controlling the filter's cutoff frequency with an envelope.

Here's a piece of advice to give a patch more character and realism. Use the filter's envelope to create changes in the amplitude and *then* use the amplitude envelope to make any remaining adjustments. Huh? You may be asking why and how the filter envelope can be used to control amplitude. Isn't that what the amplitude envelope is for? Well, it's simple really. Whenever frequencies are removed from a signal they no longer contribute to the overall power of the signal. Removing frequencies will result in an overall lower amplitude. A filtered signal isn't as loud as an unfiltered signal. Because of this a filter envelope has the effect of behaving somewhat like an amplitude envelope. This approach can be restated as the following. First use the filter envelope to control the amplitude of the higher harmonics and after that use the amplitude envelope to control the amplitude of the remaining lower harmonics that were untouched by the filter.

The filter envelope has an *amount* setting that is used to determine how much the envelope influences the cutoff frequency. Sometimes it can be a bit tricky deciding whether to adjust the cutoff frequency or the envelope amount. In these situations the best suggestion is to experiment by adjusting both.

To compare the amplitude and filter envelopes try the following. Set the oscillators to produce any waveform that you like. With the filter completely open (i.e. no filtering) raise the attack amount up on the amplitude envelope. You should hear the sound slowly get louder as you hold down a key. Now turn the amplitude attack back down and this time lower the filter cutoff frequency, increase the filter envelope amount, and increase the filter envelope attack time. As you hold down a key notice that the sound gets louder just as before but this time the timbre or tonal quality of the sound also changes.

LFOs

LFO stands for *low-frequency oscillator*. Unlike a synth's main oscillators an LFO is not heard at the output or at least not directly. It is a modulation source meaning that it is used to create changes in other parameters. LFOs are used to create cyclic fluctuations in pitch, amplitude and a variety of other parameters. To add vibrato or tremolo to a patch an LFO would be used to modulate the pitch and amplitude respectively. Another key difference is that the main oscillators produce waveforms with frequencies in the audible range of 20 Hz – 20 kHz while LFO frequencies are typically less than 20 Hz. If an LFO could be connected directly to the output it would be inaudible because of its low frequency.

LFOs are similar to the main oscillators in that they do use square, triangle, and sawtooth waves and they also commonly use sine waves and sample & hold patterns. These are all referred to as source waveforms. Sine and triangle waves are the most ideal for imitating natural effects such as vibrato and tremolo. The results they produce, when used with an LFO at least, are very similar and for this reason it is not uncommon that a synth provides one and not the other.

LFOs have a frequency or rate setting that changes the speed of the cycle. Vibrato, tremolo typically use LFO frequencies under 5 Hz. Higher frequencies can be useful for many synthetic sounds and rhythm effects. The degree to which the LFO modulates a given parameter is controlled by an *amount* setting. Amount settings may either be found in the LFO section itself and sometimes they are located in each section to which they control.

Let's look at what happens when an LFO uses a sine wave to modulate the amplitude of a patch. Here is a waveform of a patch without any LFO modulation.
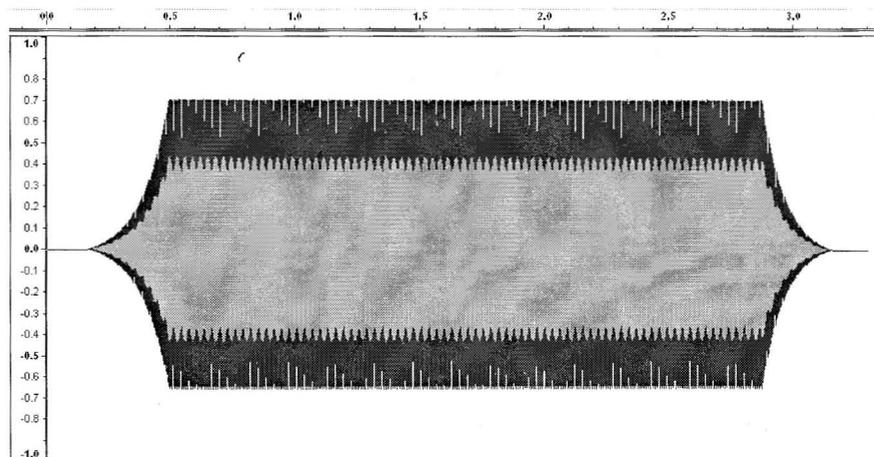
Figure 33. A patch without any LFO modulation

Following is the same patch played with the amplitude modulated by the LFO set to use a sine wave at 4.5 Hz. This creates vibrato.
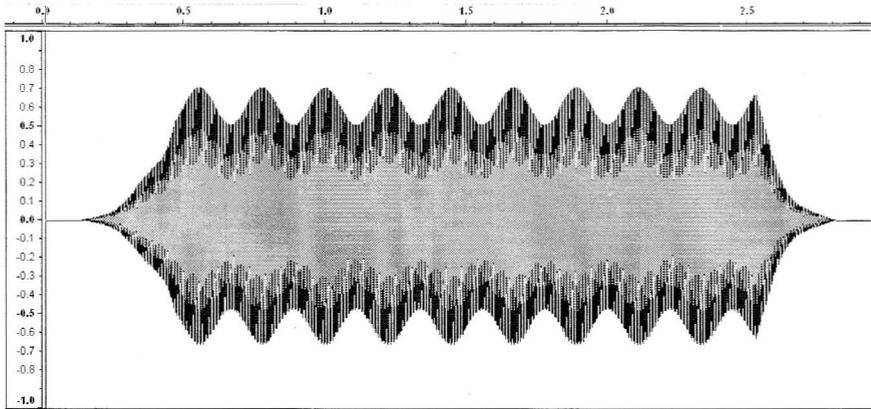
Figure 34. Patch with LFO modulation using a sine wave source set to a frequency of 4.5 Hz

In the next graph the LFO has the same frequency but uses a triangle wave.
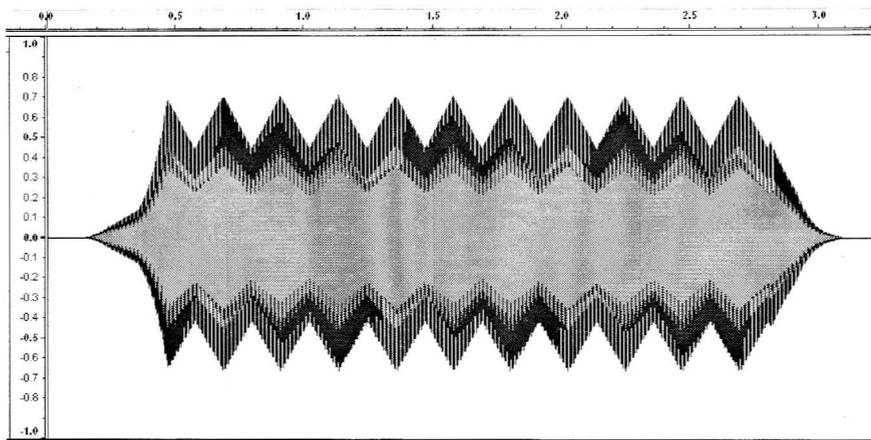
Figure 35. Patch with LFO modulation using a triangle wave source set to a frequency of 4.5 Hz

The sine wave source and the triangle source produce very dissimilar graphs but if you set your synth up to modulate amplitude and compare the use of a sine versus a triangle you will hear very little difference. When setting up the frequency of an LFO remember that 1 Hz = 1 cycle per second. In the above example the frequency was set to 4.5 Hz thus producing 4.5 cycles each second as can be seen by looking at the graphs.

Pulse-width modulation

One of the coolest uses of the LFO is to create pulse-width modulation. Recall that changing the width of a pulse changes its harmonic content. Routing an LFO, using a sine or triangle source, to modulate the pulse width creates an extremely rich sound that could best be described as a thick, string section-like sound.

Below is a graph showing the harmonics of a pulse-width modulated waveform. Notice that the higher harmonics are broadened. This is very similar to the effect heard by chorusing or unison. Keep in mind though that pulse-width modulation tends to even out the levels of the harmonics. For instance a pulse that is set to a width of 25% will have harmonics that appear to undulate in amplitude. If a heavy dose of modulation is applied to the pulse width this undulation will disappear and the harmonic amplitudes will take on characteristics similar to that of a sawtooth that has some chorusing or unison.
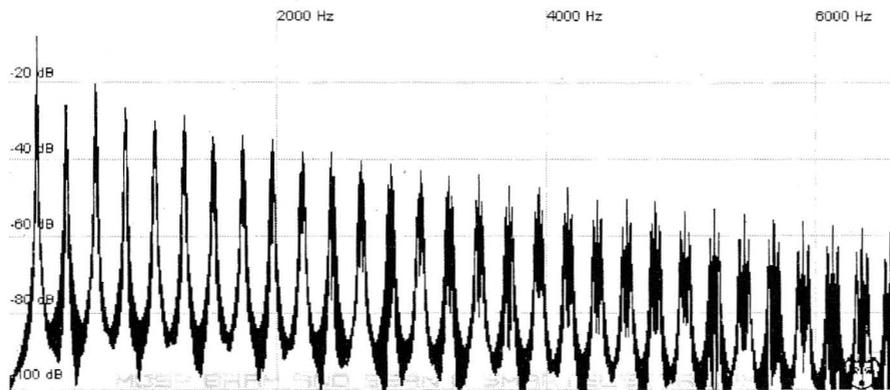
Figure 36. Pulse-width modulation causes the upper harmonics to broaden resulting is a chorusing effect. This is very useful for string and pad sounds

The trick to getting good PWM string sounds is to use pulse-width modulation on both oscillators and detune one about 10 cents high and the other about 10 cents low. Set the LFO frequency a bit under 5 Hz and the amount to a bit under 50% which will make it thick while minimizing the beating effect that is characteristic of PWM. Then apply some light chorus and delay if available.

Beat effects (gating and note switching)

Because of their cyclic nature, LFOs can also be used to create rhythm. This is where the square and sawtooth sources become the most useful. Using either one of these sources and routing the LFO to modulate the amplitude or filter cutoff frequency can result in useful gating effects. Using a square wave source will cause the output to have a repeating On/Off pulsing pattern. A sawtooth source will cause the sound to cycle through a "ramp up" or "ramp down" depending on the direction of the sawtooth. These effects are particularly useful when the LFO frequency is matched to the beat of the song. A common attribute of LFOs on many modern synthesizers is the ability to sync to a time signature provided either internally or by an external sequencer.

By routing a square wave source LFO to modulate pitch it is possible to create note switching, trill sounds. The key here is to set the LFO depth to something that will produce a musically useful interval such as a fifth (7 semitone difference) or an octave (12 semitone difference). This can be a useful way to create a simple two-note arpeggio/trill effect that can be useful particularly if your synth lacks an arpeggiator.

Sample & hold

Without getting into a deep explanation sample & hold is a random pattern that is often available as an LFO source. It's great for creating the computer-type sounds from old sci-fi movies. Set the LFO source up to be sample & hold (or noise) and route it to modulate the oscillator pitch. This should produce a cheesy computer sound. Also try routing the LFO to the filter cutoff frequency and turn the filter resonance up. This produces a different but equally as cheesy computer sound!

Sync sweeping

Here's a great way to use an LFO to create a screaming, growling lead. Set the slave oscillator to sync to the master oscillator. Keep the master oscillator turned on but keep its volume all the way down. Tune the slave oscillator higher than the master's pitch. Set the LFO to use a sine or triangle at a really, really low frequency (<1 Hz) and use this to modulate the slave's pitch. Apply portamento(glide) and lay on some unison and chorusing. Experiment with different LFO frequencies and depths. As you increase the LFO amount decrease the LFO frequency.

Keyboard Expression

Velocity Sensitivity

If a piano's keys are struck hard it will obviously sound much louder than if they are struck lightly. Not only that but the timbre of the sound will be much different as well. A hard-struck piano doesn't simply sound like a louder version of a lightly-struck piano. Both the amplitude *and* harmonic character of the sound depend on how hard the keys are played. This type of behavior can be replicated on synthesizers that have velocity sensitive keyboards. A velocity sensitive keyboard can be used to modify parameter amounts based on how fast the keys are pressed. The two most common uses for velocity sensitivity are to change the amplitude level and the filter-cutoff frequency, but it is also sometimes used to modify pitch, envelope times, LFO rates and amounts, and just about anything else. Velocity sensitive keyboards allow for much greater expression and realism. These days most keyboards have velocity sensitivity but not all do particularly inexpensive ones and even pro models produced before the mid 80s. Be sure to check before buying that next synth.

Aftertouch (pressure sensitivity)

A keyboard with aftertouch or pressure sensitivity allows the user to control parameter amounts based on how much pressure is applied to the keys as they are being held down. This can be quite useful when used with the filter to imitate wind instruments being overblown or routed to pitch for replicating string bending on a violin. Like velocity sensitivity it to can be applied to a wide range of parameters. Aftertouch keyboards come

in two different flavors. The cheapest to implement and therefore the most common is channel or global aftertouch. A keyboard with channel aftertouch averages the pressure from all the depressed keys to create a single aftertouch message. Polyphonic aftertouch which is much more rare allows each key to transmit its own aftertouch message. While it has become more common, most keyboards do *not* have aftertouch so if this is something that interests you be sure to look for it.

Synthesis Through Harmonic Analysis and Reverse Engineering

The use of harmonic analysis in synthesis is nothing new, yet it has not been used much outside of academic research and professional patch programming. There are a few reasons for this. The first is that harmonic analysis requires a fair amount of processing power that until recently was too expensive for most of us. Fortunately home computers are now powerful enough to perform the analysis in real-time while using only a small percentage of the computer's processor. The second reason is that it is generally perceived as difficult to use. There is quite a bit of mathematical theory that goes into computing the Fast-Fourier Transforms (FFT) on which harmonic analysis is based but as far as using harmonic analysis as a tool for synthesis it's rather straightforward. The third reason, somewhat as a result of the first two, is that it simply hasn't become part of convention.

This is super powerful stuff and it has the potential to change the way you program sound. Even though this section is centered on analyzing harmonics and reverse engineering musical content it should be noted that it will also improve your ability to create original patches as well. After you use it for a while and you come to understand the spectral makeup of different sounds you will rely less and less on having to look at harmonic diagrams. They'll be in your head. This book is about analog synth programming but it should be mentioned that harmonic analysis also makes FM synth programming infinitely easier to understand. Don't even bother programming FM without it! I digress...

Before proceeding I recommend that you first get a hold of software that can perform harmonic analysis/FFT in realtime and play around with it for a bit. Most of the more sophisticated sound editors provide the ability to display harmonic diagrams of sound files albeit usually not in real-time. For real-time analysis I highly recommend the Fre(a)koscope VST plug-in. This plugin was used to create all of the harmonic diagrams in this book. It has an excellent interface, is easy to use, and it's free! If you don't have a host that supports VST plugins be sure to visit kvraudio.com where you can find a number of hosts available for free. If you have any questions please email me at synthcookbook@yahoo.com

The analysis/synthesis process can be broken down into three stages for each sound.
1. The most important step. Obtain a harmonic diagram of the sound that is to be emulated or recreated. Pick the best oscillator waveforms, pitches, and amplitudes to match the diagram.
2. High frequency harmonics are generally the last to rise in amplitude and are the first to die back out. While comparing with the harmonic diagram of the original source use the synth's filter envelope to match this behavior.
3. Using a sound editor, match the synth's envelopes to the original source.

Step one is essential while steps 2 and 3 are not always possible or required. Envelopes may not require any visual analysis since they can be duplicated by ear with good results more easily than harmonics. Note: The lowest-pitched harmonics are the most important. As a general rule of thumb try to get the first 5-10 harmonics as close to their corresponding levels in the original source.

Reverse engineering a patch from another synth:

The easiest sounds to recreate are those from other analog synths. Because analogs have very similar feature sets it makes for easier translation. About ten years ago I had to sell a Sequential Circuits Six-Trak but I made sure to record samples of some of my favorite patches that I'd programmed for it. My favorite was a lead with thick unison and we'll begin by reverse engineering this patch. The sample of this patch can be found on the CD.

Playing the sample through Fre(a)koscope produces the following harmonic diagram,
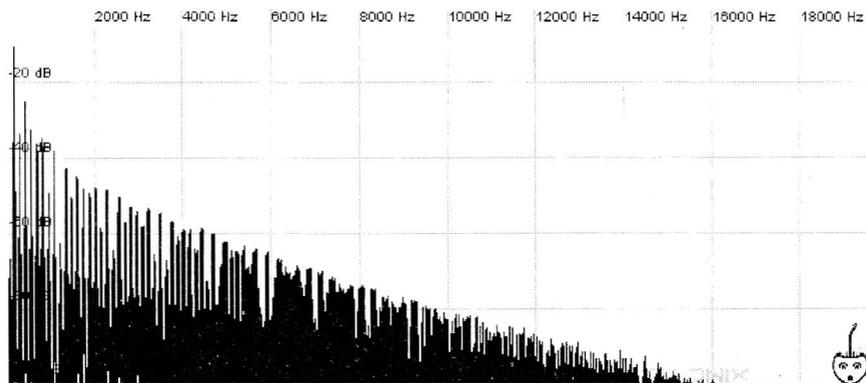
Figure 37. Harmonics of a patch originally programmed on a Sequential Circuits Six-Trak

Here Fre(a)koscope has been set to a linear display and a maximum window size with a frequency range of 20 Hz – 20 kHz. The lower harmonics are the most important so let's decrease the maximum frequency of the display so that we can see these in greater detail. Lower the maximum range down to about 6 kHz.
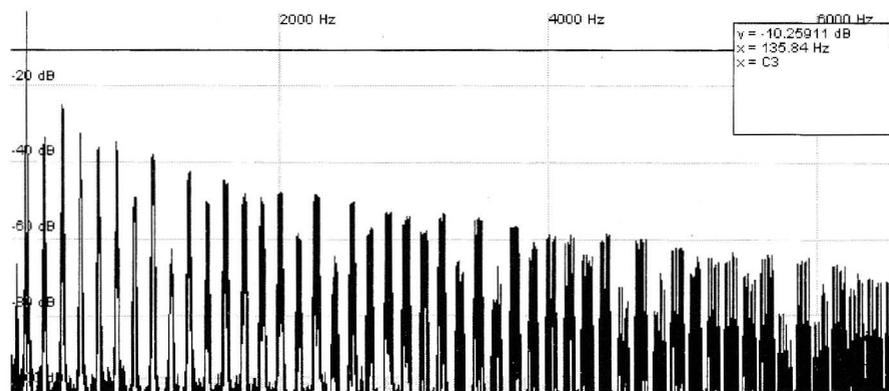
Figure 38. Six-Trak patch zoomed in to show finer detail in the lower-frequency harmonics. The pitch of the sampled note is at C3 on the keyboard

In Fre(a)koscope it is possible to get decibel value, frequency, and equivalent key number by clicking on a section of the graph. In the graph above the fundamental has been selected and from the inset box at the upper right we see that the sample is tuned to the C3 key. We need to know this so that we know what key to play on the synth that is

being programmed. There are many important details in the above graph that give us a clue as to where we should start. Notice that both even and odd harmonics are present. This rules out using a single triangle wave or square wave since they have only odd harmonics.

The two remaining options are sawtooths and pulses since they possess both even and odd harmonics. Also notice that the higher harmonics become broad and are not the narrow spikes that we would expect with pure sawtooths or pulses. This indicates that there could be some pulse-width modulation taking place or the oscillators have been set to unison. We will get back to the broad harmonics in a bit but for now look at the way the harmonic amplitudes do not decrease evenly from left to right. A sawtooth that has been synced can produce harmonics that look like this but so can a pulse with variable width. Let's take a look at the synced sawtooth option first. Set your synth up with the slave oscillator synced to the master oscillator and have the slave produce a sawtooth wave while at the same time the master's output is turned all the way down. Increase the pitch of the slave while keeping an eye on the synth's harmonic analysis. The closest that we can get to the original harmonics this way is when the slave oscillator is tuned about 8 semitones above the master.
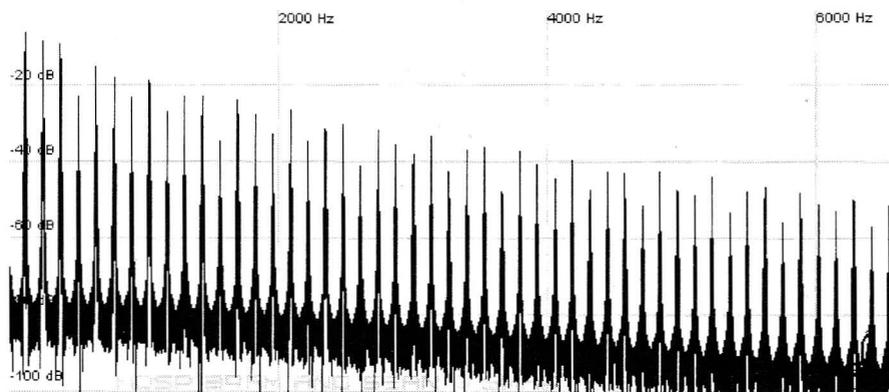
Figure 39. A synced sawtooth oscillator tuned 8 semitones above the master produces results similar to the original patch but not close enough

What are the notable differences between this graph and that of the original Six-Trak sample? Of course the harmonics are all narrow and again we will get to that in a moment but for now we only need to compare their amplitudes to the original patch. The higher harmonics in our patch are slightly louder than those in the original. Notice that they are higher on the graph. This can be remedied by applying some low-pass filtering but there is a bigger discrepancy than this. Notice that the harmonics of our patch do not undulate up and down in quite the same fashion as the original patch. It would however be a good guess that if we find out what is broadening the higher harmonics of the original patch and we apply it to our patch as it is right now that the two would nonetheless sound very similar but we are going to try to do better. Remember this was originally produced by another synthesizer so we should be able to get really close and we can probably even nail it.

