Table of Contents

1 Introduction
- 1.1 The Purpose of this Document
- 1.2 Acknowledgements

2 Oscillator Waveform Modification
- 2.1 Sync
- 2.2 Frequency Modulation Techniques
- 2.3 Wave Shaping
- 2.4 Vector Synthesis
- 2.5 Wave Sequencing
- 2.6 Audio-Rate Crossfading
- 2.7 Wave Terrain Synthesis
- 2.8 VOSIM
- 2.9 FOF Synthesis
- 2.10 Granular Synthesis

3 Filter Techniques
- 3.1 Resonant Filters as Oscillators
- 3.2 Serial and Parallel Filter Techniques
- 3.3 Audio-Rate Filter Cutoff Modulation
- 3.4 Adding Analog Feel
- 3.5 Wet Filters

4 Noise Generation
- 4.1 White Noise
- 4.2 Brown Noise
- 4.3 Pink Noise
- 4.4 Pitched Noise

5 Percussion
- 5.1 Bass Drum Synthesis
- 5.2 Snare Drum Synthesis
- 5.3 Synthesis of Gongs, Bells and Cymbals
- 5.4 Synthesis of Hand Claps

6 Additive Synthesis
- 6.1 What is Additive Synthesis?
- 6.2 Resynthesis
- 6.3 Group Additive Synthesis
- 6.4 Morphing
- 6.5 Transients
- 6.7 Which Oscillator to Use

7 Physical Modeling
- 7.1 Introduction to Physical Modeling
- 7.2 The Karplus-Strong Algorithm
- 7.3 Tuning of Delay Lines
- 7.4 Delay Line Details
- 7.5 Physical Modeling with Digital Waveguides
- 7.6 String Modeling
- 7.7 Woodwind Modeling
- 7.8 Related Links

8 Speech Synthesis and Processing
- 8.1 Vocoder Techniques
- 8.2 Speech Synthesis
- 8.3 Pitch Tracking

9 Using the Logic Modules
- 9.1 Complex Logic Functions
- 9.2 Flipflops, Counters other Sequential Elements
- 9.3 Asynchronous Elements
- 9.4 Arpeggiation

10 Algorithmic Composition
- 10.1 Chaos and Fractal Music
- 10.2 Cellular Automata
- 10.3 Cooking Noodles

11 Reverb and Echo Effects
- 11.1 Synthetic Echo and Reverb
- 11.2 Short-Time Reverb
- 11.3 Low-Fidelity Echo and Reverb

12 Distortion Effects
- 12.1 Distortion using Nonlinearities
- 12.2 Multiband Distortion
- 12.3 Polynomial Distortion
- 12.4 Distortion using Time-Variant Systems

13 Frequency and Pitch Shifting
- 13.1 Frequency Shifting
- 13.2 Pitch Shifting

14 Spatialization

15 Emulating Classic Synths

15.1 General Guidelines
15.2 Yamaha DX7
15.3 Octave Cat
15.4 ARP Odyssey
15.5 Casio
15.6 Hallsey Mark 1 and 2
15.7 Other Emulations

[Next (Introduction)]

Chapter 1. Introduction

1.1 The Purpose of this Document

This web-based hypertext document is intended to provide the advanced user of the Clavia Nord Modular synthesizer with a toolbox of techniques for creating complex and interesting patches. Although the emphasis is on the Nord Modular, the techniques described in this document can be applied to any modular synthesizer, from a modular Moog to the Native Instruments Reaktor softsynth. For more information on the Nord Modular, check out Clavia's web page at http://www.clavia.se.

This document fills a need for more advanced material beyond what can be found in the Nord Modular manual and in the Wizoo Guide to the Nord Modular (by Peter Gorges and Len Sasso, 2nd edition 2000). While both of these books are excellent introductions for the beginner, more advanced users are left thirsting for more. This document aims to quench this thirst!

In the online web-based document all of the patches shown in the figures can be downloaded by clicking on the associated image. These can be then be examined off-line with the Clavia Nord Modular editor (version 3.0 or higher) or downloaded to an actual Nord Modular. The latter is recommended as you can hear directly the sounds that are being talked about in the text.