REVERSE ENGINEERING A PATCH

Let's see what we can come up with using a pulse instead of a sawtooth. Again only use one oscillator but this time do not sync it to the other. Set it up to produce a pulse. Begin by setting the pulse width to produce a square wave. Remember that a square wave should only show odd harmonics. Now very slowly begin decreasing the pulse width. Before the width has been decreased much at all you should get harmonics that are identical in amplitude to those of the original patch. This happens to be at a pulse width of about 44%, something not much narrower than a square wave but harmonically different. Now we haven't yet applied any filtering so the harmonics will not diminish quite as rapidly as those in the original patch but the levels of each harmonic relative to their neighbors follows the same pattern as those in the original.
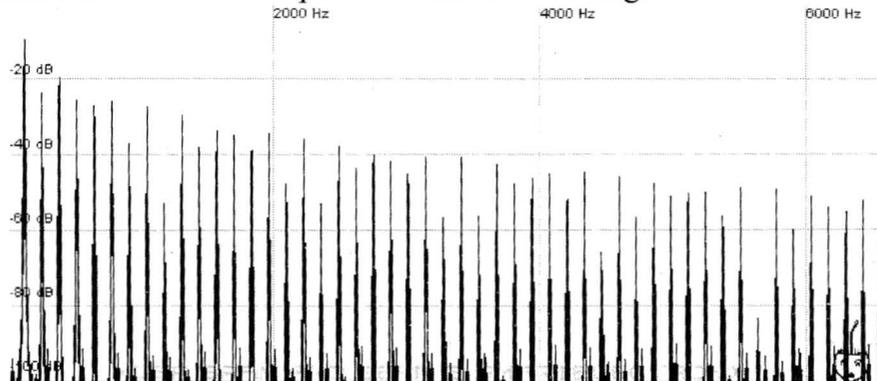
Figure 40. A pulse width of 44% produces harmonics that are very similar to the original patch

Next experiment with some low-pass filtering until the level of the higher harmonics matches those of the original. A 24 dB filter set to a cutoff frequency of 2.5 kHz produces the following graph.
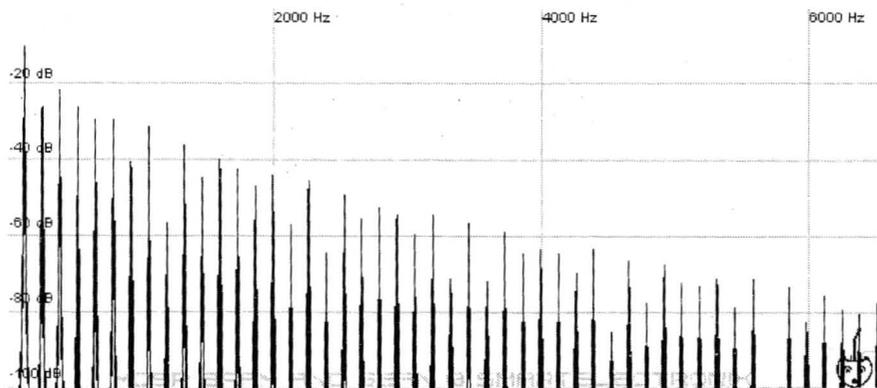
Figure 41. Apply some low-pass filtering to match the harmonic amplitudes to those in the Six-Trak patch

There are some differences between this and the original but the amplitudes are pretty darn close. While two synths can produce *practically* the same results they cannot produce *exactly* the same results.

The one last thing that we need to take care of is determining what is causing the higher harmonics to broaden. This could either be due to pulse-width modulation or oscillator

unison. Try applying some pulse-width modulation. As the pulse-width modulation amount is increased it produces the following results.
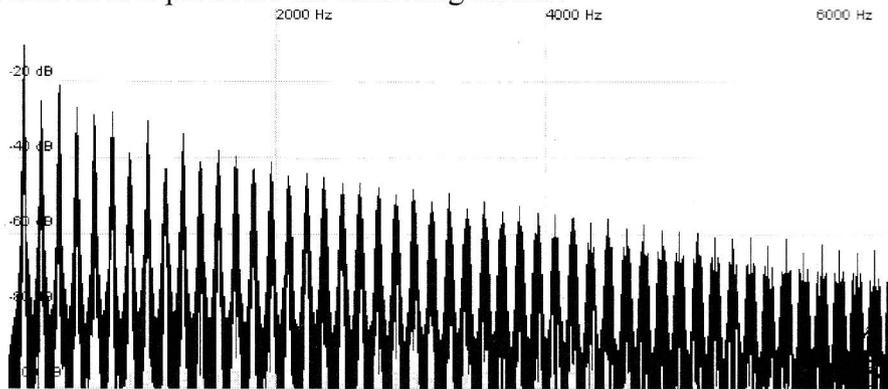
Figure 42. Pulse-width modulation broadens the higher harmonics but it also flattens out the harmonic amplitudes

This has broadened the higher harmonics as we had hoped but it has also managed to decrease the amount of undulation in the harmonics. Pulse-width modulation obviously wasn't used in the original patch.

Get rid of the pulse-width modulation and this time turn on unison. The Six-Trak had six oscillators so set your unison mode to six oscillators if applicable. If you're using the SCB synth turn unison mix and intensity all the way up and put speed at about 50%. You should get harmonics similar to those below.
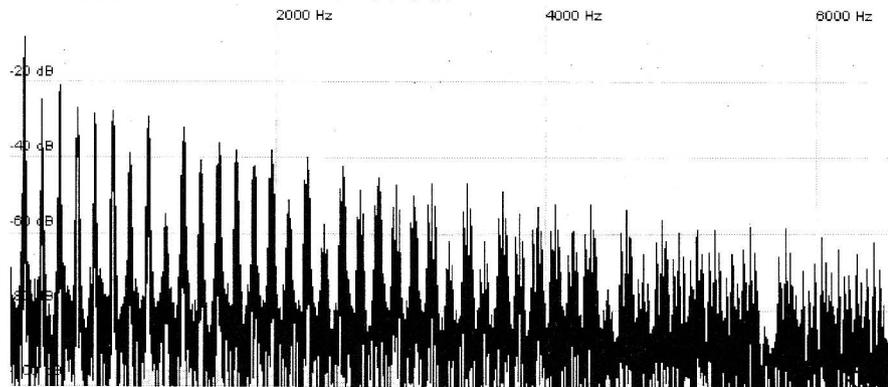
Figure 43. The harmonics of the resynthesized patch in its final form

This is very similar to the original
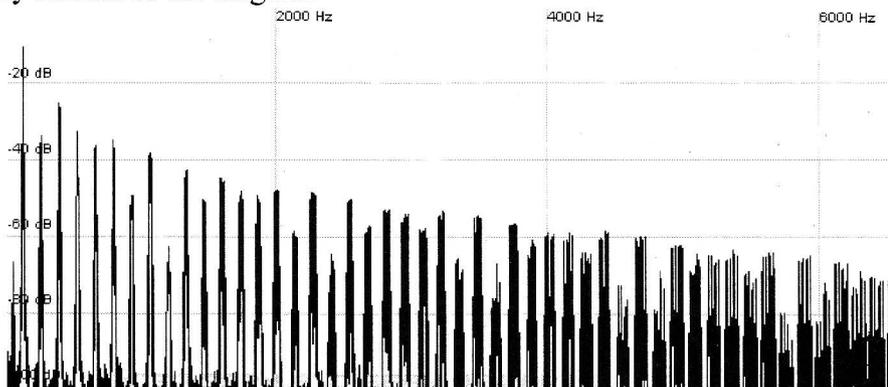
Figure 44. The harmonics of the original patch

Listen to the resynthesized patch and compare it to the original sample. They are practically indistinguishable. This also sounds far better than trying to stretch a 4 second sample across the whole keyboard using a sampler! If we wanted to do something like decrease the amount of filtering we can do that with the resynthesized patch but would not be able to using the audio sample.

Reverse engineering a sound from a song:

The tricky part to reverse engineering a sound that's in a song is the presence of other interfering material. Here the goal is to find a clip of the sound that is as isolated as possible from other competing sounds in the mix.

For an example of reverse engineering a sound from a song we'll lift one off of Shape 1 from the Shape album which can be found on the CD. There's a cool bass that starts at about 2:45 into the song. Use a waveform editor to take a look at the region from 2:45 to 2:50. An audio clip of this section can also be found on the CD. For waveform editing, I use Audacity which is an excellent freeware sound editor.
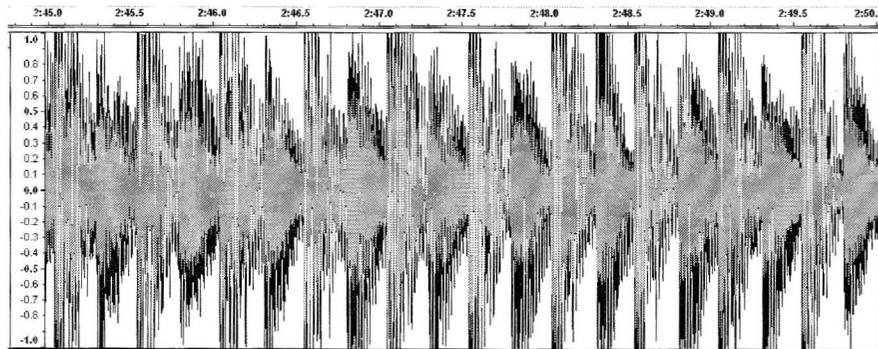
Figure 45. Audio clip containing the bass sound that is to be duplicated as well as a bass drum and even the tail end of a pad. The bass sound must be isolated from a section of this recording

In this section there are three different sounds. There is the bass sound that we would like to isolate for resynthesis, a bass drum, and at the beginning of the section there is the tail end of the pad-type sound from earlier in the song. The bass and the bass drum both play ten times in the clip shown above. The bass is the quieter of the two. Because of the interference of the pad at the beginning of the clip we will isolate one of the bass voicings found after 2:47. Let's take the one from region 2:47.79 to 2:48.04 . Yes that's right, we are going to recreate the patch from a mere ¼ second of sound!
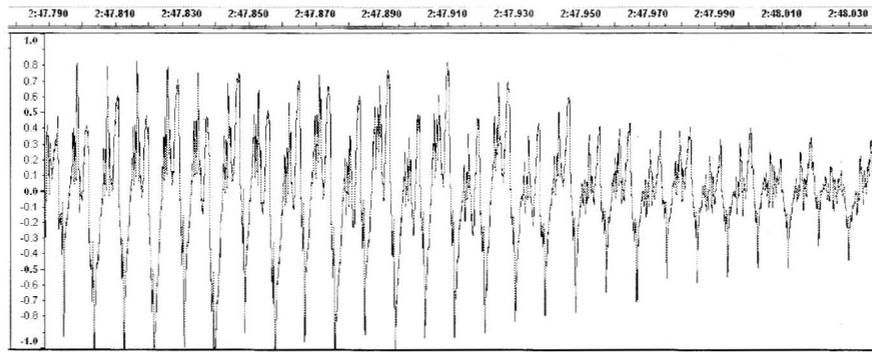
Figure 46. Audio clip of the bass sound isolated from the region 2:47.79 - 2:48:04

Running this clip through a harmonic analyzer set to a linear view produces the following harmonic diagram. The range has been set to show the lowest harmonics.
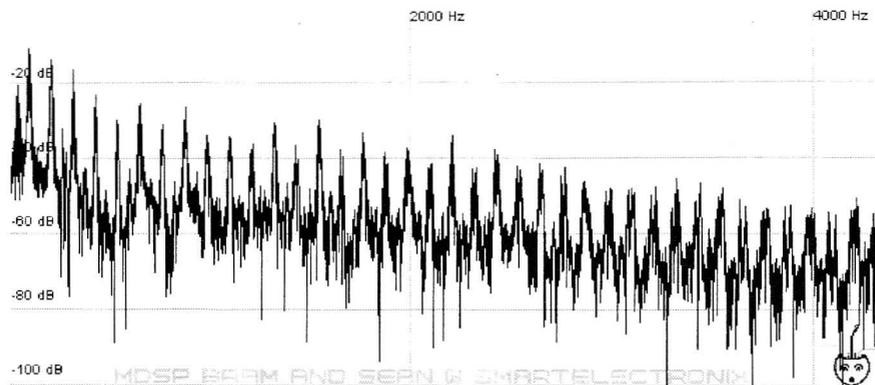
**Figure 47. Harmonics of the isolated bass sound. The high noise floor is
the left overs of other sounds and effects in the mix**

Notice that the noise floor isn't very far below the tops of the harmonics. This is most likely the left over garbage from other sounds and effects in the mix. Look at the far left of the diagram and at what appears to be the first harmonic. The dark nub just below the -20 dB line. This too is most likely an artifact since its frequency doesn't relate to any of the other harmonics. If this were a sample of an acoustic instrument this statement might not be true but the one thing we know about this sample is that it came from another synthesizer which most likely used standard waveforms having harmonics at evenly-spaced frequency intervals.

As in the previous example we need to determine the pitch of the sample so that we know what key to play on the keyboard. In the case of Fre(a)koscope the cursor indicates that it is pitched to the A2 key. There are some similarities between this diagram and the one of the original Six-Trak patch back in the preceeding example. Like before, the broadening of the harmonics at high frequency indicates the presence of oscillator unison so we will want to turn that on. Here we also have undulating harmonics but unlike the previous example there is more of a pattern to the way they go up and down in amplitude. Notice that from about 1000 Hz to just above 2000 Hz that every other harmonic is lower than its neighbor. Also in the earlier example some harmonics were drastically diminished while others were only slightly. Here the change in amplitude is more uniform. These are

tell-tale signs that we are looking at a sawtooth wave mixed with either a second sawtooth or a square wave and not oscillator syncing or pulse-width modulation. The reason that one of these must be a sawtooth is that if we use just square waves or triangle waves, which have only odd harmonics, no matter how they are detuned from one another there will always be some missing harmonics. In the graph above there are *no* missing harmonics. Try playing with just square and triangle waves and detuning one of the oscillators to move its harmonics around. You'll find that no matter how the tuning is changed there are always missing harmonics.

We must now determine what waveform is used by the second oscillator as well as its pitch. Look at the harmonics of a single sawtooth tuned to A2 with unison and compare it to the harmonics of the sample.
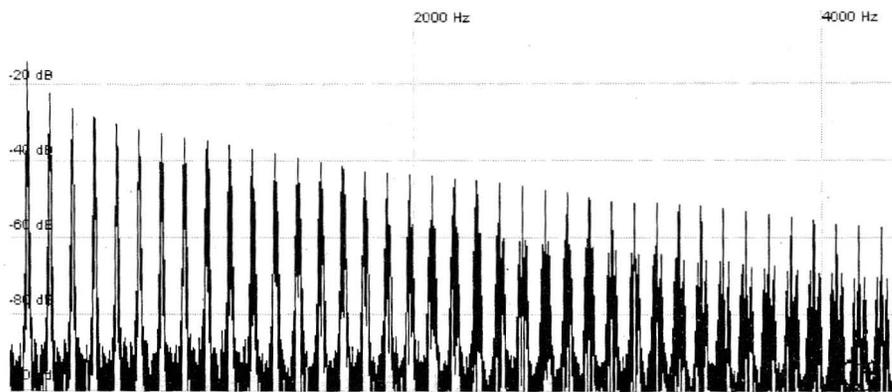
Figure 48. Sawtooth wave tuned to A2 with unison

We need to increase the amplitude of the even harmonics while leaving the odd harmonics unchanged. A square wave won't work because it is made of *only* odd harmonics. If we add a square wave to the sawtooth we get undulating harmonics but they are in the wrong order and the sound doesn't have the same bite as the original. Here is a graph of that.
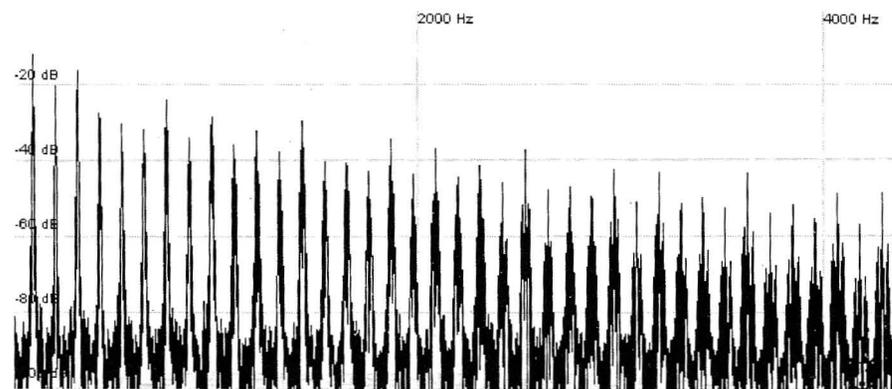
Figure 49. A square wave added with a sawtooth produces undulating harmonics but they are in the wrong order

If we tune the square up an octave its even worse. Now every fourth harmonic is increased starting at harmonic two.
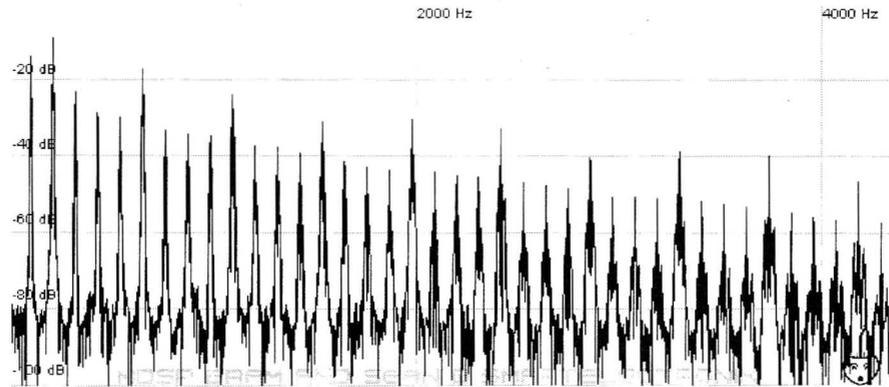
Figure 50. If the square wave is tuned up an octave the results match even less than before

The square wave doesn't work so we'll have to try using a second sawtooth. This second sawtooth must be tuned up one octave. Why? Everytime a sound is tuned up an octave not only does the frequency of the fundamental double but more importantly for this particular situation the frequency distance of the overtone harmonics also doubles. Here are the harmonics of a sawtooth pitched to A3
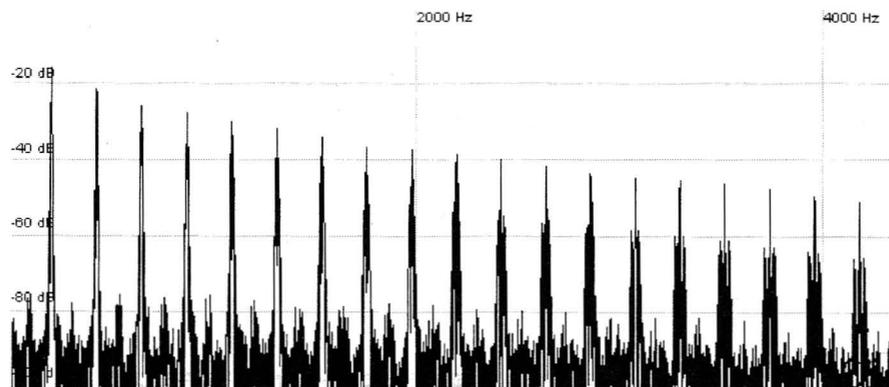
Figure 51. Harmonics of a sawtooth tuned up an octave to A3. Notice that the distance between harmonics has doubled

Notice that the harmonics are twice as far apart as those in the A2 sawtooth diagram. If we add the A2 sawtooth and the A3 sawtooth we will get the correct undulating harmonics.
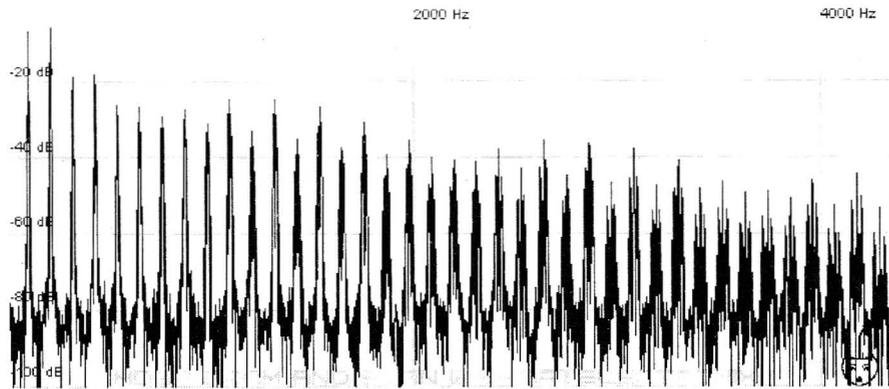
Figure 52. Two sawtooth waves tuned an octave apart are used to create the final product

Compare this to the original.
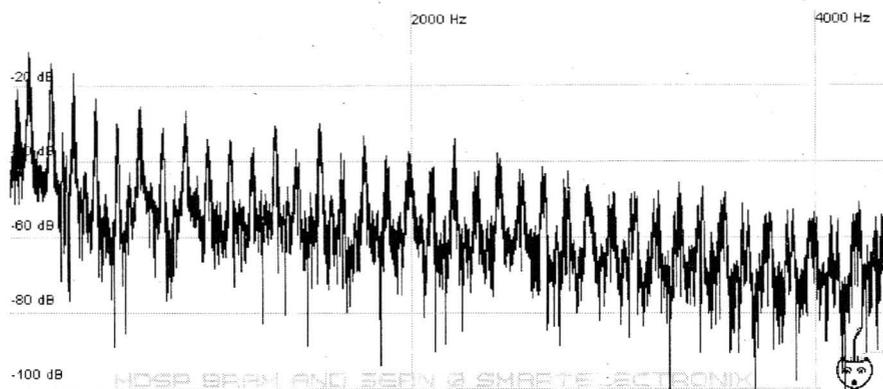
Figure 53. Harmonics of the original sample.

The majority of the harmonics are within 5 dB of the positions in the original and the undulating pattern is the same. There are slight differences but again these are to be expected since no two synthesizers are *exactly* the same. Set the attack and release times of the amplitude envelope to zero for a good bass patch. Compare the sound of the original to our new patch and notice they are very similar.

Emulating an acoustic instrument's harmonics and envelope: Clarinet

In this example we will not only imitate the harmonics of the instrument but we will also use the filter envelope to recreate the original envelope. Here we will recreate the sound of the clarinet taken from a sample provided by the royalty-free online sample library of the University of Iowa Electronic Music Studios (http://theremin.music.uiowa.edu/MIS.html). This sample can also be found on the CD.

Recreating the harmonics of acoustic instruments is more subjective than recreating patches produced by other analog synths. When recreating a synth patch we are basically trying to determine the settings of the synth from which it originally came and the end result is usually quite similar to the original. In the case of acoustic instruments however there are usually multiple ways to achieve harmonics that are similar to the original but with slight differences between the various methods. One method may do an excellent job of recreating the very lowest harmonics while another does a good job with higher harmonics. Recreating this clarinet as you will see is no exception.

Acoustic harmonic spectra are usually more complicated than those of analog synths. The goal is to match the synth's output as close as possible to the acoustic instrument while expecting to make sacrifices in one area of the audio spectrum to improve quality in other areas. The harmonics with the lowest frequencies are the most important. Take a look at the spectra of the clarinet.
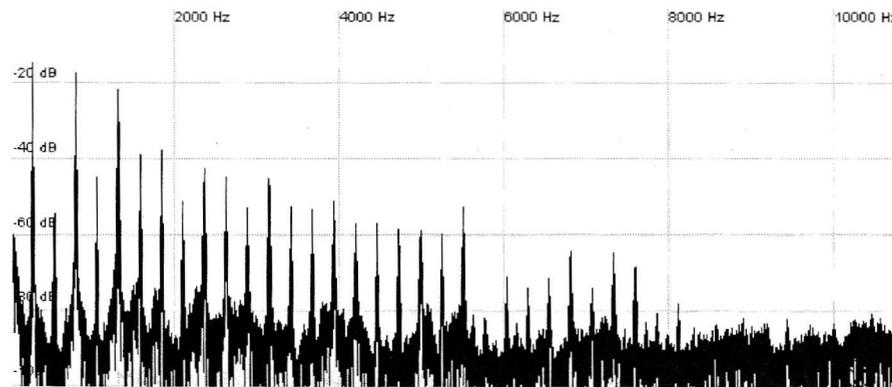
Figure 54. Harmonics of a clarinet

Here are the major details of the graph: The fundamental has a frequency corresponding to C4, harmonics 1, 3, and 5 are dramatically louder than the rest, and there are not really any harmonics after 8 kHz. Even though there are both even and odd harmonics such as would be provided by a sawtooth, the first three odd harmonics (1, 3, and 5) stand above the rest. This sound can probably be duplicated by mixing a sawtooth with either a square or triangle since either of these have odd harmonics. Because none of the odd harmonics above harmonic 5 really seem to stand out would suggest that we try a triangle over a square since the triangle's harmonics are quieter at higher frequencies. Here's what you'll get if you mix a triangle wave at full amplitude with a sawtooth wave -25 dB quieter than the triangle. Both at the same pitch.
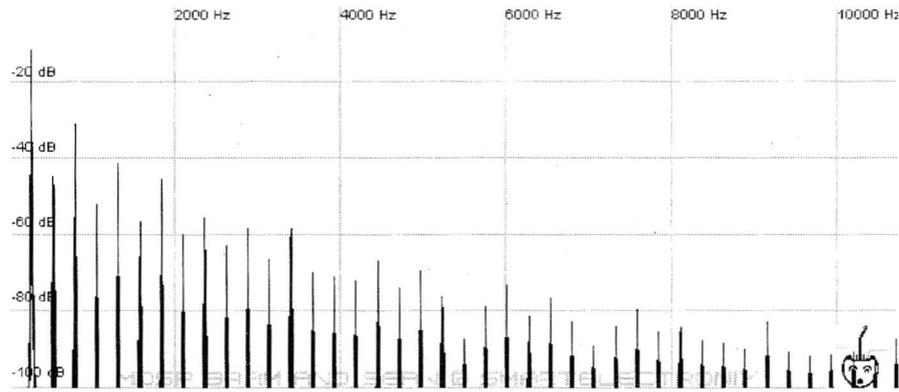
Figure 55. A triangle wave mixed with a quieter sawtooth wave produces harmonics that are similar

It sounds close and the harmonics look somewhat similar. There are probably clarinets out there that are nearly identical to this, but if we want to match this *particular* clarinet I think we can do better.

The way the first six harmonics undulate could be duplicated with a single pulse so let's give that a try. Starting with a pulse width of 50% (square wave) and barely decreasing the width much at all (only down to 49%) we get the following harmonic diagram.
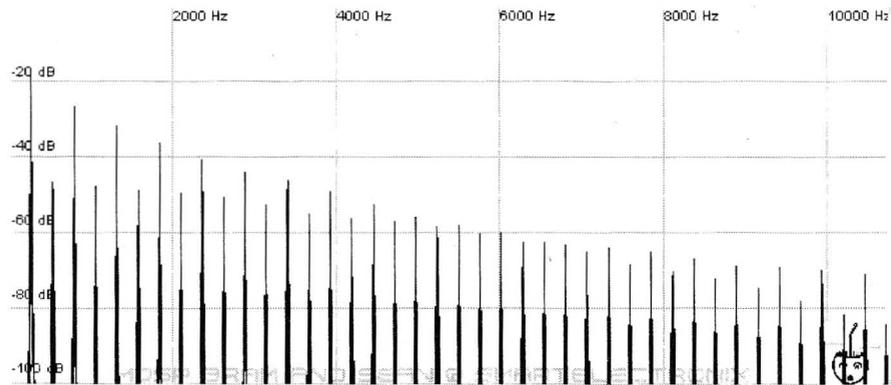
Figure 56. Pulse width set to 49%

There is in fact hardly any difference between this and a square wave. A pulse width of 49% versus a width of 50% may have dissimilar looking harmonic signatures yet they sound practically identical. Sound wise this is really close and we're probably splitting hairs to get closer but it is possible. We'll use the pulse width to help produce the first three loud odd harmonics but now we have to figure out how to produce the rest. If we ignore the first three odd harmonics then harmonics 6 and 7 would be the loudest and all harmonics to the left and to the right of these would fall away getting quieter in both directions. How do we imitate this odd and peculiar behavior? Often the best way to make harmonics do odd and peculiar things in a patch is to use a synced oscillator. Here's what we will do. Turn oscillator sync on and set the slave oscillator to use a square wave. Make sure the master oscillator is on but that its volume is turned all the way down at least to start with. We don't want to hear the master but we do want the slave to be able to sync to it. Now, while looking at the harmonic output of the slave's

square wave begin increasing the slave oscillator's pitch up quite high and as you do you should notice that successively higher pitched harmonics become the loudest on the display. Keep increasing the slave's pitch setting until it is 2 octaves and 5 semitones above the master and you should get harmonics similar to the following graph.
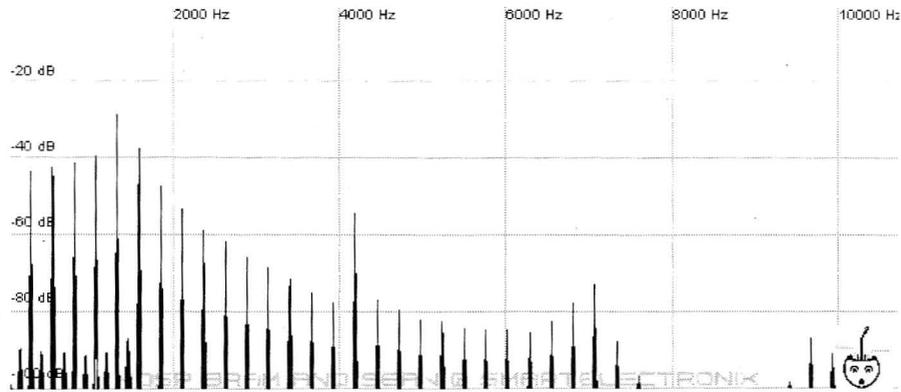
Figure 57. Output from a synced sawtooth tuned 2 octaves and 5 semitones above the master

With the exception of harmonics 1, 3, and 5 this matches the original spectra closely. Now to finish out the sound set the master oscillator to produce a pulse with a width of 49% just like we did a few paragraphs back. Turn the volume of the master all the way up and turn the slave down to where it is rather quiet around -30 dB below the master and what you get should be similar to the following graph.
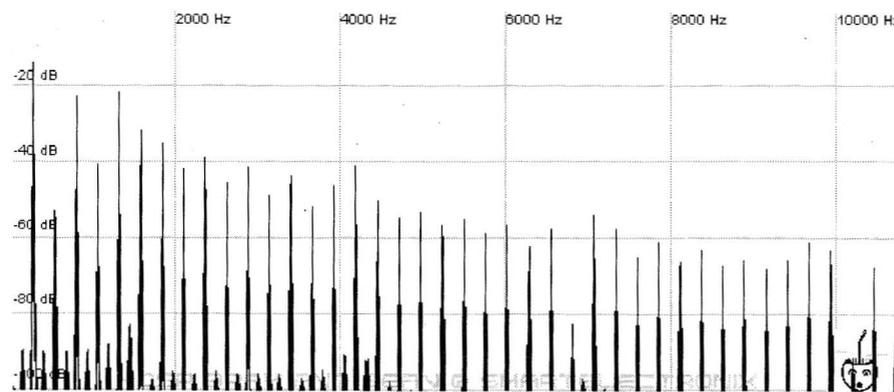
Figure 58. The master oscillator mixed with the slave oscillator

This is very close to what we want but the higher harmonics are still a bit loud so apply some low-pass filtering. A 24 dB filter set to a cutoff frequency of 2.5 kHz was used to produce the following result.
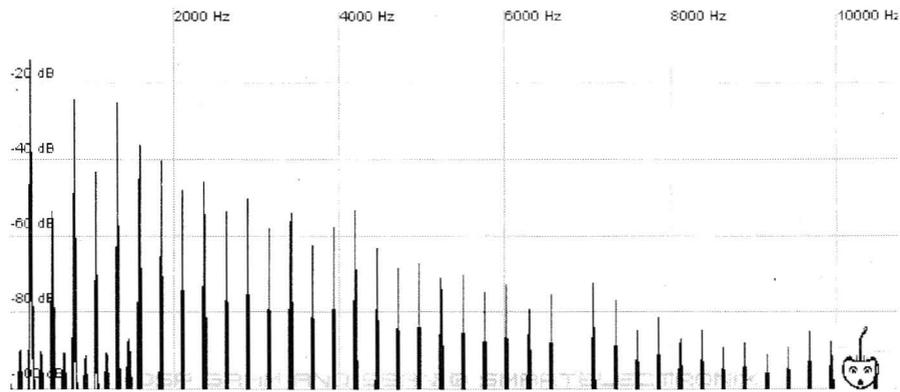
Figure 59. Applying some low-pass filtering to the mixed waveforms creates results that are very similar to the acoustic clarinet

Compare this to the original clarinet.
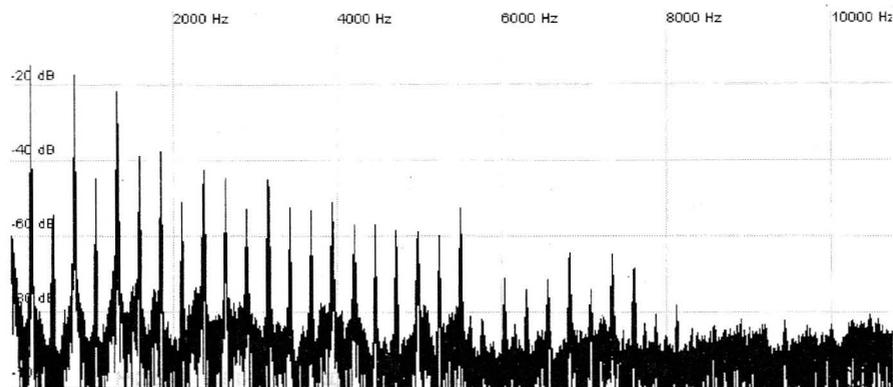
Figure 60. Harmonics of the clarinet

Since the first 5-10 harmonics are the most important we can consider our synthetic clarinet to be very similar to the acoustic clarinet. Not bad for old-school analog/subtractive synthesis! Call it good on the harmonics.   Now lets move on to the envelopes.

Here is the amplitude envelope of the clarinet in the audio sample.
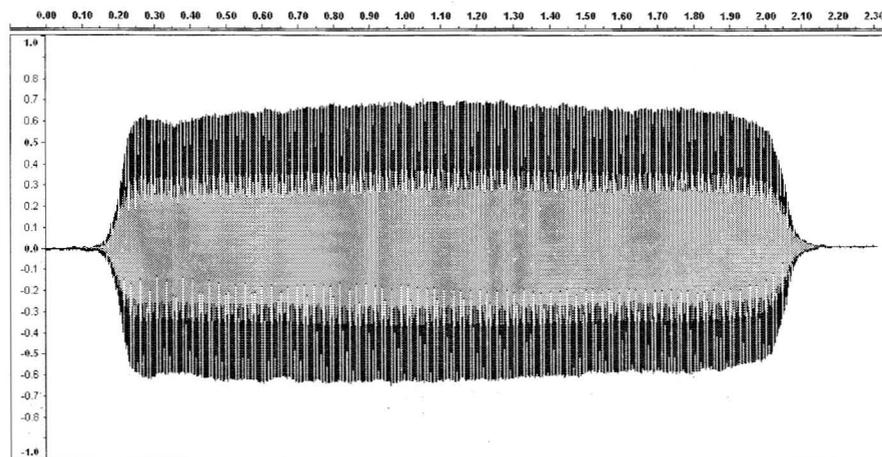
Figure 61. Envelope of the clarinet

EMULATING AN ACOUSTIC INSTRUMENT

We are going to duplicate this using the synthesizer's filter envelope. Usually when a synth programmer looks at a waveform like the one above the first inclination is to duplicate it using the synth's amplitude envelope. The above waveform is no doubt the natural amplitude envelope of the clarinet, however a more accurate reproduction of this envelope will actually use the synth's filter envelope. Keep in mind that whereas the amplitude envelope changes the amplitude of *all* harmonics equally the filter envelope changes the amplitude of *select* harmonics. In the case of a low-pass filter, the filter envelope changes the amplitude of the higher frequencies more dramatically. Therefore, the filter envelope is after a fashion a form of amplitude envelope.

Listen to the very beginning of the clarinet sample on the CD where the player just begins to blow into the instrument. The most obvious thing that you'll notice is that the amplitude of the note increases. Listen a bit more closely and you should also be able to notice that within that first fraction of a second the instrument sound becomes not only louder but also brighter. As the player begins to breath low-pitched harmonics amplify first followed by successively higher-pitched harmonics. As more and more high harmonics are added the overall sound becomes brighter and louder. At the end of the sample when the player stops blowing, the high harmonics are also the first to die out. This is true to varying extents in just about all instruments. Woodwinds in particular have a very strong relation between overall amplitude and amplitude changes of individual high harmonics.

Zooming into the beginning of the sample we see that the clarinet has an attack of about 0.15 seconds in length.
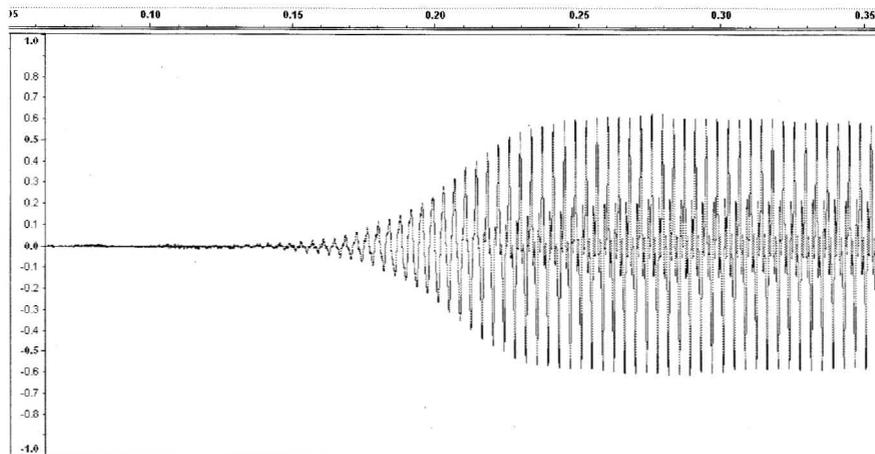
Figure 62. The attack portion of the clarinet sample

Zooming into the end we see that the "release" portion takes roughly 0.2 seconds
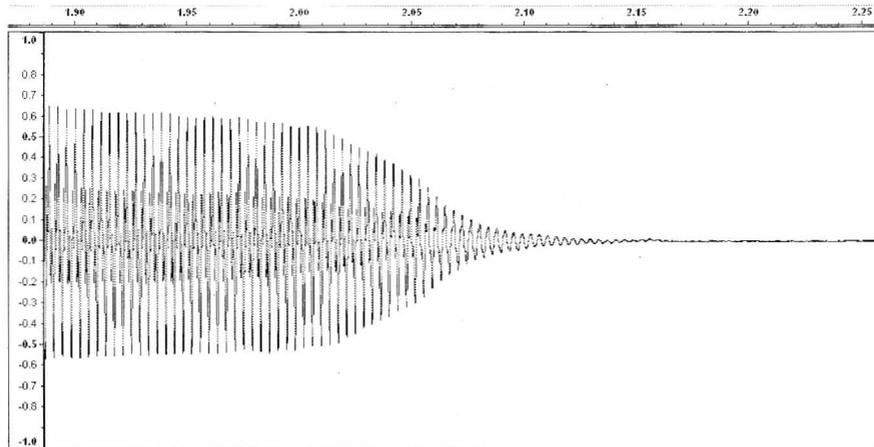
Figure 63. The release portion of the clarinet sample

In the synthesizer set the filter's cutoff frequency to its lowest setting, turn the filter envelope amount up to its maximum setting. Set the filter envelope's attack to 0.15 s, release to 0.2 s, and sustain all the way up to 100%. The decay time doesn't matter since sustain is at 100%. To keep the amplitude envelope from interfering make sure its attack is set to 0, sustain 100%, and release set to maximum. Now play a note on the keyboard. There's now a gradual attack and release but the sound is way too bright. While holding down a key, reduce the filter envelope's amount setting until the output sounds more like it did before we started messing with the envelope. To be a real stickler for accuracy look at your harmonic display while reducing the envelope amount until the higher-frequency harmonics are back where they were originally. Try playing it now and it should sound quite good.