1.2 Acknowledgements

The document incorporates many random thoughts and ideas gleaned from the Nord Modular mailing list, as well as, to a lesser extent, the Native Instruments Reaktor and Analog Heaven mailing lists. All of these mailing lists are excellent sources of information and support for people interested in modular music synthesis.

Many people have contributed ideas and material to this work. I would especially like to thank Rob Hordijk, Wout Blommers, Jan Punter, Kees van der Maarel, Roland Kuit, Ico Doornekamp, Robin Whittle, Chet Singer, Dave Peck, Sam Streeper, and K. Lex Pattyrson (or whatever his name is) for their contributions and inspiration.

Chapter 2. Oscillator Waveform Modification

Most of the oscillators in the Nord Modular provide only the most basic waveform shapes: Sine, Triangle, Sawtooth, and Square (or Pulse). The Formant oscillator and the Spectral Oscillator are exceptions to this, and we will discuss them later.

The so-called *subtractive* approach to sound synthesis uses filtering of these basic waveforms to provide control over the harmonic content of a sound. A sinewave has only a single harmonic, and therefore filtering does not alter its harmonic content at all, merely the amplitude of the sound. A triangle wave has more harmonics, but with very low amplitudes, so they too are not too useful as sources for subtractive synthesis. The most useful waveforms for subtractive synthesis are the sawtooth and square (or pulse) waveforms. These waveforms have strong harmonics, suitable for alteration through filtering. A sawtooth waveform contains all harmonics of the fundamental pitch of the waveform, while the square waveform contains just the odd harmonics. A pulse waveform, with a duty cycle different than 50% also has all harmonics, even and odd. Changing the pulse width alters the relative strength of the individual harmonics, with the harmonic levels become more equal as the pulse narrows.

In the subtractive synthesis approach, variations in the spectrum of the sound (or *timbre*) are usually generated by varying the characteristics, such as cutoff frequency and resonance, of the filters that the oscillator waveforms are passed through. While this is effective for synthesizing many pleasing and interesting sounds, the result is limited somewhat by the rather sparse choices of spectra for the oscillators. Because of this, synthesizer manufacturers and patch programmers have come up with a number of ways in which to expand the spectral possibilities beyond those of the basic oscillator waveforms. In this chapter we will examine some of these approaches and show how they can be implemented in the Nord Modular.

2.1 Oscillator Sync

Oscillator Sync alters the waveform of an oscillator by resetting the phase of the oscillator waveform when a sync pulse is received. If this sync pulse occurs at a rate faster than the nominal oscillator frequency, the oscillator will become locked to the sync rate. The specific harmonic structure of the synced oscillator waveform will depend on the nominal oscillator frequency. If the sync pulse occurs at a rate slower than the nominal oscillator frequency, the oscillator will not become locked to the sync rate, but will gain strong harmonic components corresponding to the sync frequency. I will not go into too much additional detail here, but instead refer you to the excellent tutorial on Oscillator Sync by Rob Hordijk, found on the Clavia web site at www.clavia.se/nordmodular/Modularzone/Hardsync.html and www.clavia.se/nordmodular/Modularzone/Softsync.html.

I will give here a soft-sync technique that was not covered in the tutorial. This technique was used in the RMS (Rivera Music Systems) synthesizer as well as in the Wiard synthesizers. In this technique, the synced oscillator produces a triangle wave (which could be passed through a wave shaper to produce other waveforms). To this triangle waveform a narrow pulse from the syncing oscillator is added. The sum is compared to a reference level. If the sum exceeds this reference level, the synced oscillator is reset, or the triangle wave reversed. The probability of a syncronization event being created can be seen to be related to the amplitude of the syncing pulse as well as the amplitude of the triangle wave output of the synced oscillator. If the syncing pulse has a low amplitude, then syncing will occur only if the syncing pulse happens very close in time to the peak of the triangle waveform. Otherwise no syncing will occur. At the other extreme, if the syncing pulse has a very large amplitude, then syncing will occur on every pulse, no matter what the phase of the triangle waveform. In this case, the sync action will be that of regular hard-sync. By adjusting the amplitude of the syncing pulse one can vary the sync effect from soft to hard. This technique is illustrated by the following Nord Modular patch. It is not exactly as in the RMS and Wiard synths, since the triangle wave is reset on reception of a sync signal, rather than being reversed. But for those intrepid patchers who want more authenticity, follow the instructions in Rob Hordijk's above referenced softsync tutorial, where he describes how to reverse the slope of a triangle waveform on the Nord Modular.