Let's now add a faint hint of vibrato. Look closely at the original waveform in Figure 61 and notice how there are shallow peaks and valleys. They are hardly pronounced enough to be considered true vibrato. Excuse the jargon but they're more like quasi-periodic fluctuations caused by variations in the player's breath with a period of about 0.05 seconds. Remember frequency is equal to one over the period so: 1/0.05 s = 20 Hz. So set the LFO to around 20 Hz and route it to the amplitude. Set the LFOs modulation amount to where it just barely creates a vibrato effect. Around 2% is good if not less.
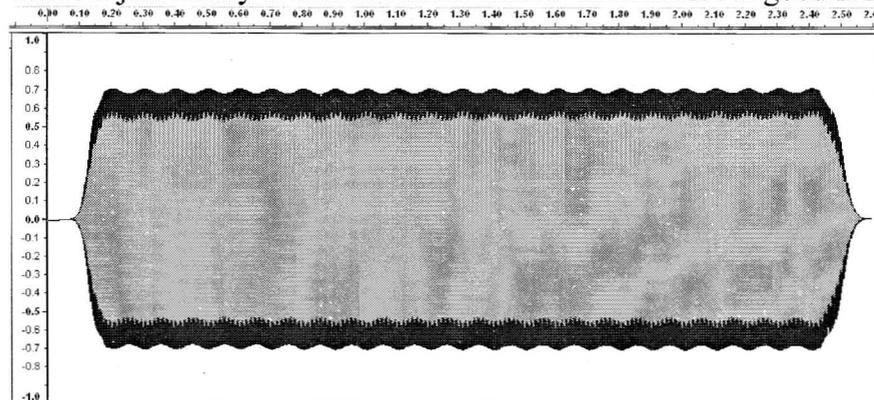
Figure 64. LFO used to create vibrato

Compare this with the original.
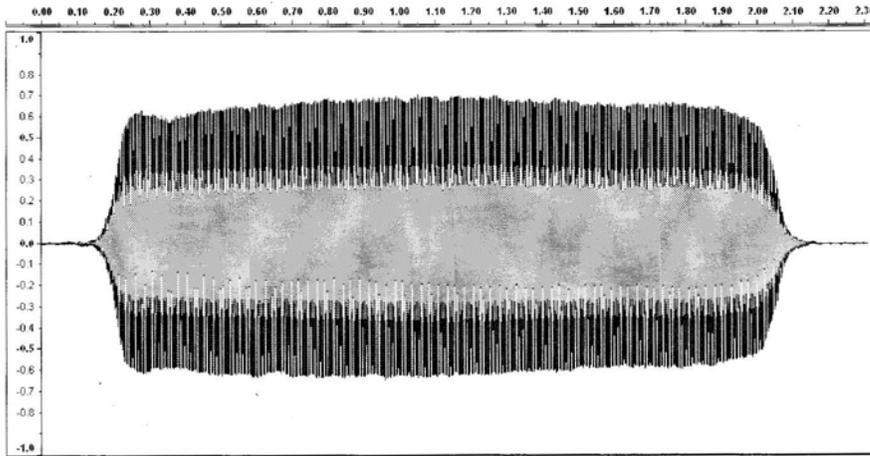
Figure 65. Notice how closely the attack and release in fig. 64 match with the original. The LFO was used to mimic the slight amplitude fluctuation in the original clarinet sample

The attack and release are very similar and we were able to use the LFO to somewhat duplicate the amplitude fluctuations. It sound's quite similar and we probably aren't going to get much closer than this. This not only sounds like a clarinet, but it sounds like the *specific* clarinet from the audio sample. Harmonic analysis allows us to get exceedingly close to the original source material. Analyze the harmonics of your own voice and synthesize it. Trust me, it's kind of creepy!

Ok, so the amplitude envelope wasn't used. Why are synthesizers built with amplitude envelopes then? Like I said before, woodwinds like the clarinet have envelopes that are very dependent upon the amplitude changes of their higher harmonics. When synthesizing most acoustic instruments both envelopes are usually used. Most instruments do not have such an extreme change in timbre at the attack and release as woodwinds like the clarinet and therefore the amplitude envelope can be given more control on these patches. An amplitude envelope can also be used to change the contour of the overall envelope. Look at the attack of our clarinet patch for example.
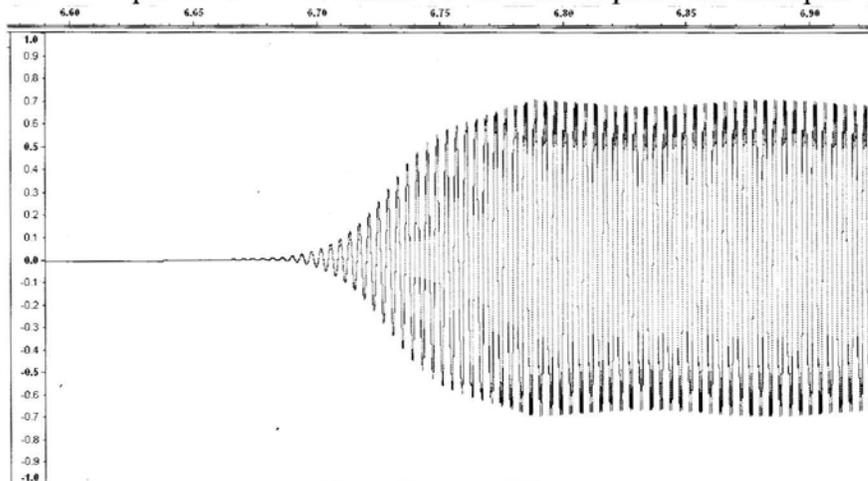
Figure 66. Attack portion of the clarinet patch

If we instead set both the filter *and* amplitude attacks to 0.15 seconds the overall attack will take the same amount of time but will come up more abruptly at the end of the attack.
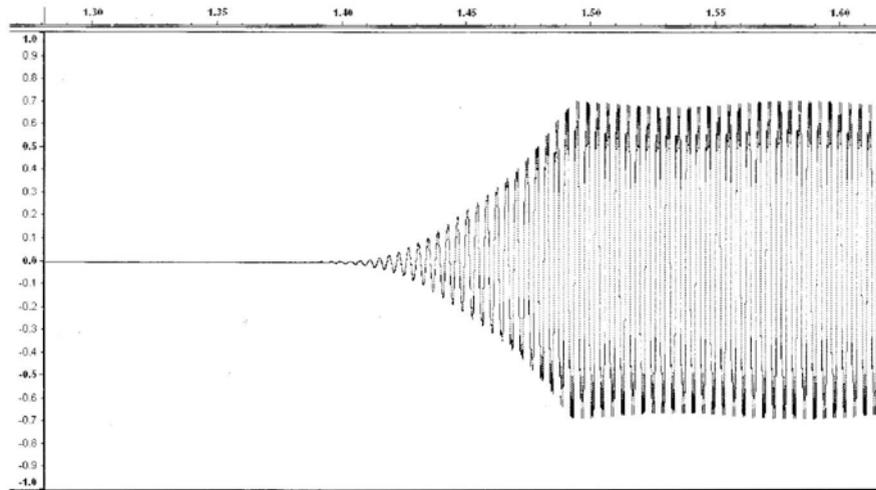
Figure 67. Setting both the filter and amplitude attack times to 0.15 seconcds changes the contour of the attack

The filter attack and amplitude attack have been used together to change the contour of the overall attack. Though we don't need to use this approach on the clarinet, it can be very useful for contouring the releases of percussion and plucked and hammered strings.

HOW TO PROGRAM THE COOKBOOK PATCHES

Oscillators

Wave : Waveform choices include sawtooth, square, triangle, and pulse width (PW). Pulse width values are given as a percentage of the waveform period. For instance a pulse width of 50% is a square wave and a pulse width of 25% is a pulse with a width that is half that of a square and sounds more "brassy." Pulse widths greater than 50% are not included as they are just inverted versions of those between 0% and 50% and therefore sound the same. You will need to know if your synth's pulse-width setting goes from 0%-50% or, more commonly, 0%-100%. If you have a synthesizer such as the ImpOscar which uses values of 50%-100% simply subtract the pulse width values in the book from 100 to get the value you need to use. If the synthesizer lacks the ability to set the pulse width of the square wave then substitute pulse-width values 0%-30% with a sawtooth wave and values from 30% - 50% with a square wave. If the synthesizer lacks a triangle wave then use a square wave in it's place when called for along with slightly more low-pass filtering. Triangle and square waves sound similar because they are both composed of odd harmonics but the higher harmonics of the triangle are less audible.

Tune: Values are listed in octaves, semitones, and cents. This is the value by which one oscillator should be detuned from the other oscillator. For patches where both oscillators have a tuning value listed it is the amount by which each oscillator is to be detuned from a common pitch.   If the tracking for oscillator 2 is turned off this field will list the specific pitch for which oscillator 2 is to be tuned.

Mix: This is the amplitude of the oscillator. Values are given in decibels where the maximum amplitude is 0 dB and quieter amplitudes are given as negative values. Some synthesizers display amplitudes where the maximum amplitude is not 0 dB. It may for instance be some positive value. In these cases just subtract the value listed in this book from the maximum value displayed by the synthesizer. For example if the synthesizer displays a maximum value of 13 dB

for the volume of one of the oscillators and the setting called for in this book is -7 dB then the oscillator should be set to a value of 6 dB. It's not that big of a deal just as long as you make sure the oscillators are separated by the same amount. The percentage to the right of the decibel value is the percentage of the maximum setting of the oscillator amplitude. This can be used for synthesizers that do not display values in decibels. Since oscillators on different synths have amplitude adjusters with different responses this percentage is a ballpark approximation and may require some adjustment by ear. For more accuracy it is best to calibrate the synths mix settings as explained in the last section of this book.

Oscillator 2 tracking: This is either on or off. When on the pitch of oscillator 2 will track the keyboard. When off oscillator 2 will play the same pitch on all keys.

Oscillator 2 sync: When on it syncs oscillator 2 to oscillator 1. You may need to check your synthesizer's documentation to be sure which of its oscillators syncs to the other when syncing is turned on.

Noise

Noise: White noise source set to either on or off.

Mix: Amplitude of the noise source given in decibels. The statements made above in the section on oscillator mix apply here as well.

LFO

Routing: Sets the destination of the LFO. Destinations include amplitude, pitch, filter cutoff, and pulse width.

Wave: Waveform used by the LFO as a modulation source. Waves used are sine, triangle, and square.

Frequency: Sets the rate of the LFO. Values are given in Hertz (Hz). Qualitative suggestions are given in italics.

Depth: Sets the amount by which the destination is modulated by the LFO. This is given as a percent of the settings range when modulating amplitude, filter cutoff, and pulse width. The knob/slider that controls depth would be turned half way for a depth of 50%. When pitch is set as the destination the depth is given in terms of the amount of pitch that a played note will vary.

Glide: Also known as portamento. Set to either on or off

Time: Glide time. The amount of time that it takes for the pitch to change from one struck note to the next.

Unison: Set to on or off. Your synthesizer may allow you to control the number of oscillators that are used. Usually 4 to 6 oscillators sound good.

Voices: The polyphony of the patch. Either set to a specific number, mono, or multi. Multi means there is no restriction on the amount of polyphony.

Low-Pass Filter

Cutoff: Values are given for 24 dB filters and 12 dB filters. Values are given in hertz and kilohertz and go from 20 Hz to 20 KHz. If your synthesizer uses something other than a 24 dB or 12 dB filter you can figure out the required setting by extrapolating from the given values for 24 and 12 dB filters. Generally, the smaller the amount of attenuation (the decibel number) the lower the cutoff frequency should be. With a greater amount of attenuation set the cutoff to a higher frequency. For example, if the listing for a 12 dB filter calls for a cutoff of 200 Hz then a 6 dB filter would be set to a frequency less than 200 Hz. A 36 dB filter would be set to a cutoff frequency higher than what is listed for the 24 dB filter value, etc.

To the right of each cutoff frequency is listed the percentage of the full dial

position. Of course each synthesizer's filter cutoff will follow a different response curve so this value is provided as a rough estimate. It is based on the average taken from ten different synthesizers. Actual values can vary drastically. *Following these percentages will require at least some adjustment by ear.* If you patch a sound and it doesn't sound right then most likely it is due to a discrepancy between the filter-cutoff setting percentages and the actual filter cutoff of the synth. As a simple test I would recommend patching the Angels patch on pg 33. If it sounds more like a brass instrument than a vocal then the listed percentages are higher than what your synth produces. Set the cutoff for each patch below the listed percentage value. If the angels sound like they are being smothered then the listed percentages are too low. Set the cutoff for each patch above the listed percentage value. To achieve higher accuracy on synthesizers that do not display cutoff in hertz it is recommended to follow the calibration instructions at the end of the book. This calibration procedure for the filter while not critical is HIGHLY recommended! If you only calibrate one parameter it should be the filter cutoff.

Resonance: The amount of resonance applied to the filter. Values are given as a percentage where 0% applies no resonance and 100% applies the maximum amount.

Envelope: Sets the amount by which the filter envelope controls the amount of filtering. On some synths this is called the Initial Level setting. Again 0% is the minimum value and 100% is the maximum.

ADSR: Attack, Decay, and Release are given in terms of time with values in seconds. Sustain is given as a percentage of its maximum amount.

Amplifier

ADSR: The same as the envelope for the filter. Attack, Decay, and Release are given in terms of time with values in seconds. Sustain is given as a percentage of its maximum amount.

Alternate Patch Settings

Some of the patches in the book have alternate patch settings which are intended to be used if the synth does not have the feature listed in the primary patch. For instance let's say that you are programming the cello patch which calls for oscillator 1 to use a pulse with a width of 10% but your synth doesn't have the ability to produce a pulse. Use the alternate setting which calls for a sawtooth wave instead.

Note about times: Some synthesizers list times in seconds (s) and others list times in milliseconds (ms). This book uses seconds but the conversion is straightforward. Simply move the decimal three places. The relation is:

1 s = 1000 ms

0.1 s = 100 ms

0.01 s = 10ms

0.001 s = 1 ms

So for example 0.08 s is the same thing as 80 ms, 0.28 s is also 280 ms, and 1.35 s is 1350 ms, etc.

STRINGS

Banjo

Oscillator 1		LFO		Low-pass filter			
Wave:	PW = 20%	Routing:	Amplitude		24db	12db	
Tune:	-	Wave:	Triangle	Cutoff:	2.9khz *72%*	1.5khz *63%*	
		Frequency:	10hz *fast*	Resonance:	0%		
Mix:	0db *100%*	Depth:	10%	Envelope:	75%		
Oscillator 2		Glide:	-	**A**	**D**	**S**	**R**
Wave:	PW = 10%	time:	-	0s	0.19s	0%	0.19s
Tune:	+5 semi	Unison:	off	**Amplifier**			
		Voices:	5	**A**	**D**	**S**	**R**
Mix:	-7.6db *80%*			0s	0.67s	0%	0.67s
Osc 2 Track:	on						
Osc 2 Sync:	**on**						
Noise:	off						
Mix:	-						

Cello

Oscillator 1		LFO		Low-pass filter			
Wave:	PW = 10%	Routing:	Amplitude		24db	12db	
Tune:	-	Wave:	Sine	Cutoff:	40hz *10%*	40hz *10%*	
		Frequency:	7.5hz *moderate*	Resonance:	0%		
Mix:	0db *100%*	Depth:	5%	Envelope:	90%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	Square	time:	-	0s	3.29s	78%	max
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	0db *100%*			0.06s	max	100%	0.30s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Alternate: Osc 1: sawtooth

Double Bass

Oscillator 1			LFO			Low-pass filter			
Wave:	PW =45%		Routing:	Pitch				24db	12db
Tune:	-1 octave		Wave:	Triangle		Cutoff:		1.6khz *63%*	750hz *52%*
			Frequency:	5hz *moderate*		Resonance:		0%	
Mix:	0db *100%*		Depth:	11 cents *light*		Envelope:		0%	
Oscillator 2			Glide:	Off		**A**	**D**	**S**	**R**
Wave:	Square		time:	-		-	-	-	-
Tune:	-		Unison:	Off		**Amplifier**			
			Voices:	Multi		**A**	**D**	**S**	**R**
Mix:	-17db *60%*					0.35s	max	100%	0.19s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Dulcimer

Oscillator 1			LFO			Low-pass filter			
Wave:	PW=25%		Routing:	Amplitude				24db	12db
Tune:	-7 semi		Wave:	Triangle		Cutoff:		600hz *49%*	200hz *33%*
			Frequency:	1.5hz *very slow*		Resonance:		0%	
Mix:	-7.6db *80%*		Depth:	22%		Envelope:		50%	
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=5%		time:	-		0s	1.69s	0%	1.78s
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0s	4.00s	0%	4.00s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Guitar Acoustic

Oscillator 1			LFO			Low-pass filter				
Wave:	PW =25%		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		3.1khz *73%*	2.0khz *67%*	
			Frequency:	-		Resonance:		0%		
Mix:	0db *100%*		Depth:	-		Envelope:		70%		
Oscillator 2			Glide:	off		**A**		**D**	**S**	**R**
Wave:	PW=10%		time:	-		0s		0.35s	0%	0.29s
Tune:	+10 semi		Unison:	off		**Amplifier**				
			Voices:	multi		**A**		**D**	**S**	**R**
Mix:	-4db *90%*					0s		1.70s	0%	1.70s
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

Guitar Electric

Oscillator 1			LFO			Low-pass filter				
Wave:	PW =20%		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		20khz *100%*	20khz *100%*	
			Frequency:	-		Resonance:		0%		
Mix:	-13.4db *65%*		Depth:	-		Envelope:		0%		
Oscillator 2			Glide:	off		**A**		**D**	**S**	**R**
Wave:	PW=15%		time:	-		-		-	-	-
Tune:	+10semi		Unison:	off		**Amplifier**				
			Voices:	multi		**A**		**D**	**S**	**R**
Mix:	0db *100%*					0s		1.70s	0%	1.70s
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

Harp

Oscillator 1		LFO		Low-pass filter				
Wave:	Any waveform	Routing:	Amplitude			__24db__	__12db__	
Tune:	-	Wave:	Triangle	Cutoff:	40hz _10%_	40hz _10%_		
		Frequency:	7.5hz _moderate_	Resonance:	5%			
Mix:	-40db _0%_	Depth:	6%	Envelope:	60%			
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**	
Wave:	Square	time:	-	0s	0.37s	78%	0.94s	
Tune:	+6 semi	Unison:	off	**Amplifier**				
		Voices:	multi	**A**	**D**	**S**	**R**	
Mix:	0db _100%_			0s	1.30s	0%	3.30s	
Osc 2 Track:	on							
Osc 2 Sync:	**on**							
Noise:	off							
Mix:	-							

Alternate: Osc 1: sawtooth, 0 db 100%, Osc 2: triangle, tune to same pitch as osc 1 sync off

Hurdy Gurdy

Oscillator 1		LFO		Low-pass filter				
Wave:	PW = 15%	Routing:	-			__24db__	__12db__	
Tune:	-	Wave:	-	Cutoff:	40hz _10%_	40hz _10%_		
		Frequency:	-	Resonance:	0%			
Mix:	-4.1db _90%_	Depth:	-	Envelope:	100%			
Oscillator 2		Glide:	on	**A**	**D**	**S**	**R**	
Wave:	Square	time:	0.04s _very short_	0.04s	max	100%	0.23s	
Tune:	D3	Unison:	off	**Amplifier**				
		Voices:	mono	**A**	**D**	**S**	**R**	
Mix:	0db _100%_			0s	max	100%	0.85s	
Osc 2 Track:	off							
Osc 2 Sync:	off							
Noise:	off							
Mix:	-							

Kora

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	Pitch			**24db**	**12db**		
Tune:	-		Wave:	Triangle		Cutoff:	20khz *100%*	20khz *100%*		
			Frequency:	8hz *fast*		Resonance:	0%			
Mix:	0db *100%*		Depth:	8 cents *light*		Envelope:	0%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	-		-	-	-	-	
Tune:	+1 octave		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	2.33s	0%	2.33s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Lute

Oscillator 1			LFO			Low-pass filter				
Wave:	Sawtooth		Routing:	Pitch			**24db**	**12db**		
Tune:	+1 octave		Wave:	Triangle		Cutoff:	20khz *100%*	20khz *100%*		
			Frequency:	5hz *moderate*		Resonance:	0%			
Mix:	0db *100%*		Depth:	47 cents *medium*		Envelope:	0%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	-		-	-	-	-	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	-7db *85%*					0s	0.66s	0%	0.66s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

note: The vibrato caused by the LFO should have a "warbling" quality

Mandocello

Oscillator 1			LFO			Low-pass filter			
Wave:	PW=25%		Routing:	Amplitude			**24db**	**12db**	
Tune:	-		Wave:	Triangle		Cutoff:	12khz *92%*	12khz *92%*	
			Frequency:	3hz *slow*		Resonance:	30%		
Mix:	0db *100%*		Depth:	20%		Envelope:	35%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=15%		time:	-		0s	1.78s	0%	0s
Tune:	+4 cents		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0s	1.50s	0%	1.50s
Osc 2 Track:	on								
Osc 2 Sync:	-								
Noise:	off								
Mix:	-								

Mandolin

Oscillator 1			LFO			Low-pass filter			
Wave:	PW=25%		Routing:	Amplitude			**24db**	**12db**	
Tune:	-		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*	
			Frequency:	2hz *slow*		Resonance:	0%		
Mix:	0db *100%*		Depth:	9%		Envelope:	100%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=15%		time:	-		0s	2.07s	0%	2.13s
Tune:	+1 octave		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	-9.6db *75%*					0s	0.09s	20%	2.33s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Riti

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	Pitch				**24db**	**12db**	
Tune:	-		Wave:	Triangle		Cutoff:	7.9khz *86%*	6.5khz *84%*		
			Frequency:	6hz *moderate*		Resonance:	100%			
Mix:	0db *100%*		Depth:	10 cents *shallow*		Envelope:	85%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	PW = 40%		time:	0.02s		0s	1.19s	0%	0s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	0.09s	0%	0s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Sitar

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 10%		Routing:	Pitch				**24db**	**12db**	
Tune:	-		Wave:	Triangle		Cutoff:	20khz *100%*	20khz *100%*		
			Frequency:	6.5hz *moderate*		Resonance:	0%			
Mix:	0db 100%		Depth:	40 cents *medium*		Envelope:	0%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW=35%		time:	-		-	-	-	-	
Tune:	C3		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db 100%					0s	0.93s	0%	0.93s	
Osc 2 Track:	**off**									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Standup Bass

Oscillator 1	
Wave:	PW =25%
Tune:	-1 octave
Mix:	0db 100%
Oscillator 2	
Wave:	Triangle
Tune:	-1 octave
Mix:	0db 100%
Osc 2 Track:	on
Osc 2 Sync:	off
Noise:	off
Mix:	-

LFO	
Routing:	Amplitude
Wave:	Triangle
Frequency:	15hz *fast*
Depth:	10%
Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter

	24db	12db		
Cutoff:	40 hz *10%*	40 hz *10%*		
Resonance:	0%			
Envelope:	75%			
A	**D**	**S**	**R**	
0s	2.33s	60%	2.33s	

Amplifier

A	**D**	**S**	**R**
0s	1.28s	0%	1.38s

Viola

Oscillator 1	
Wave:	Sawtooth
Tune:	-
Mix:	-11.6db *70%*
Oscillator 2	
Wave:	PW=25%
Tune:	+7 semi
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	**on**
Noise:	off
Mix:	-

LFO	
Routing:	Pitch osc 2
Wave:	Triangle
Frequency:	4hz *moderate*
Depth:	80 cents
Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter

	24db	12db		
Cutoff:	4.9khz *80%*	2.3khz *69%*		
Resonance:	0%			
Envelope:	0%			
A	**D**	**S**	**R**	
-	-	-	-	

Amplifier

A	**D**	**S**	**R**
0.81s	max	100%	0.73s

Alternate: Osc 2: triangle, tune to same pitch as osc 1, sync off, LFO: routing – amplitude, frequency 5 hz (moderate), depth 8%, Filter, 24 db and 12 db - 20khz(100%

Violin

Oscillator 1		LFO		Low-pass filter		
Wave:	Any waveform	Routing:	Pitch osc 2		24db	12db
Tune:	-	Wave:	Triangle	Cutoff:	3.2khz 73%	1.9khz 66%
		Frequency:	5hz *moderate*	Resonance:	0%	
Mix:	-40db 0%	Depth:	88 cents	Envelope:	0%	

		LFO		Amplifier A / D / S / R

	A	D	S	R
(Filter)	-	-	-	-
Amplifier	0.03s	max	100%	0.35s

Oscillator 2			
Wave:	Sawtooth	Glide:	off
Tune:	+6 semi	time:	-
Mix:	0db 100%	Unison:	off
		Voices:	multi
Osc 2 Track:	on		
Osc 2 Sync:	**on**		
Noise:	off		
Mix:	-		

Alternate: Osc 1: PW=25%, 0db (100%), Osc 2: Square, tune same as osc 1,sync off LFO: route amplitude, frequency 3.5 hz(slow), depth 10%, Filter: 24db and 12db – 9.8khz(90%)

Note: Use any waveform for osc 1 and make sure it is on and level is all the way down. Osc 2 is the only sound source but must sync to osc 1

WOODWINDS

Bagpipes

Oscillator 1			LFO			Low-pass filter			
Wave:	Sawtooth		Routing:	-			24db	12db	
Tune:	-		Wave:	-		Cutoff:	20khz *100%*	20khz *100%*	
			Frequency:	-		Resonance:	0%		
Mix:	0db *100%*		Depth:	-		Envelope:	0%		
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**
Wave:	Square		time:	0.04s		-	-	-	-
Tune:	C3		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	-11db *75%*					0.02s	max	100%	0.85s
Osc 2 Track:	off								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Bass Clarinet

Oscillator 1			LFO			Low-pass filter			
Wave:	Square		Routing:	Pitch			24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*	
			Frequency:	7.5hz *moderate*		Resonance:	0%		
Mix:	0db *100%*		Depth:	5 cents *very light*		Envelope:	65%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=5%		time:	-		0.09s	max	100%	0.23s
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0s	max	100%	0.23s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Alternate: Osc 1: -3.6db (85%), Osc 2: sawtooth, -1 octave, -14 db (60%), Filter: envelope 75%

Bassoon

Oscillator 1	
Wave:	PW = 10%
Tune:	-
Mix:	0db *100%*

Oscillator 2	
Wave:	Square
Tune:	+1 octave
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	off

Noise:	off
Mix:	-

LFO	
Routing:	Amplitude
Wave:	Triangle
Frequency:	7.5hz *moderate*
Depth:	5%

Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter		24db	12db
Cutoff:		40hz *10%*	40hz *10%*
Resonance:		0%	
Envelope:		60%	

A	D	S	R
0.16s	1.13s	83%	0.28s

Amplifier			
A	D	S	R
0s	max	100%	max

Alternate: Osc 1: Square, Filter: envelope: 75%

Clarinet

Oscillator 1	
Wave:	PW = 45%
Tune:	-
Mix:	0db *100%*

Oscillator 2	
Wave:	-
Tune:	-
Mix:	-
Osc 2 Track:	on
Osc 2 Sync:	off

Noise:	off
Mix:	-

LFO	
Routing:	Pitch
Wave:	Sine
Frequency:	7.5hz *moderate*
Depth:	1 cent *very light*

Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter		24db	12db
Cutoff:		40hz *10%*	40hz *10%*
Resonance:		0%	
Envelope:		55%	

A	D	S	R
0.15s	max	100%	max

Amplifier			
A	D	S	R
0s	max	100%	0.32s

Alternate: Osc 1: square, Osc2: sawtooth, -9.6db (70%), Filter: envelope 100%

Conch Shell

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	Amplitude				**24db**	**12db**	
Tune:	-		Wave:	Triangle		Cutoff:	1khz *56%*	400hz *43%*		
			Frequency:	2hz *slow*		Resonance:	0%			
Mix:	0db *100%*		Depth:	5%		Envelope:	55%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	0.012s *fast*		0.11s	0.07s	80%	0.13s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-14.1db *65%*					0.11s	max	100%	0.13s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Contrabassoon

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 10%		Routing:	Amplitude				**24db**	**12db**	
Tune:	-1 octave		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	7.5hz *moderate*		Resonance:	0%			
Mix:	0db *100%*		Depth:	5%		Envelope:	60%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	-		0.09s	1.13s	83%	0.28s	
Tune:	+1 octave		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.29s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Alternate: Osc 1: square, Osc 2: square, Filter: envelope: 75%

Didgeridoo

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	Filter cutoff				24db	12db	
Tune:	-1 octave		Wave:	Triangle		Cutoff:		1.5khz *62%*	800hz *53%*	
			Frequency:	1hz *slow*		Resonance:		80%		
Mix:	0db *100%*		Depth:	10% *shallow*		Envelope:		0%		
Oscillator 2			Glide:	on		**A**	**D**		**S**	**R**
Wave:	PW = 25%		time:	moderate		-	-		-	-
Tune:	+9 semi		Unison:	off		Amplifier				
			Voices:	mono		**A**	**D**		**S**	**R**
Mix:	0db *100%*					0.03s	max		100%	0.50s
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

notes: Play in lower octaves. Try unison using 2-3 voices

English Horn (Cor Anglais)

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 15%		Routing:	Pitch				24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:		40hz *10%*	40hz *10%*	
			Frequency:	7.5hz *moderate*		Resonance:		0%		
Mix:	0db *100%*		Depth:	3 cents shallow		Envelope:		60%		
Oscillator 2			Glide:	-		**A**	**D**		**S**	**R**
Wave:	Triangle		time:	-		0.07s	max		100%	max
Tune:	+1 octaves		Unison:	off		Amplifier				
			Voices:	multi		**A**	**D**		**S**	**R**
Mix:	-28.6db *30%*					0s	max		100%	0.37s
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Alternate: Osc 1: triangle, Osc 2: +2 octaves, Filter: envelope: 95%

67
WOODWINDS

Flute

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 25%		Routing:	Amplitude				24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	3.5hz *moderate*		Resonance:	0%			
Mix:	0db *100%*		Depth:	7%		Envelope:	60%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	-		time:	0.015s		0.11s	max	100%	0.28s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	-					0.11s	max	100%	0.28s	
Osc 2 Track:	-									
Osc 2 Sync:	-		*Alternate: Osc 1: sawtooth*							
Noise:	off									
Mix:	-									

Oboe

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 20%		Routing:	Pitch				24db	12db	
Tune:	-		Wave:	Sine		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	7.5hz *moderate*		Resonance:	25%			
Mix:	0db *100%*		Depth:	1 cents *very light*		Envelope:	60%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW = 5%		time:	-		0.15s	max	100%	max	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.14s	
Osc 2 Track:	on									
Osc 2 Sync:	off		*Alternate: Osc 1: sawtooth, -3.6db (85%), Osc 2: triangle, +2 octaves, -24.9 db (40%),*							
Noise:	off		*Filter resonance 20%, Envelope 80%*							
Mix:	-									

Piccolo

Oscillator 1			LFO			Low-pass filter			
Wave:	Sawtooth		Routing:	Amplitude			24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*	
			Frequency:	5hz *moderate*		Resonance:	50%		
Mix:	-9db *80%*		Depth:	5%		Envelope:	60%		
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**
Wave:	Triangle		time:	0.01s		0.11s	max	100%	0.28s
Tune:	+1 octave		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0.11s	max	100%	0.28s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Alternate: Osc 1: square, +1 octave, -25db (40%), Osc 2: +1 octave, -18db (55%), Filter envelope 100%

BRASS

French Horn

Oscillator 1			LFO			Low-pass filter			
Wave:	PW = 10%		Routing:	-			24db	12db	
Tune:	-		Wave:	-		Cutoff:	40hz 10%	40hz 10%	
			Frequency:	-		Resonance:	20%		
Mix:	0db 100%		Depth:	-		Envelope:	45%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	-		time:	-		0.05s	5.76s	94%	0.39s
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	-					0s	3.9s	96%	0.93s
Osc 2 Track:	-								
Osc 2 Sync:	-								
Noise:	off								
Mix:	-								

Alternate: Osc 1: sawtooth, Osc 2: triangle, +2 octaves, Fiter: envelope 55%

Harmonica

Oscillator 1			LFO			Low-pass filter			
Wave:	PW=2%		Routing:	PW osc 1			24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	1.9khz 66%	1.2khz 59%	
			Frequency:	1.6hz *slow*		Resonance:	0%		
Mix:	-11db 75%		Depth:	85% *deep*		Envelope:	65%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=15%		time:	-		0.16s	max	100%	0.16s
Tune:	+8 semi		Unison:	off		**Amplifier**			
	+7 cents		Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db 100%					0.13s	0.33s	50%	0.14s
Osc 2 Track:	on								
Osc 2 Sync:	**on**								
Noise:	off								
Mix:	-								

Penny Whistle

Oscillator 1		
Wave:	Square	
Tune:	-	
Mix:	0db *100%*	
Oscillator 2		
Wave:	Triangle	
Tune:	+2 octaves	
Mix:	-7db *85%*	
Osc 2 Track:	on	
Osc 2 Sync:	off	
Noise:	off	
Mix:	-	

LFO		
Routing:	Amplitude	
Wave:	Triangle	
Frequency:	90hz *extreme*	
Depth:	5% *shallow*	
Glide:	on	
time:	0.04s	
Unison:	off	
Voices:	multi	

Low-pass filter			
	24db	12db	
Cutoff:	1.1khz *58%*	800hz *53%*	
Resonance:	80%		
Envelope:	77%		
A	**D**	**S**	**R**
0.11s	max	100%	max
Amplifier			
A	**D**	**S**	**R**
0.11s	max	100%	0.57s

Saxophone

Oscillator 1		
Wave:	PW = 30%	
Tune:	-	
Mix:	0db *100%*	
Oscillator 2		
Wave:	PW = 45%	
Tune:	+8 semi	
Mix:	-9.6db *75%*	
Osc 2 Track:	on	
Osc 2 Sync:	**on**	
Noise:	off	
Mix:	-	

LFO		
Routing:	Pitch osc1, osc2	
Wave:	Sine	
Frequency:	7.5hz *moderate*	
Depth:	10 cnts *(shallow)*	
Glide:	off	
time:	-	
Unison:	off	
Voices:	multi	

Low-pass filter			
	24db	12db	
Cutoff:	40hz *10%*	40hz *10%*	
Resonance:	0%		
Envelope:	90%		
A	**D**	**S**	**R**
0.14s	0.37s	78%	max
Amplifier			
A	**D**	**S**	**R**
0s	max	100%	0.30s

Alternate: Osc 1: sawtooth, -3.6db (85%), Osc 2: sawtooth, +1 octave, sync off, LFO: route amplitude, depth 6%

Trombone

Oscillator 1				
Wave:	Sawtooth			
Tune:	-			
Mix:	0db *100%*			
Oscillator 2				
Wave:	Triangle			
Tune:	+1 octave			
Mix:	0db *100%*			
Osc 2 Track:	on			
Osc 2 Sync:	off			
Noise:	off			
Mix:	-			

LFO		
Routing:	Amplitude	
Wave:	Triangle	
Frequency:	5hz *moderate*	
Depth:	5%	
Glide:	off	
time:	-	
Unison:	off	
Voices:	multi	

Low-pass filter				
		24db	12db	
Cutoff:		900hz *55%*	200hz *33%*	
Resonance:		0%		
Envelope:		30%		
A		**D**	**S**	**R**
0.11s		max	100%	0.18s
Amplifier				
A		**D**	**S**	**R**
0.06s		max	100%	0.50s

Trumpet

Oscillator 1				
Wave:	Sawtooth			
Tune:	-			
Mix:	0db *100%*			
Oscillator 2				
Wave:	Sawtooth			
Tune:	+1 octave			
Mix:	-14.4db *65%*			
Osc 2 Track:	on			
Osc 2 Sync:	off			
Noise:	off			
Mix:	-			

LFO		
Routing:	Amplitude	
Wave:	Triangle	
Frequency:	7.5hz *moderate*	
Depth:	5%	
Glide:	off	
time:	-	
Unison:	off	
Voices:	multi	

Low-pass filter				
		24db	12db	
Cutoff:		50hz *13%*	40hz *10%*	
Resonance:		15%		
Envelope:		75%		
A		**D**	**S**	**R**
0.08s		max	100%	0.19s
Amplifier				
A		**D**	**S**	**R**
0s		max	100%	0.19s

Tuba

Oscillator 1			LFO			Low-pass filter				
Wave:	Sawtooth		Routing:	Amplitude				24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	2.4hz *slow*		Resonance:	-			
Mix:	-6db *85%*		Depth:	5%		Envelope:	60%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	-		0.7s	max	100%	0.11s	
Tune:	-1 octave		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0.03s	max	100%	0.11s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

KEYBOARDS

Accordion

Oscillator 1			LFO			Low-pass filter			
Wave:	PW=10%		Routing:	Amplitude			24db	12db	
Tune:	-		Wave:	Triangle		Cutoff:	8.2khz *87%*	6.0khz *82%*	
			Frequency:	13hz *fast*		Resonance:	0%		
Mix:	0db *100%*		Depth:	5%		Envelope:	0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	PW=10%		time:	-		-	-	-	-
Tune:	+15 cents		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0.18s	max	100%	0.18s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

Celeste

Oscillator 1			LFO			Low-pass filter			
Wave:	Triangle		Routing:	-			24db	12db	
Tune:	-		Wave:	-		Cutoff:	1.2khz *59%*	500hz *46%*	
			Frequency:	-		Resonance:	-		
Mix:	0db *100%*		Depth:	-		Envelope:	100%		
Oscillator 2			Glide:	-		**A**	**D**	**S**	**R**
Wave:	-		time:	-		0s	0.07s	45%	2.10s
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	all		**A**	**D**	**S**	**R**
Mix:	-					0s	2.10s	0%	2.10s
Osc 2 Track:	-								
Osc 2 Sync:	-								
Noise:	off								
Mix:	-								

note: If using a square wave instead of a triangle wave for oscillator 1 then reduce the filter frequency by half

Clavichord

Oscillator 1		LFO		Low-pass filter				
Wave:	Sawtooth	Routing:	Pitch		**24db**	**12db**		
Tune:	-	Wave:	Triangle	Cutoff:	1.8khz *65%*	1khz *57%*		
		Frequency:	1hz *very slow*	Resonance:	0%			
Mix:	0db 100%	Depth:	15 cent *shallow vibrato*	Envelope:	70%			
Oscillator 2		Glide:	-	**A**	**D**	**S**	**R**	
Wave:	Sawtooth	time:	-	0s	0.10s	65%	max	
Tune:	+1 octave	Unison:	off	**Amplifier**				
	+4 semi	Voices:	multi	**A**	**D**	**S**	**R**	
Mix:	0db 100%			0s	1.9s	0%	0.35s	
Osc 2 Track:	on							
Osc 2 Sync:	**on**							
Noise:	off							
Mix:	-							

Electric Piano

Oscillator 1		LFO		Low-pass filter				
Wave:	Sawtooth	Routing:	-		**24db**	**12db**		
Tune:	-	Wave:	-	Cutoff:	630hz *50%*	275hz *38%*		
		Frequency:	-	Resonance:	0%			
Mix:	0db *100%*	Depth:	-	Envelope:	50%			
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**	
Wave:	PW=5%	time:	-	0s	0.33s	50%	max	
Tune:	+2 octaves	Unison:	off	**Amplifier**				
	+9 semi	Voices:	multi	**A**	**D**	**S**	**R**	
Mix:	0db *100%*			0s	5.14s	0%	0.66s	
Osc 2 Track:	on							
Osc 2 Sync:	**on**							
Noise:	off							
Mix:	-							

Harpsichord

Oscillator 1			LFO			Low-pass filter				
Wave:	PW=25%		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		2.4khz *69%*	2.4khz *69%*	
			Frequency:	-		Resonance:		0%		
Mix:	-3db *95%*		Depth:	-		Envelope:		100%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW=5%		time:	-		0s	0.23s	65%	max	
Tune:	+9 semi		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	2s	0%	0.35s	
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

Organ

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		20khz *100%*	20khz *100%*	
			Frequency:	-		Resonance:		0%		
Mix:	0db 100%		Depth:	-		Envelope:		0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	-		-	-	-	-	
Tune:	-2 octaves		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db 100%					0.6s	max	100%	0.40s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Piano

Oscillator 1				
Wave:	Sawtooth			
Tune:	-			
Mix:	-10.4db 75%			

Oscillator 2				
Wave:	PW=15%			
Tune:	+1 octave			
	+2 semi			
Mix:	0db 100%			
Osc 2 Track:	on			
Osc 2 Sync:	**on**			

Noise:	off
Mix:	-

LFO	
Routing:	-
Wave:	-
Frequency:	-
Depth:	-
Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter			
		24db	12db
Cutoff:		40hz 10%	40hz 10%
Resonance:		0%	
Envelope:		75%	
A	D	S	R
0s	5.22s	0%	max

Amplifier			
A	D	S	R
0s	0.67s	25%	0.50s

Alternate: Osc 1: 0db (100%), Osc 2: sawtooth, +1 octave, 0db (100%), sync off

VOCALS

Angels

Oscillator 1		LFO		Low-pass filter			
Wave:	Sawtooth	Routing:	Pitch		**24db**	**12db**	
Tune:	-	Wave:	Triangle	Cutoff:	900hz *55%*	900hz *55%*	
		Frequency:	2.4hz *slow*	Resonance:	70%		
Mix:	0db *100%*	Depth:	20 cent *shallow*	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	-	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-			0.32s	max	100%	0.93s
Osc 2 Track:	-						
Osc 2 Sync:	-						
Noise:	off						
Mix:	-						

notes: C4-C6, 24 dB filter works best

Choir

Oscillator 1		LFO		Low-pass filter			
Wave:	PW=15%	Routing:	Pitch		**24db**	**12db**	
Tune:	-2 octaves	Wave:	Triangle	Cutoff:	1.7khz *64%*	1.0khz *56%*	
		Frequency:	2.4hz *slow*	Resonance:	0%		
Mix:	0db *100%*	Depth:	20 cent *shallow*	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	PW=25%	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	0db *100%*			0.32s	max	100%	0.93s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Vocal Female

Oscillator 1		LFO		Low-pass filter			
Wave:	PW=5%	Routing:	Pitch		24db	12db	
Tune:	-	Wave:	Triangle	Cutoff:	1.2khz 59%	1.2khz 59%	
		Frequency:	2.4hz slow	Resonance:	50%		
Mix:	0db 100%	Depth:	20 cent shallow	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	PW=25%	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-13.9db 65%			0.32s	max	100%	0.93s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

notes: 1. C4-C6   2. 24db filter is best to use

Vocal Male

Oscillator 1		LFO		Low-pass filter			
Wave:	PW=15%	Routing:	Pitch		24db	12db	
Tune:	-1 octave	Wave:	Triangle	Cutoff:	2.0khz 67%	1.2khz 59%	
		Frequency:	2.4hz slow	Resonance:	0%		
Mix:	-0db 100%	Depth:	20 cent shallow	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	PW=25%	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-4.3db 90%			0.32s	max	100%	0.93s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Alternate: Osc 1: sawtooth, 0db (100%), Osc 2: sawtooth, 0db (100%), Filter: 24db 900hz (55%), 12db 600hz (49%), resonance 70%

note: 12 dB filter works best

Whistling

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	Pitch				24db	12db	
Tune:	+2 octaves		Wave:	Triangle		Cutoff:	500hz *46%*	200hz *33%*		
			Frequency:	7hz *moderate*		Resonance:	0%			
Mix:	0db *100%*		Depth:	12 cents *shallow*		Envelope:	40%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	-		time:	-		0.04s	max	100%	max	
Tune:	-		Unison:	off		Amplifier				
			Voices:	2		**A**	**D**	**S**	**R**	
Mix:	-					0.03s	max	100%	0.35s	
Osc 2 Track:	-									
Osc 2 Sync:	-									
Noise:	off									
Mix:	-									

notes: 1. Lower the cutoff frequency when playing lower pitches to avoid woodwind characteristics 2. When using a square wave instead of a triangle wave for oscillator 1 lower the filter cutoff frequency to 24db; 0.3khz or 12db;0.1khz

TUNED PERCUSSION

Bell

Oscillator 1		LFO		Low-pass filter				
Wave:	Triangle	Routing:	Amplitude			24db	12db	
Tune:	-	Wave:	Triangle	Cutoff:	20khz *100%*	20hz *100%*		
		Frequency:	7.5hz *moderate*	Resonance:	0%			
Mix:	0db *100%*	Depth:	*5%*	Envelope:	0%			
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**	
Wave:	Square	time:	-	-	-	-	-	
Tune:	+5 semi	Unison:	off	**Amplifier**				
		Voices:	multi	**A**	**D**	**S**	**R**	
Mix:	-14db *65%*			0s	1.30s	0%	3.30s	
Osc 2 Track:	on							
Osc 2 Sync:	off							
Noise:	off							
Mix:	-							

Bongos

Oscillator 1		LFO		Low-pass filter				
Wave:	Triangle	Routing:	-			24db	12db	
Tune:	-	Wave:	-	Cutoff:	600hz *49%*	400hz *43%*		
		Frequency:	-	Resonance:	0%			
Mix:	0db *100%*	Depth:	-	Envelope:	60%			
Oscillator 2		Glide:	-	**A**	**D**	**S**	**R**	
Wave:	Square	time:	-	0s	0.11s	0%	0.11s	
Tune:	-	Unison:	off	**Amplifier**				
		Voices:	2	**A**	**D**	**S**	**R**	
Mix:	-13.9db *65%*			0s	0.22s	0%	0.22s	
Osc 2 Track:	on							
Osc 2 Sync:	off							
Noise:	off	*notes: G2-G4*						
Mix:	-							

Conga

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	-			24db	12db		
Tune:	-		Wave:	-		Cutoff:	300hz *39%*	150hz *29%*		
			Frequency:	-		Resonance:	40%			
Mix:	0db *100%*		Depth:	-		Envelope:	50%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		0s	0.02s	80%	max	
Tune:	+1 octave		Unison:	off		**Amplifier**				
	+2 semi		Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-6.5db *85%*					0s	0.15s	0%	0.15s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Glockenspiel

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	-			24db	12db		
Tune:	-		Wave:	-		Cutoff:	3.8khz *76%*	1.9khz *66%*		
			Frequency:	-		Resonance:	-			
Mix:	0db *100%*		Depth:	-		Envelope:	75%			
Oscillator 2			Glide:	-		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	-		0s	0.10s	25%	0.10s	
Tune:	+2 octave		Unison:	off		**Amplifier**				
	+6 semi		Voices:	all		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	1.20s	0%	1.20s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

note: Play above G5

Marimba

Oscillator 1		LFO		Low-pass filter			
Wave:	Triangle	Routing:	Amplitude			24db	12db
Tune:	-	Wave:	Triangle	Cutoff:		2.9khz 72%	1.9khz 66%
		Frequency:	8hz *fast*	Resonance:		0%	
Mix:	0db *100%*	Depth:	11%	Envelope:		0%	
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	Triangle	time:	-	-	-	-	-
Tune:	+2 octaves	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-9.6db *75%*			0s	1.66s	20%	1.66s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Timpani

Oscillator 1		LFO		Low-pass filter			
Wave:	Square	Routing:	-			24db	12db
Tune:	-1 octave	Wave:	-	Cutoff:		800hz 53%	500hz 46%
		Frequency:	-	Resonance:		45%	
Mix:	0db *100%*	Depth:	-	Envelope:		35%	
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	Square	time:	-	0	0.35s	0%	0.38s
Tune:	-1 octave	Unison:	off	**Amplifier**			
	-10 semi	Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-15.3db *60%*			0s	2.89s	0%	1.90s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Xylophone

Oscillator 1	
Wave:	Triangle
Tune:	+1 octave
	+10 semi
Mix:	0db 100%

Oscillator 2	
Wave:	Triangle
Tune:	-
Mix:	0db 100%
Osc 2 Track:	on
Osc 2 Sync:	off

Noise:	off
Mix:	-

LFO	
Routing:	Amplitude
Wave:	Triangle
Frequency:	7.5hz *moderate*
Depth:	5%

Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter			
	24db	12db	
Cutoff:	300hz *39%*	100hz *23%*	
Resonance:	-		
Envelope:	60%		

A	D	S	R
0s	0.53s	45%	0.50s

Amplifier			
A	D	S	R
0s	1.88s	0%	1.78s

UNTUNED PERCUSSION

Bass Drum

Oscillator 1		LFO		Low-pass filter				
Wave:	Triangle	Routing:	-			24db	12db	
Tune:	-2 oct	Wave:	-	Cutoff:	20khz *100%*	20khz *100%*		
		Frequency:	-	Resonance:	0%			
Mix:	0db *100%*	Depth:	-	Envelope:	0%			
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**	
Wave:	Triangle	time:	-	-	-	-	-	
Tune:	-2 oct	Unison:	off	**Amplifier**				
		Voices:	multi	**A**	**D**	**S**	**R**	
Mix:	0db *100%*			0s	0.12s	0%	0.12s	
Osc 2 Track:	on							
Osc 2 Sync:	off							
Noise:	off							
Mix:	-							

Castanets

Oscillator 1		LFO		Low-pass filter				
Wave:	-	Routing:	-			24db	12db	
Tune:	-	Wave:	-	Cutoff:	7.6khz *86%*	7.6khz *86%*		
		Frequency:	-	Resonance:	100%			
Mix:	-	Depth:	-	Envelope:	-			
Oscillator 2		Glide:	-	**A**	**D**	**S**	**R**	
Wave:	-	time:	-	-	-	-	-	
Tune:	-	Unison:	off	**Amplifier**				
		Voices:	1	**A**	**D**	**S**	**R**	
Mix:	-			0s	0.04s	0%	0.04s	
Osc 2 Track:	-							
Osc 2 Sync:	off							
Noise:	**on**							
Mix:	0db *100%*							

Clap

Oscillator 1			LFO			Low-pass filter			
Wave:	-		Routing:	Amplitude			**24db**	**12db**	
Tune:	-		Wave:	Square		Cutoff:	1.6khz *63%*	1.1khz *58%*	
			Frequency:	32hz *very fast*		Resonance:	30%		
Mix:	-		Depth:	45%		Envelope:	50%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	-		time:	-		0s	0.15s	0%	0.15
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	-					0s	0.11s	0%	0.11s
Osc 2 Track:	-								
Osc 2 Sync:	-								
Noise:	**on**								
Mix:	0db *100%*								

notes:1. 12 db filter works best 2. Set LFO to retrigger. If LFO can't retrigger then it may be best not to use since it will cause each hit to have a different amplitude

Claves

Oscillator 1			LFO			Low-pass filter			
Wave:	-		Routing:	-			**24db**	**12db**	
Tune:	-		Wave:	-		Cutoff:	2.1khz *67%*	2.1khz *67%*	
			Frequency:	-		Resonance:	100%		
Mix:	-		Depth:	-		Envelope:	0%		
Oscillator 2			Glide:	-		**A**	**D**	**S**	**R**
Wave:	-		time:	-		-	-	-	-
Tune:	-		Unison:	off		**Amplifier**			
			Voices:	mono		**A**	**D**	**S**	**R**
Mix:	-					0s	0.14s	0%	0.14s
Osc 2 Track:	-								
Osc 2 Sync:	off								
Noise:	**on**								
Mix:	0db *100%*								

note: 1. Change filter cutoff amount to alter pitch. 2. Using 12db filter allows full resonance to be added without significant reduction of higher frequencies.

Cowbell

Oscillator 1	
Wave:	Square
Tune:	-
Mix:	0db *100%*
Oscillator 2	
Wave:	PW=35%
Tune:	+1 octave
	+2 semi
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	off
Noise:	off
Mix:	-

LFO	
Routing:	-
Wave:	-
Frequency:	-
Depth:	-
Glide:	off
time:	-
Unison:	off
Voices:	mono

Low-pass filter	24db	12db
Cutoff:	8.8khz *88%*	8.8khz *88%*
Resonance:	55%	
Envelope:	65%	

A	D	S	R
0s	0.02s	65%	max

Amplifier			
A	D	S	R
0s	0.15s	0%	0.15s

note: Play around the fifth octave

Cowbell (analog drum machine)

Oscillator 1	
Wave:	PW = 10%
Tune:	-
Mix:	0db *100%*
Oscillator 2	
Wave:	PW = 35%
Tune:	+5 semi
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	off
Noise:	off
Mix:	-

LFO	
Routing:	-
Wave:	-
Frequency:	-
Depth:	-
Glide:	off
time:	-
Unison:	off
Voices:	mono

Low-pass filter	24db	12db
Cutoff:	8.1khz *87%*	3.4khz *74%*
Resonance:	0%	
Envelope:	65%	

A	D	S	R
0s	0.02s	65%	max

Amplifier			
A	D	S	R
0s	0.15s	0%	0.15s

note: Play around the fifth octave

Cymbal

Oscillator 1			LFO			Low-pass filter				
Wave:	-		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		9.4khz 89%	9.4khz 89%	
			Frequency:	-		Resonance:	50%			
Mix:	-		Depth:	-		Envelope:	70%			
Oscillator 2			Glide:	-		**A**	**D**	**S**	**R**	
Wave:	-		time:	-		0s	0.14s	0%	1.80s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-					0s	1.10s	0%	1.00s	
Osc 2 Track:	-									
Osc 2 Sync:	-									
Noise:	**on**									
Mix:	0db 100%									

note: Use EQ or a high-pass filtqer to remove low end frequencies and apply some distortion. Probably the hardest instrument to synthesize. This patch took the longest of any in the book to create (~ 4 hours) and yet is still the least realistic. At any rate it can fill the role of a cymbal or at least provide a good laugh and is included for completeness. I challenge anyone to do better using two oscillators, noise, and a single low-pass filter! To produce a cymbal, the TR-808 used six enharmonically, detuned square wave oscillators, split by three band-pass filters. Each band was sent to its own enveloped VCA before being sent to high-pass filters... oh hell, you get the idea! As early as the TR-909 the cymbal was sampled due to this complexity.

Side Stick

Oscillator 1			LFO			Low-pass filter				
Wave:	-		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		2.7khz 71%	2.7khz 71%	
			Frequency:	-		Resonance:	100%			
Mix:	-		Depth:	-		Envelope:	85%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	-		time:	-		0s	1.19s	0%	0s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-					0s	0.09s	0%	0s	
Osc 2 Track:	-									
Osc 2 Sync:	-									
Noise:	**on**									
Mix:	0db *100%*									

Snare Drum

Oscillator 1		LFO		Low-pass filter			
Wave:	Triangle	Routing:	-		**24db**	**12db**	
Tune:	-1 octave	Wave:	-	Cutoff:	20khz *100%*	20khz *100%*	
		Frequency:	-	Resonance:	-		
Mix:	0db *100%*	Depth:	-	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	-	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-			0s	0.27s	0%	0.27s
Osc 2 Track:	-						
Osc 2 Sync:	-						
Noise:	**on**						
Mix:	-31.4db *20%*						

Tambourine

Oscillator 1		LFO		Low-pass filter			
Wave:	-	Routing:	-		**24db**	**12db**	
Tune:	-	Wave:	-	Cutoff:	5.3khz *81%*	5.3khz *81%*	
		Frequency:	-	Resonance:	100%		
Mix:	-	Depth:	-	Envelope:	0%		
Oscillator 2		Glide:	-	**A**	**D**	**S**	**R**
Wave:	-	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	mono	**A**	**D**	**S**	**R**
Mix:	-			0s	0.34s	0%	0.34s
Osc 2 Track:	-						
Osc 2 Sync:	-						
Noise:	**on**						
Mix:	0db 100%						

note: Make sure that if the synthesizer's filter has keyboard tracking that it is turned off.

Wheels of Steel

Oscillator 1						
Wave:	-					
Tune:	-					
Mix:	-					
Oscillator 2						
Wave:	-					
Tune:	-					
Mix:	-					
Osc 2 Track:	-					
Osc 2 Sync:	-					
Noise:	**on**					
Mix:	0db 100%					

LFO	
Routing:	-
Wave:	-
Frequency:	-
Depth:	-
Glide:	-
time:	-
Unison:	off
Voices:	mono

Low-pass filter

	24db	12db
Cutoff:	40hz *10%*	40hz *10%*
Resonance:	40%	
Envelope:	100%	

A	D	S	R
0.02s	0.09s	0%	0.09s

Amplifier

A	D	S	R
0s	1.65s	0%	0.05s

LEADS

Brass Section

Oscillator 1		LFO		Low-pass filter				
Wave:	Square	Routing:	PW osc1, osc 2			24db	12db	
Tune:	-10 cents	Wave:	Triangle / Sine	Cutoff:	40hz *10%*	40hz *10%*	
		Frequency:	5.5 Hz	Resonance:	0%		
Mix:	0db *100%*	Depth:	45%	Envelope:	100%		

Oscillator 1		LFO		Low-pass filter				
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	PW = 20%	time:	-	0.03s	max	100%	0.60s
Tune:	+1 oct	Unison:	off	**Amplifier**			
	+10 cents	Voices:	multi	**A**	**D**	**S**	**R**
Mix:	0db *100%*			0s	max	100%	0.35s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Mellow 70's Lead

Oscillator 1		LFO		Low-pass filter				
Wave:	Square	Routing:	-			24db	12db	
Tune:	-10 cents	Wave:	-	Cutoff:	20khz *100%*	20khz *100%*	
		Frequency:	-	Resonance:	0%		
Mix:	0db *100%*	Depth:	-	Envelope:	0%		
Oscillator 2		Glide:	on	**A**	**D**	**S**	**R**
Wave:	Square	time:	0.02s	-	-	-	-
Tune:	+10 cents	Unison:	off	**Amplifier**			
		Voices:	mono	**A**	**D**	**S**	**R**
Mix:	0db *100%*			0s	max	100%	0.35s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Mono Solo

Oscillator 1	
Wave:	PW = 30%
Tune:	-
Mix:	0db *100%*
Oscillator 2	
Wave:	PW = 20%
Tune:	+4 cents
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	off
Noise:	off
Mix:	-

LFO	
Routing:	Pitch osc1, osc2
Wave:	Triangle / Sine
Frequency:	4.5 hz *moderate*
Depth:	22 cents *moderate*
Glide:	on
time:	0.04s *very fast*
Unison:	off
Voices:	multi

Low-pass filter		24db	12db
Cutoff:		40hz *10%*	40hz *10%*
Resonance:		45%	
Envelope:		80%	

A	D	S	
0s	0.20s	70%	

Amplifier			
A	D	S	
0s	max	100%	

New Age Lead

Oscillator 1	
Wave:	Square
Tune:	+5 cents
Mix:	0db *100%*
Oscillator 2	
Wave:	Square
Tune:	-5 cents
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	off
Noise:	off
Mix:	-

LFO	
Routing:	PW osc1, osc2
Wave:	Triangle / Sine
Frequency:	5.5 Hz
Depth:	45%
Glide:	off
time:	-
Unison:	off
Voices:	multi

Low-pass filter		24db	12db
Cutoff:		1.4khz *61%*	1.0khz *56%*
Resonance:		65%	
Envelope:		75%	

A	D	S	R
0s	1.20s	20%	0.60s

Amplifier			
A	D	S	R
0.05s	max	100%	0.35s

R&B Slide

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	-				24db	12db	
Tune:	+1 oct		Wave:	-		Cutoff:		1.9khz 66%	1.9khz 66%	
			Frequency:	-		Resonance:	45%			
Mix:	0db 100%		Depth:	-		Envelope:	65%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	0.02s		1.20s	max	100%	6.30s	
Tune:	+3 oct		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-6db 85%					0.02s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-		*note: Play in the higher octaves. Also try using light unison*							

Screaming Sync

Oscillator 1			LFO			Low-pass filter				
Wave:	Any waveform		Routing:	Pitch osc2				24db	12db	
Tune:	-1 oct		Wave:	Triangle / Sine		Cutoff:		20khz 100%	20khz 100%	
			Frequency:	0.66 Hz		Resonance:	0%			
Mix:	-40db 0%		Depth:	4.5cents *medium*		Envelope:	0%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	Square		time:	0.02s		-	-	-	-	
Tune:	-		Unison:	on		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db 100%					0s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-		*note: make sure oscillator 1 is on but its volume is all the way down*							

Strings Pulse-Width Modulation

Oscillator 1			LFO			Low-pass filter			
Wave:	Square		Routing:	PW osc1,osc2			24db	12db	
Tune:	-10 cents		Wave:	Sine		Cutoff:	2khz *67%*	2khz *67%*	
			Frequency:	2hz *slow*		Resonance:	0%		
Mix:	0db *100%*		Depth:	47%		Envelope:	100%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	Square		time:	-		0.09s	max	100%	max
Tune:	+10 cents		Unison:	off		**Amplifier**			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0.11s	max	100%	0.35s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-		notes: use thick chorus and delay						

Trance 5th

Oscillator 1			LFO			Low-pass filter			
Wave:	Square		Routing:	PW osc1, osc2			24db	12db	
Tune:	-		Wave:	Triangle / Sine		Cutoff:	20khz *100%*	20khz *100%*	
			Frequency:	6 Hz *moderate*		Resonance:	0%		
Mix:	0db *100%*		Depth:	80%		Envelope:	0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	Square		time:	-		-	-	-	-
Tune:	+7 semi		Unison:	on		**Amplifier**			
			Voices:	mono		**A**	**D**	**S**	**R**
Mix:	0db *100%*					0s	max	100%	0s
Osc 2 Track:	on								
Osc 2 Sync:	off								
Noise:	off								
Mix:	-								

BASS

Acid Bass

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 25%		Routing:	-				24db	12db	
Tune:	+10 cents		Wave:	-		Cutoff:	450hz *45%*	330hz *40%*		
			Frequency:	-		Resonance:	60%			
Mix:	-12db *70%*		Depth:	-		Envelope:	0%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		0s	-	-	-	
Tune:	-2 oct		Unison:	off		Amplifier				
	-10 cents		Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	0.45s	15%	0.26s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

note: turn filter cutoff knob during play for acid effect

Bass of the Time Lords

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 30%		Routing:	-				24db	12db	
Tune:	-2 oct		Wave:	-		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	-		Resonance:	0%			
Mix:	0db *100%*		Depth:	-		Envelope:	85%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW=30%		time:	-		0s	0.25s	25%	0.25s	
Tune:	-1 oct		Unison:	off		Amplifier				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	0.65s	0%	0.70s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Detroit Bass

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	-				24db	12db	
Tune:	-1 oct		Wave:	-		Cutoff:	20khz *100%*	20khz *100%*		
			Frequency:	-		Resonance:	0%			
Mix:	0db *100%*		Depth:	-		Envelope:	0%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	0.01s		-	-	-	-	
Tune:	-2 oct		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Deutsche Bass

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 25%		Routing:	-				24db	12db	
Tune:	-2 oct		Wave:	-		Cutoff:	800hz *53%*	500hz *46%*		
			Frequency:	-		Resonance:	35%			
Mix:	0db *100%*		Depth:	-		Envelope:	50%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	-		0s	0.15s	0%	0.15s	
Tune:	-1 oct		Unison:	off		**Amplifier**				
	-3 semi		Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.15s	
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

Digital Bass

Oscillator 1		LFO		Low-pass filter				
Wave:	Square	Routing:	-			**24db**	**12db**	
Tune:	-1 oct	Wave:	-	Cutoff:	122hz *26%*	75hz *19%*	
		Frequency:	-	Resonance:	0%		
Mix:	-6db *85%*	Depth:	-	Envelope:	100%		
Oscillator 2				**A**	**D**	**S**	**R**
Wave:	Sawtooth	Glide:	off	0s	0.15s	0%	0s
Tune:	-2 oct	time:	-	**Amplifier**			
		Unison:	off	**A**	**D**	**S**	**R**
Mix:	0db *100%*	Voices:	mono	0s	max	100%	1.00s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Funk Bass

Oscillator 1		LFO		Low-pass filter				
Wave:	Sawtooth	Routing:	-			**24db**	**12db**	
Tune:	-2 oct	Wave:	-	Cutoff:	40hz *10%*	40hz *10%*	
		Frequency:	-	Resonance:	50%		
Mix:	0db *100%*	Depth:	-	Envelope:	85%		
Oscillator 2				**A**	**D**	**S**	**R**
Wave:	PW = 30%	Glide:	on	0.15s	0.10s	45%	0.08s
Tune:	-1 oct	time:	0.02s	**Amplifier**			
		Unison:	off	**A**	**D**	**S**	**R**
Mix:	0db *100%*	Voices:	mono	0s	0.29s	30%	0.40s
Osc 2 Track:	on						
Osc 2 Sync:	off						
Noise:	off						
Mix:	-						

Growling Bass

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	-				24db	12db	
Tune:	-2 oct		Wave:	-		Cutoff:		122hz *26%*	75hz *19%*	
			Frequency:	-		Resonance:		0%		
Mix:	0db *100%*		Depth:	-		Envelope:		100%		
Oscillator 2			Glide:	Off		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	-		0s	0.50s	20%	0.60s	
Tune:	-9 semi		Unison:	Off		**Amplifier**				
			Voices:	Mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Rez Bass

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	Pitch osc2				24db	12db	
Tune:	-2 oct		Wave:	Triangle / Sine		Cutoff:		6.5khz *84%*	6.5khz *84%*	
			Frequency:	10 hz *fast*		Resonance:		50%		
Mix:	0db *100%*		Depth:	30 cents *shallow*		Envelope:		0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	0.08s		-	-	-	-	
Tune:	-2 oct		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.10s	
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

PADS

Android Dreams

Oscillator 1			LFO			Low-pass filter			
Wave:	Square		Routing:	*see note			24db	12db	
Tune:	-2 oct		Wave:	Triangle / Sine		Cutoff:	2.4khz 69%	1.7khz 64%	
			Frequency:	0.25 hz *very slow*		Resonance:	30%		
Mix:	0db 100%		Depth:	*see note		Envelope:	0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	Sawtooth		time:	-		-	-	-	-
Tune:	-1 semi		Unison:	off		Amplifier			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	0db 100%					0.45s	max	100%	2.25s
Osc 2 Track:	on								
Osc 2 Sync:	**on**								
Noise:	**on**								
Mix:	-6 db 90%								

note: multiple LFO routings with their respective depths: PW osc1&2 45%, filter cutoff 10%, pitch osc2 3-semitones

Celestial Wash

Oscillator 1			LFO			Low-pass filter			
Wave:	-		Routing:	Resonance			24db	12db	
Tune:	-		Wave:	Square		Cutoff:	1khz 57%	500hz 46%	
			Frequency:	5.5 hz *moderate*		Resonance:	100%		
Mix:	-		Depth:	100%		Envelope:	0%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**
Wave:	-		time:	-		-	-	-	-
Tune:	-		Unison:	off		Amplifier			
			Voices:	multi		**A**	**D**	**S**	**R**
Mix:	-					1.65s	5.85s	0%	4.40s
Osc 2 Track:	-								
Osc 2 Sync:	-								
Noise:	**on**								
Mix:	0 dB 100%								

note: turn filter's keyboard tracking on if available

Dark City

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	PW osc1, osc2				24db	12db	
Tune:	-2 oct		Wave:	Triangle / Sine		Cutoff:		300hz *39%*	200hz *33%*	
			Frequency:	4 Hz *moderate*		Resonance:		0%		
Mix:	0 db *100%*		Depth:	45%		Envelope:		75%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	-		5.15s	6.35s	50%	5.20s	
Tune:	-11 semi		Unison:	off		**Amplifier**				
	-92 cents		Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0 db *100%*					0s	3.90s	70%	6.35s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Aurora

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	PW osc1, osc2				24db	12db	
Tune:	+10 cents		Wave:	Triangle / Sine		Cutoff:		40hz *10%*	40hz *10%*	
			Frequency:	4 Hz *moderate*		Resonance:		65%		
Mix:	0db *100%*		Depth:	40%		Envelope:		50%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		1.50s	max	100%	0.60s	
Tune:	-10 cents		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0.60s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Galactic Cathedral

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	Filter cutoff				24db	12db	
Tune:	-11 cents		Wave:	Triangle / Sine		Cutoff:		1.2khz *59%*	700hz *51%*	
			Frequency:	0.3 hz *very slow*		Resonance:		60%		
Mix:	0 db *100%*		Depth:	10%		Envelope:		0%		
Oscillator 2			Glide:	off		**A**		**D**	**S**	**R**
Wave:	Square		time:	-		-		-	-	-
Tune:	+2 oct		Unison:	off		**Amplifier**				
	+11 cents		Voices:	multi		**A**		**D**	**S**	**R**
Mix:	-6 db *85%*					0.40s		max	100%	2.25s
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Galactic Chapel

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	Cutoff & Amp				24db	12db	
Tune:	-		Wave:	Triangle / Sine		Cutoff:		1.4khz *61%*	1khz *57%*	
			Frequency:	0.3 hz *very slow*		Resonance:		65%		
Mix:	0 db *100%*		Depth:	cutoff 10%, amp 17%		Envelope:		0%		
Oscillator 2			Glide:	off		**A**		**D**	**S**	**R**
Wave:	Triangle		time:	-		-		-	-	-
Tune:	+1 oct		Unison:	off		**Amplifier**				
			Voices:	multi		**A**		**D**	**S**	**R**
Mix:	-6 db *85%*					0.35s		max	100%	2.25s
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Portus

Oscillator 1			LFO			Low-pass filter				
Wave:	Triangle		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:		75hz *19%*	75hz *19%*	
			Frequency:	-		Resonance:		70%		
Mix:	0db *100%*		Depth:	-		Envelope:		70%		
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	Triangle		time:	1.5s		1.80s	max	100%	2.90s	
Tune:	+1 oct		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					1.50s	max	100%	3.30s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Post-Apocalyptic Sync Sweep

Oscillator 1			LFO			Low-pass filter				
Wave:	Square		Routing:	Pitch osc2				24db	12db	
Tune:	-2 oct		Wave:	Triangle / Sine		Cutoff:		40hz *10%*	40hz *10%*	
			Frequency:	0.7 hz *very slow*		Resonance:		65%		
Mix:	0db *100%*		Depth:	4.5 semi *moderate*		Envelope:		50%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		2.10s	5.20s	0%	3.00s	
Tune:	-1 oct		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	-6db *85%*					0s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	**on**									
Noise:	off									
Mix:	-									

Terra Enceladus

Oscillator 1			LFO			Low-pass filter				
Wave:	Sawtooth		Routing:	PW osc2				24db	12db	
Tune:	-		Wave:	Triangle / Sine		Cutoff:	1.3khz *60%*	600hz *49%*		
			Frequency:	1.75hz *slow*		Resonance:	0%			
Mix:	0db *100%*		Depth:	45%		Envelope:	0%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		-	-	-	-	
Tune:	-1 oct		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	-3db *90%*					0.40s	max	100%	2.25s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

SOUND EFFECTS

Cat

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 10%		Routing:	Pitch				24db	12db	
Tune:	+1 octave		Wave:	Sine Wave		Cutoff:		270hz *37%*	160hz *30%*	
	+5 semi		Frequency:	1.7hz *slow*		Resonance:		65%		
Mix:	0db *100%*		Depth:	11 cents		Envelope:		65%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW = 5%		time:	-		0.57s	1.90s	55%	0.29s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0.51s	1.05s	55%	0.93s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Digital Alarm Clock

Oscillator 1			LFO			Low-pass filter				
Wave:	PW = 5%		Routing:	Amplitude				24db	12db	
Tune:	-		Wave:	square		Cutoff:		9.5khz *89%*	5.7khz *82%*	
			Frequency:	2hz *slow*		Resonance:		0%		
Mix:	0db *100%*		Depth:	100%		Envelope:		40%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Sawtooth		time:	-		-	-	-	-	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db 100%					0s	max	100%	0s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

Journey to the Core

Oscillator 1			LFO			Low-pass filter				
Wave:	Sawtooth		Routing:	*see note				24db	12db	
Tune:	-2 oct		Wave:	Sawtooth		Cutoff:		120hz 26%	120hz 26%	
			Frequency:	7.5hz *moderate*		Resonance:		45%		
Mix:	0db *100%*		Depth:	*see note		Envelope:		100%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	Square		time:	-		3.30s	3.20s	0%	3.20s	
Tune:	-1 oct		Unison:	off		**Amplifier**				
			Voices:	multi		**A**	**D**	**S**	**R**	
Mix:	0db *100%*					0s	max	100%	0.80s	
Osc 2 Track:	on									
Osc 2 Sync:	-									
Noise:	off									
Mix:	-									

note: LFO routing and depth – Pitch osc1, osc2 = 7 semitones; PW osc2 = 100%

Kazoo

Oscillator 1			LFO			Low-pass filter				
Wave:	PW=15%		Routing:	Pitch				24db	12db	
Tune:	-		Wave:	Noise or Saw		Cutoff:		1.2khz 59%	600hz 49%	
			Frequency:	32hz *very fast*		Resonance:		30%		
Mix:	0db 100%		Depth:	70 cents *deep*		Envelope:		50%		
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	PW=2%		time:	-		0.06s	1.20s	0%	0.63s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	0db 100%					0.03s	3.9s	0%	0.33s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

note: If available, use noise for the LFO wave source otherwise use a sawtooth

Laser

Oscillator 1			LFO			Low-pass filter				
Wave:	-		Routing:	-				24db	12db	
Tune:	-		Wave:	-		Cutoff:	40hz *10%*	40hz *10%*		
			Frequency:	-		Resonance:	100%			
Mix:	-		Depth:	-		Envelope:	100%			
Oscillator 2			Glide:	off		**A**	**D**	**S**	**R**	
Wave:	-		time:	-		0s	0.38s	0%	0.25s	
Tune:	-		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-					0s	0.45s	0%	0.30s	
Osc 2 Track:	-									
Osc 2 Sync:	-									
Noise:	**on**									
Mix:	0 db *100%*									

Motor

Oscillator 1			LFO			Low-pass filter				
Wave:	PW=25%		Routing:	Amplitude				24db	12db	
Tune:	-2 octaves		Wave:	Square		Cutoff:	20khz *100%*	20khz *100%*		
			Frequency:	32hz *very fast*		Resonance:	0%			
Mix:	0db *100%*		Depth:	40%		Envelope:	0%			
Oscillator 2			Glide:	on		**A**	**D**	**S**	**R**	
Wave:	PW=25%		time:	1.3s *long*		-	-	-	-	
Tune:	-1 octave		Unison:	off		**Amplifier**				
			Voices:	mono		**A**	**D**	**S**	**R**	
Mix:	-14.3db *65%*					0s	max	100%	0.60s	
Osc 2 Track:	on									
Osc 2 Sync:	off									
Noise:	off									
Mix:	-									

note: having the glide on is very important for this patch to sound right!

Nerd-O-Tron 2000

Oscillator 1		LFO		Low-pass filter			
Wave:	Sawtooth	Routing:	Filter cutoff		**24db**	**12db**	
Tune:	-	Wave:	Noise	Cutoff:	680hz *51%*	680hz *51%*	
		Frequency:	0.9hz *very slow*	Resonance:	85%		
Mix:	0db *100%*	Depth:	25%	Envelope:	0%		
Oscillator 2		Glide:	off	**A**	**D**	**S**	**R**
Wave:	Square	time:	-	-	-	-	-
Tune:	+2 octaves	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	0db *100%*			0s	max	100%	2.00s
Osc 2 Track:	on						
Osc 2 Sync:	-						
Noise:	off						
Mix:	-						

Ocean Waves (with foghorn)

Oscillator 1		LFO		Low-pass filter			
Wave:	-	Routing:	Amp osc1, noise		**24db**	**12db**	
Tune:	-	Wave:	Triangle	Cutoff:	4.3khz *78%*	2.0khz *67%*	
		Frequency:	0.2hz *very slow*	Resonance:	30%		
Mix:	-	Depth:	75%	Envelope:	0%		
Oscillator 2		Glide:	Off	**A**	**D**	**S**	**R**
Wave:	-	time:	-	-	-	-	-
Tune:	-	Unison:	off	**Amplifier**			
		Voices:	multi	**A**	**D**	**S**	**R**
Mix:	-			0.40s	max	100%	1.00s
Osc 2 Track:	-						
Osc 2 Sync:	-						
Noise:	**on**						
Mix:	0 db *100%*						

note: for foghorn set oscillator 1 to Wave: Sawtooth, Tune: -1 oct, Mix: 0 db 100%

...sitronic Rhythm

Oscillator 1		LFO		Low-pass filter		
Wave:	Sawtooth	Routing:	*see note		**24db**	**12db**
Tune:	-2 oct	Wave:	Square	Cutoff:	165hz *30%*	165hz *30%*
		Frequency:	4 hz *moderate*	Resonance:	45%	
Mix:	0 db *100%*	Depth:	*see note	Envelope:	100%	
Oscillator 2		Glide:	off	**A**	**D**	**S** / **R**
Wave:	Square	time:	-	3.30s	4.00s	0% / 2.90s
Tune:	-1 oct	Unison:	off	**Amplifier**		
		Voices:	multi	**A**	**D**	**S** / **R**
Mix:	0 db *100%*			0s	max	100% / 0.80s
Osc 2 Track:	on					
Osc 2 Sync:	off					
Noise:	off					
Mix:	-					

note: LFO routing and depth – Pitch osc1, osc2 = 7 semitones; PW osc2 = 100%

Space Attack!

Oscillator 1		LFO		Low-pass filter		
Wave:	Square	Routing:	Pitch osc1, osc2		**24db**	**12db**
Tune:	-1 oct	Wave:	Sawtooth	Cutoff:	20khz *100%*	20khz *100%*
		Frequency:	2.5hz moderate	Resonance:	0%	
Mix:	0 db *100%*	Depth:	1 oct *deep*	Envelope:	0%	
Oscillator 2		Glide:	Off	**A**	**D**	**S** / **R**
Wave:	Sawtooth	time:	-	-	-	- / -
Tune:	-1 oct	Unison:	off	**Amplifier**		
		Voices:	mono	**A**	**D**	**S** / **R**
Mix:	0 db *100%*			0s	max	100% / 2.00s
Osc 2 Track:	on					
Osc 2 Sync:	off					
Noise:	off					
Mix:	-					

note: if 2nd LFO is available set it up as Routing: Amplitude osc1, osc2,
Wave: Sawtooth, Frequency: 15 hz (fast), Depth: 100%

Toad

Oscillator 1	
Wave:	Triangle
Tune:	+1 octave
Mix:	0db *100%*

Oscillator 2	
Wave:	Square
Tune:	-2 octaves
Mix:	0db *100%*
Osc 2 Track:	on
Osc 2 Sync:	-

Noise:	off
Mix:	-

LFO	
Routing:	Amplitude
Wave:	Triangle
Frequency:	32hz *very fast*
Depth:	100%

Glide:	on
time:	1.3s *long*
Unison:	off
Voices:	multi

Low-pass filter

	24db	12db		
Cutoff:	40hz *10%*	40hz *10%*		
Resonance:	0%			
Envelope:	0%			
A	**D**	**S**	**R**	
-	-	-	-	

Amplifier			
A	**D**	**S**	**R**
0.09s	0.23s	0%	0.23s

Wind

Oscillator 1	
Wave:	-
Tune:	-
Mix:	-

Oscillator 2	
Wave:	-
Tune:	-
Mix:	-
Osc 2 Track:	-
Osc 2 Sync:	-

Noise:	**on**
Mix:	0 db *100%*

LFO	
Routing:	Noise amplitude
Wave:	Noise
Frequency:	0.7hz *very slow*
Depth:	40%

Glide:	Off
time:	-
Unison:	off
Voices:	multi

Low-pass filter

	24db	12db		
Cutoff:	780hz *53%*	780hz *53%*		
Resonance:	75%			
Envelope:	0%			
A	**D**	**S**	**R**	
-	-	-	-	

Amplifier			
A	**D**	**S**	**R**
0.40s	max	100%	2.70s

CALIBRATION

This section is intended for those synthesizers that do not give the values of their settings in decibels, hertz, or seconds. By first calibrating such a synthesizer the patches in this book can be programmed much more accurately than using the suggested percentage settings that are given in italics. It is advisable to at the very least calibrate the oscillator mix and filter cutoff frequency. There are two parts to this section. The first part explains how to calibrate a synth using the sound files on the CD and the second part explains how to using a sound editor and other metering devices. Using the CD is probably the quickest way to get started while using the directions in the second part will produce slightly more accurate results.

The most optimal way to use these two sections is probably to use the CD to calibrate the filter's cutoff frequency and if you can record your synth into a sound editor then use Part B to determine oscillator mix levels, LFO frequency and ADSR values. If you would like more clarification on any part of this section please feel free to email me at *fred_v_welsh@yahoo.com*

Part A - Using the CD

Using the sounds files on the CD is quite simple. In a nutshell you are basically just comparing the output of your synth to each file and figuring out values such as decibels, Hz, and seconds from the filename of the sound that most closely matches what you hear from your synth.

Oscillator Mix:
A.   Get a piece of paper and a pencil or pen.
B.   Go into the *Oscillator Mix* folder on the CD. Although the filter isn't being calibrated at this point it is still important to use samples that correspond to the synth's filter type. Open the folder that corresponds to the filter type used by your synth. If both filter types are available then you can just pick one but of course make sure the synth is set to that filter.
C.   Now you should see two folders. We are going to use the *Dual Sawtooth Waveforms* folder. The *Dual Sawtooth Waveforms* folder contains samples where both oscillators are set to sawtooth and one is raised two octaves. Set your synthesizer up

with both oscillators set to sawtooth, one raised two octaves. The purpose of this is so that the amplitude of one oscillator can be compared to the other

D.   In each one of these samples the high-pitch oscillator is left at its maximum setting. This maximum setting corresponds to 0 dB. The name of each file indicates the amplitude of the low-pitch oscillator. On your synth, turn the amplitude of both sawtooths all the way up. It should now produce the same tonal characteristics as the 0 dB file when you play middle C on the keyboard. Also make sure your filter is completely open and that amplitude velocity sensitivity is off.

E.   Write down the number, dial position, or whatever your synth uses that corresponds to the maximum setting and next to it write down "0 dB." Make sure to write neatly because you are going to be using these values to patch the sounds in this book.

F.   Listen to the tonal characteristics of the -5 dB file. Now lower the amplitude of your low-pitch oscillator until it produces the same tonal characteristics as the -5 dB file. Write down the position followed by "-5 dB."

G.   Continue to do the same thing for each of the remaining sound files

The "Single Sawtooth Waveform" folder contains samples of a single sawtooth oscillator which just like in the instructions above is lowered in 5 dB increments. This is just an alternative to the Dual Sawtooth instructions above and is carried out in the exact same way.

Filter

Determining Filter Type

If you already know whether your filter is of the 12 dB or 24 dB variety then you can skip this section and proceed to the *Determining Filter Cutoff* section.

A.   Open the "*Filter*" folder and then open the "*Determine Filter Type*" folder. There are four files in here. For each set (12 dB or 24 dB) there is a file where the filter resonance is set to a pitch of 440 Hz and after about ten seconds the resonance is turned all the way down. There are also two files where the filter is completely open at 20,000 Hz and resonance is set to zero. Play both of these files and you'll notice that the 12 dB filter produces a slightly brighter sound. That's because a filter that is "open" isn't completely open! It will still have some effect on the tone.

B. Set your synth up so that it produces a single sawtooth waveform. Turn your filter resonance all the way up. If it distorts then turn it down some.

C. While playing A440 (the A below mid-C) on your keyboard, lower the filter cutoff frequency until it matches the resonant pitch at the beginning of either one of these sound files.

D. Turn the resonance all the way down so that you only hear the filtered sawtooth. Compare the tonal characteristics that you hear to those in the second half of each of the 12 dB and 24 db *"filter at 440 Hz"* files.

E. Whichever file sounds the same indicates whether you have a 12 dB or 24 dB filter.

Determining Filter Cutoff

A. Open the *"Filter"* folder and then open the folder that corresponds with your synth's filter type, either the *"12db"* or *"24db"* folder.

B. Set the synth up to produce a single sawtooth waveform.

C. Turn the filter cutoff all the way up so that it is completely open.

D. Now play the 20,000 Hz file. Does it have the same tonal characteristics as your synth played at middle C? Make sure you are listening to your synth and the wave file at relatively the same amplitude. This is important because the human ear perceives amplitude changes differently for different frequencies. That's also why it is a good idea to not have the volume cranked during the final mix of a song...but I digress! If your synth doesn't sound as bright as the 20,000 Hz file then try the 16,000 Hz or the 12,000 Hz files. If your synth sounds brighter than the 20,000 Hz file it is probably only slightly and for all practical purposes you can label it as 20,000 Hz.

E. On a piece of paper write down the maximum setting value/dial position as well as the cutoff frequency that it corresponds to.

F. What we are going to do now is lower the filter cutoff setting in evenly spaced increments. Use whatever increments are convenient for the way your filter displays its setting. Anywhere from five to ten total values works fine.

G. For each setting find the file that sounds the closest in tone and write down both the dial setting and cutoff frequency.

CALIBRATION

LFO Frequency

A.   Open the *"LFO"* folder and you will see a bunch of sound files where the names correspond to the frequency of the recorded LFO.

B.   Set the synth up to produce a single sawtooth waveform. Set the LFO up to use a square wave as its source. Route the LFO to the amplitude of this oscillator and set the modulation amount to 100%.

C.   Set the LFO frequency setting to its minimum. You should of course not hear any change in the amplitude when you strike a key.

D.   Listen to the 0.5 Hz file. Notice how long it takes for the amplitude of the sawtooth to "pulse." Adjust your synth's LFO so that it pulses at the same rate. Write down the setting position/value along with the corresponding LFO frequency from the filename.

E.   Now match your LFO frequency to the pulsing of the 1 Hz sound file and again write down the position and frequency values.

F.   Work your way up the list of sound files matching the LFO frequency and writing the values down. You don't have to do this for every sound file, but I would do it for at least every other one.

ADSR VALUES

A.   In the *"ADSR"* folder there is an *"A"* folder and a *"D and R"*. Of course *"A"* stands for attack and *"D and R"* stand for decay and release. Remember that sustain is always set as a percentage of the maximum setting so it doesn't need to be calibrated.

B.   *Finding the attack time*: Open the *"A"* folder and you will see a series of sound files where each has a filename that indicates the length of the attack.

C.   Set all the ADSR values to their minimum. No sound should be produced when you strike a key.

D.   We are going to increase the attack setting in evenly spaced increments. Pick increment values that are convenient for the way your synth displays its values. Again, five to ten increments should work well. Be sure to hold down the keys long enough.

E.   At each increment compare your synths output to the sound files and once you've found a file that is a close match write down both the attack setting and the time listed in the name of the file. Repeat this for each increment until the attack setting is maxed out.

F. *Finding the decay time*: Follow the same procedure as outlined in steps B-E only this time raise the value of the decay.

G. *Finding the release time:* Same as steps B-E but this time also begin with sustain set to 100% and quickly strike the keys. Do not hold them down.

Part B - Using Sound Editors, Meters, and by Ear

Oscillator Mix:

A. The levels for the amplitude of each oscillator are given in terms of decibels. The maximum level is at 0 decibels with lower levels given in negative values. To determine your synthesizer's oscillator levels in terms of decibels you will need to monitor the synth's output using either a sound editor or a level meter that displays amplitude in decibels. A sound editor is the best choice since a recording can be made which can then be normalized.

B. Make sure the filter is completely open and that the amplitude is not being enveloped.

C. Begin by turning the amplitude of one oscillator up to its maximum setting and turning all other oscillators and noise sources off.

D. If you are using a sound editor start recording now.

E. The maximum setting of the level corresponds to 0 dB so it will be convenient if at this point the output from the synth can be displayed as 0 dB on whatever is being used to meter the signal (Here we are using decibels on a relative scale rather than a measure of actual audio output). If you are using something other than a sound editor to meter the output then use your synth's master volume, the channel in the host application or mixing board or computer's audio mixer to bring the synth's level to 0 dB on the meter while keeping the oscillators own level setting at the maximum setting. In other words don't manipulate the oscillator's level setting to bring the meter to 0 dB. If you are using a sound editor don't bother messing with the level on the meter since we are going to normalize the recording to 0 dB anyways. If you are not able to adjust the signal to 0 dB on the display it's no big deal as the final values can be adjusted. This is explained more below.

F. *Using a sound editor*:   While recording, bring the level of the audible oscillator down in equally-spaced increments. 10 increments are fine and 5 are also okay but more is always better. Play a note once at each increment. Once the minimum level setting has been reached stop the recording and normalize it. This will bring the maximum value to 0 dB. Now make a note of the corresponding decibel levels for each setting of the oscillator level.

G. *Not using a sound editor*: Bring the level of the audible oscillator down in equally-spaced increments while writing down the decibel level at each interval. 10 increments are fine and 5 are also okay but more is always better. If you were unable to adjust the level so that at it's maximum it displayed as 0 dB on the meter, simply take the difference between 0 dB and what you were able to get it to on the display and add this difference to each one of your decibel values. By adding this difference the maximum value becomes 0 dB and each of the successive values of course increases by the value of the difference.

Filter Cutoff:

A. Some synthesizers have filters that are self resonating. That means the filter will still produce resonance even if all of the oscillators or noise generators are turned off. If the synthesizer you are using is self resonating then conduct this procedure with all oscillators and noise sources off.   If the synthesizer is not self resonating then the next best thing to use is a noise source and if this is not available use a single oscillator.

B. If the filter has keyboard tracking make sure it is turned off.

C. Also turn off the filter envelope and filter velocity response if present.

D. Turn the filter resonance all the way up.

E. Experiment by changing the amount of filter cutoff and notice how the pitch of the filter resonance changes. We are going to use this to determine the frequency cutoff settings. The frequency of the resonance corresponds roughly with the frequency of the filter cutoff. The two are close enough that for the purposes of this book we can take the resonance frequency as being the same as the filter cutoff frequency. This process to an extent ignores the fact that filters are logarithmic. Despite these approximations the results are fairly close to the actual values of the filter.

F. Whereas with the other setting calibrations we might start at the lowest value and work upward in intervals, here we are going to start in the middle of the filter cutoff's range and work outward. This is done because it can be very difficult to hear, let alone identify, the high and low frequencies produced by the resonator so it is best to start in the middle.

G. With the cutoff frequency setting placed in the middle of its range try to identify the resonator's pitch with a note either on another keyboard or by momentarily turning on one of your synthesizer's oscillators. Make sure that whatever source you use for comparison that it is not detuned!

H. Locate the key in Table 1 to determine the corresponding frequency of the filter cutoff setting. Adjusting the filter cutoff up or down, work outward in evenly-spaced intervals using this same approach. Remember to write down the settings along with their frequencies.

I. At extreme high and low cutoff frequencies the resonance may become inaudible. The thing to do here is to determine these frequencies by basing them off of the frequencies from the intervals that have already been determined. Looking at the values you've written down notice that with each interval the frequency multiplies by *about* the same amount each time. Now since filters are logarithmic the value doesn't multiply by exactly the same amount each time but it's fairly close. Knowing this makes it simple to find the high and low frequencies. Ok, maybe it's not so simple!

J. Divide the highest frequency that you found by ear by the second highest frequency to form a ratio. Now multiply the ratio by the highest frequency. This new value is the frequency of the next highest interval that could not be found by ear. Multiply this new frequency by the ratio to find the frequency of the next interval and so on until you have all the inaudible high frequencies.

K. To find the low frequencies divide the lowest frequency found by ear by the second lowest frequency to again form a ratio. Multiply this ratio by the lowest frequency that could be found by ear to determine the frequency of the next lowest interval. Repeat this process just like before. For more clarification please look at the example on the next page.

CALIBRATION

Table 1. Note Frequencies

NOTE	FREQUENCY (Hz)	NOTE	FREQUENCY (Hz)	NOTE	FREQUENCY (Hz)
C1	32.703	F#4	369.994	C8	4186.009
C#1	34.648	G4	391.995	C#8	4434.922
D1	36.708	G#4	415.305	D8	4698.636
D#1	38.891	A4	440.000	D#8	4978.032
E1	41.203	A#4	466.164	E8	5274.041
F1	43.654	B4	493.883	F8	5587.652
F#1	46.249	C5	523.251	F#8	5919.911
G1	48.999	C#5	554.365	G8	6271.927
G#1	51.913	D5	587.330	G#8	6644.875
A1	55.000	D#5	622.254	A8	7040.000
A#1	58.270	E5	659.255	A#8	7458.620
B1	61.735	F5	698.456	B8	7902.133
C2	65.406	F#5	739.989	C9	8372.018
C#2	69.296	G5	783.991	C#9	8869.844
D2	73.416	G#5	830.609	D9	9397.273
D#2	77.782	A5	880.000	D#9	9956.063
E2	82.407	A#5	932.328	E9	10548.082
F2	87.307	B5	987.767	F9	11175.303
F#2	92.499	C6	1046.502	F#9	11839.822
G2	97.999	C#6	1108.731	G9	12543.854
G#2	103.826	D6	1174.659	G#9	13289.750
A2	110.000	D#6	1244.508	A9	14080.000
A#2	116.541	E6	1318.510	A#9	14917.240
B2	123.471	F6	1396.913	B9	15804.266
C3	130.813	F#6	1479.978	C10	16744.036
C#3	138.591	G6	1567.982	C#10	17739.688
D3	146.832	G#6	1661.219	D10	18794.545
D#3	155.563	A6	1760.000	D#10	19912.127
E3	164.814	A#6	1864.655	E10	21096.164
F3	174.614	B6	1975.533	F10	22350.607
F#3	184.997	C7	2093.005	F#10	23679.643
G3	195.998	C#7	2217.461	G10	25087.708
G#3	207.652	D7	2349.318	G#10	26579.501
A3	220.000	D#7	2489.016	A10	28160.000
A#3	233.082	E7	2637.020	A#10	29834.481
B3	246.942	F7	2793.826	B10	31608.531
C4(mid C)	261.626	F#7	2959.955		
C#4	277.183	G7	3135.963		
D4	293.665	G#7	3322.438		
D#4	311.127	A7	3520.000		
E4	329.628	A#7	3729.310		
F4	349.228	B7	3951.066		

Example:

1. Below are the values of the filter cutoff frequency of a particular synthesizer that were found by matching the resonance pitch to its corresponding note on a keyboard and using Table 1 to find the frequency. Values from 0%-20% and 80%-100% are absent because an accurate pitch could not be determined by ear. Those values will be found using some math.

Cutoff Setting	Cutoff Frequency
30%	836 Hz
40%	1463 Hz
50%	2540 Hz
60%	4277 Hz
70%	6906 Hz

2. Finding higher frequency settings: Take the highest frequency and divide it by the next lowest frequency to find a ratio.

$$\frac{6906\,Hz}{4277\,Hz} = 1.62$$

Multiply the value of this ratio by 6906 Hz in order to find the frequency at a cutoff setting of 80%

$$6906\ Hz \times 1.62 = 11{,}155\ Hz,$$

at 90%

$$11{,}155\ Hz \times 1.62 = 17{,}273\ Hz,$$

at 100%

$$17{,}273\ Hz \times 1.62 = 25{,}735\ Hz,$$

Notice that for each interval the ratio is multiplied by the previous frequency.

3. Finding lower frequency settings: Take the lowest frequency and divide it by the next *highest* frequency.

$$\frac{836\,Hz}{1463\,Hz} = 0.57$$

Multiply the value of this ratio by 836 Hz in order to find the frequency at a cutoff setting of 20%

836 *Hz* x 0.57 = 478 *Hz*,

at 10%

478 *Hz* x 0.57 = 274 *Hz*,

at 0%

274 *Hz* x 0.57 = 156 *Hz*,

4. Below on the left are the values that have been found. Those values found by ear are in bold and values that were calculated are in italics. Below on the right are the actual values of the filter at their respective cutoff settings.

CALCULATED VALUES		ACTUAL VALUES	
Cutoff Setting	**Cutoff Frequency**	**Cutoff Setting**	**Cutoff Frequency**
0%	*156 Hz*	0%	*26 Hz*
10%	*274 Hz*	10%	*253 Hz*
20%	*478 Hz*	20%	*483 Hz*
30%	**836 Hz**	30%	836 *Hz*
40%	**1,463 Hz**	40%	1,463 *Hz*
50%	**2,540 Hz**	50%	2,540 *Hz*
60%	**4,277 Hz**	60%	4,277 *Hz*
70%	**6,906 Hz**	70%	6,906 *Hz*
80%	*11,155 Hz*	80%	10,695 *Hz*
90%	*17,273 Hz*	90%	15,134 *Hz*
100%	*25,735 Hz*	100%	20,546 *Hz*

The values that were calculated are quite a bit off from the actual values at the extreme ends of the filter. This is because we've ignored the fact that the filter is logarithmic. This is also why it is best to find as many frequencies as possible by ear. The

inaccuracies of the values found between 10%- 20% and 80%-90% won't make a dramatic difference in the tonal color of the sound. In fact most cutoff frequencies above 12 kHz sound the same. At 0% it is safe to assume a value of 20 Hz with just about any synthesizer. For the 100% setting keep in mind that filters typically open up to 12 kHz, 16 kHz or most commonly 20 kHz.

It is not necessary that your intervals be done in 10% increments. These were just chosen here for the example. However, it is important that the cutoff settings be changed by the same amount each time. For instance if the filter is set using a dial then rotate the dial by the same amount each time. It would be possible to find values for all of the settings above even if only two frequencies had been determined in step one, but the more frequencies that are found by ear the more accurate the list will be.

LFO Frequency:

A. To determine the frequency settings of the LFO requires the use of a sound editor.

B. Route the LFO to the amplitude and set its waveshape to square. Set the LFO frequency to its highest (fastest) setting. Set the LFO amount to its maximum.

C. Begin recording with the sound editor. Play a note on the synthesizer for a couple of seconds.

D. Decrease the frequency setting in evenly-spaced intervals and each time you decrease it play a note for a couple of seconds.

E. When the frequency setting is turned all the way down stop recording.

F. The wave file should look something like this:
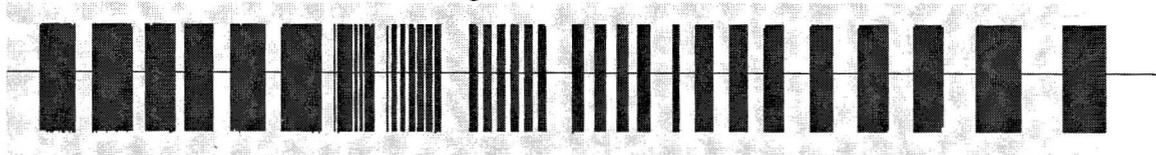
G. Zoom into the recording of the first key that was played until you can easily identify individual pulses. It will look something like this:
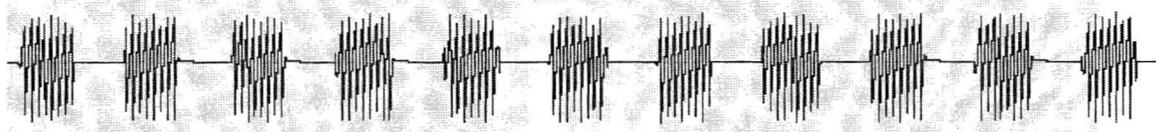

H. In order to determine frequency we must first determine period. Measure the time interval between the leading edges of two adjacent pulses. This time interval is the period. In many sound editors this can be done easily just by highlighting the region between the two leading edges and reading the time interval that becomes displayed.

Here's an example:
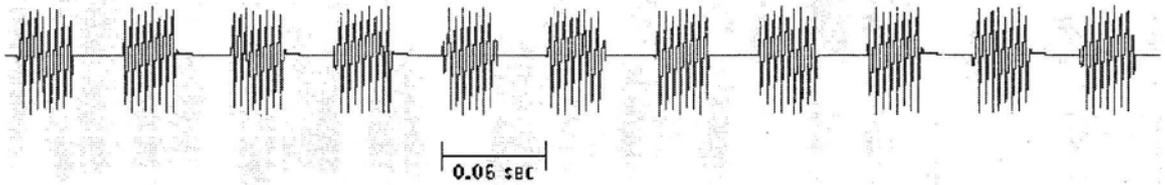
I. Take the measured period and divide it into 1 in order to find the frequency.

J. Using the period of 0.06 seconds as an example: $\dfrac{1}{0.06 \sec} = 16.7 Hz$

K. Repeat steps G through I for each LFO setting in the recording.

L. The lower LFO frequencies may be difficult to visually discern from one another in the sound editor window since it may be difficult to determine where one recording of an LFO setting ends and the next one begins. Look at the last third of the figure from step F:
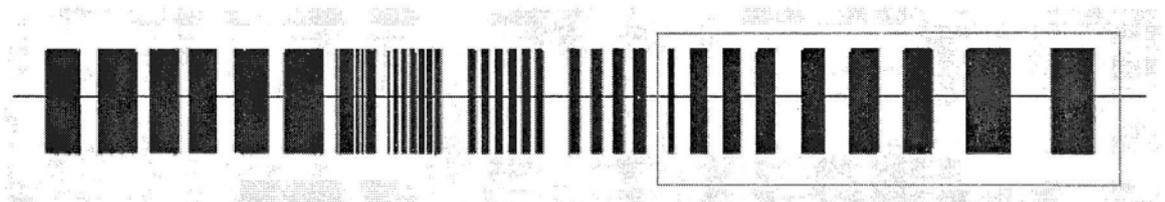

The period between LFO pulses is nearly the same length as the time between successive key strikes. Inside the above figure in the box there are actually three different LFO frequencies. The way to discern them is by noting that the width of the pulses become wider as the LFO frequency is decreased. In the figure below the three LFO settings have been separated. Notice that the pulse width in each group is different. The narrow slice in the first box is due to the LFO not being retriggered.
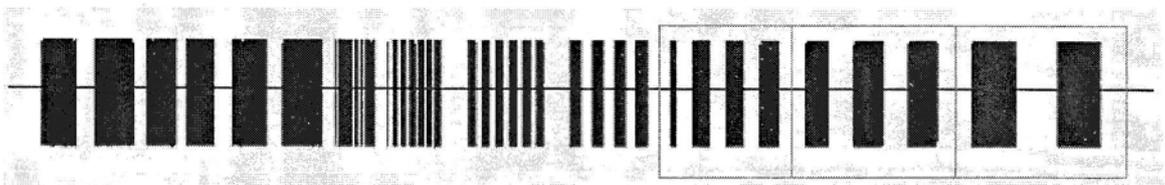
Glide time:

A. This calibration assumes a glide/portamento that is based on time rather than being proportional to the pitch interval between the two struck notes.

B. Use a sound editor to record the sound of one note gliding to another note. Hold the lowest note on the keyboard followed by the highest note so that you hear a rise in pitch.

C. In the recording zoom into the region where the two notes begin to glide. An example is given in the figure below.

D. Measure the time interval from just before the pitch begins to change until just after it stops changing. This is glide time at that setting. In the figure below for example the glide time is measured to be 0.092 seconds.

E. Complete this procedure multiple times at evenly-spaced intervals of the glide time setting while making note of the values.

0.092 sec
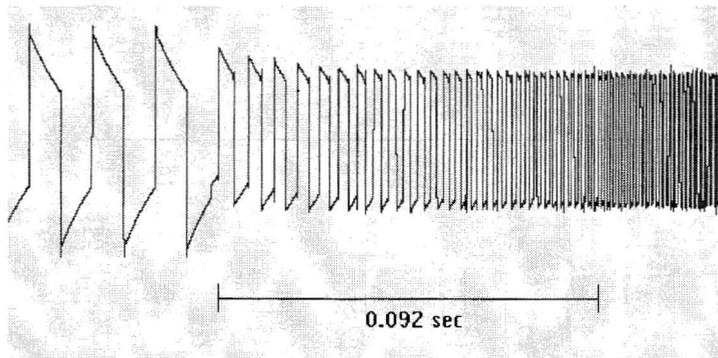
ADSR Envelopes:

A. It can usually be assumed that the envelopes for amplitude and filtering have the same response. We will determine the envelope settings for both using the amplitude envelope but these same values can also be used for the filter envelope.

B. Attack:

    1. Set decay, sustain, and release all to zero. Set the attack time to its maximum setting.

2. Use a sound editor to record a note from the synthesizer. Hold the note long enough for the attack to finish. You will hear the amplitude increase and then instantly drop back to zero. In the sound editor measure the amount of time it took for the amplitude to max out. Make note of the values.

3. Repeat step 2 for successively smaller attack times by decreasing the setting in evenly-spaced intervals.

C. Decay:

1. Set attack, sustain and release to zero. Set the decay time to it's maximum setting.

2. Use a sound editor to record a note from the synthesizer. The decay will cause the sound to slowly decrease in amplitude. Hold the note long enough for the amplitude to go to zero. In the sound editor measure the amount of time it takes for the amplitude to go to zero.

3. Repeat step 2 for successively smaller decay times by decreasing the setting in evenly-spaced intervals. Again, make note of the values.

D. Sustain: this is measured as a percentage of the full setting rather than of time and therefore doesn't require any determination. For example, if the knob/slider is turned halfway then the sustain is at 50%.

E. Release:

1. Set attack and decay to zero. Set sustain at 100%.

2. Use a sound editor to record a note from the synthesizer. Quickly strike but do not hold a note on the synthesizer. In the sound editor measure the amount of time it takes for the amplitude to go to zero.

3. Repeat step 2 for successively smaller release times by decreasing the setting in evenly-spaced intervals. Make note of the values.

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PATCH QUICK START: The filter cutoff frequency is the one parameter where getting the correct setting is most critical. If your synthesizer's filter doesn't give cutoff frequency values in Hz/kHz and the percentage values in the book don't seem to produce the correct results and if you don't feel like going through the whole calibration routine at the back of the book then to get off to a quick start do the following: Set the synth to output a single sawtooth. Set the filter's cutoff frequency to the very middle of its range and match the output to one of the sound files on the CD in the Calibration Sound Files → Filter → 12 db *or* 24 db subfolder. The name of the sound file that matches the output gives the approximate cutoff frequency (in Hz) that corresponds to the middle position. Assume the lowest dial position of your synth's cutoff frequency is 20 Hz and the highest position is a cutoff frequency of 20,000 Hz (20 kHz). These two values may not be correct for your synth but they will be close enough sound wise. Knowing just these three values, low, mid, and high, will actually make approximating the filter cutoff settings listed in the patches quite a bit more accurate.

Note: filter cutoff frequency settings generally do not operate on a linear scale so if you come up with a middle setting of something like 300 Hz, 600 Hz, 1200 Hz etc. it is probably correct. In other words, the mid value is not going to be anywhere near 10,010 Hz which would be the *numerical* mid point between 20 Hz and 20,000 Hz.