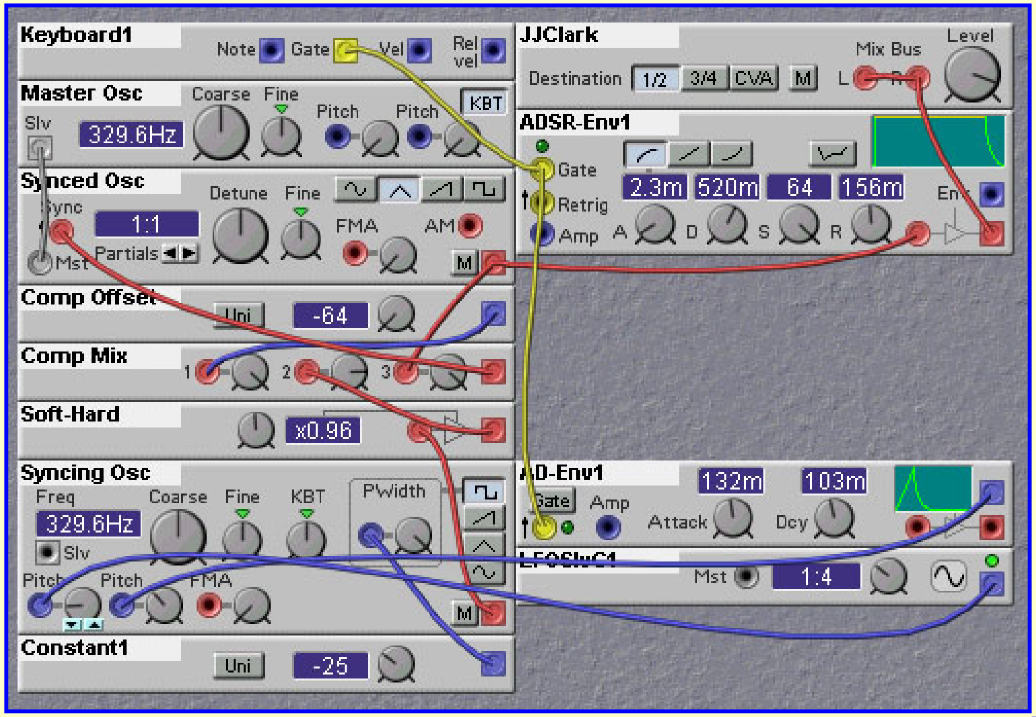


Figure 2.1. Patch illustrating a softsync technique (J. Clark).

For the technical minded among you, there are some ways to make sync patches sound even better. The NM, as we all know, is a digital system at heart, even though it might sound "analog" from time to time. Because of the finite sampling rate used in the NM, the syncing waveform is actually quantized in time. This means that the synced oscillator is not only synced to the syncing waveform but also to the sampling clock. At low frequencies of the syncing waveform the effect of the sampling clock sync will not be too noticeable, but at high frequencies it becomes quite significant. Rob Hordijk suggests a way to remedy this problem. The jitter caused by the syncing to the sampling clock is most noticeable at the transient that inevitably appears at the point where the syncing signal causes the synced waveform to switch. One can apply a mask, or envelope, over the synced oscillator waveform, to suppress this transient. A simple way to do this is to use a downward sawtooth as the syncing waveform and also use it as an amplitude envelope that modulates the synced waveform. This has the drawback of distorting the synced waveform, of course. An improvement can be had by crossfading two such synced waveforms, each out of phase with respect to each other. This will minimize the distorting effect of the masking. The following patch by Rob Hordijk illustrates the approach.

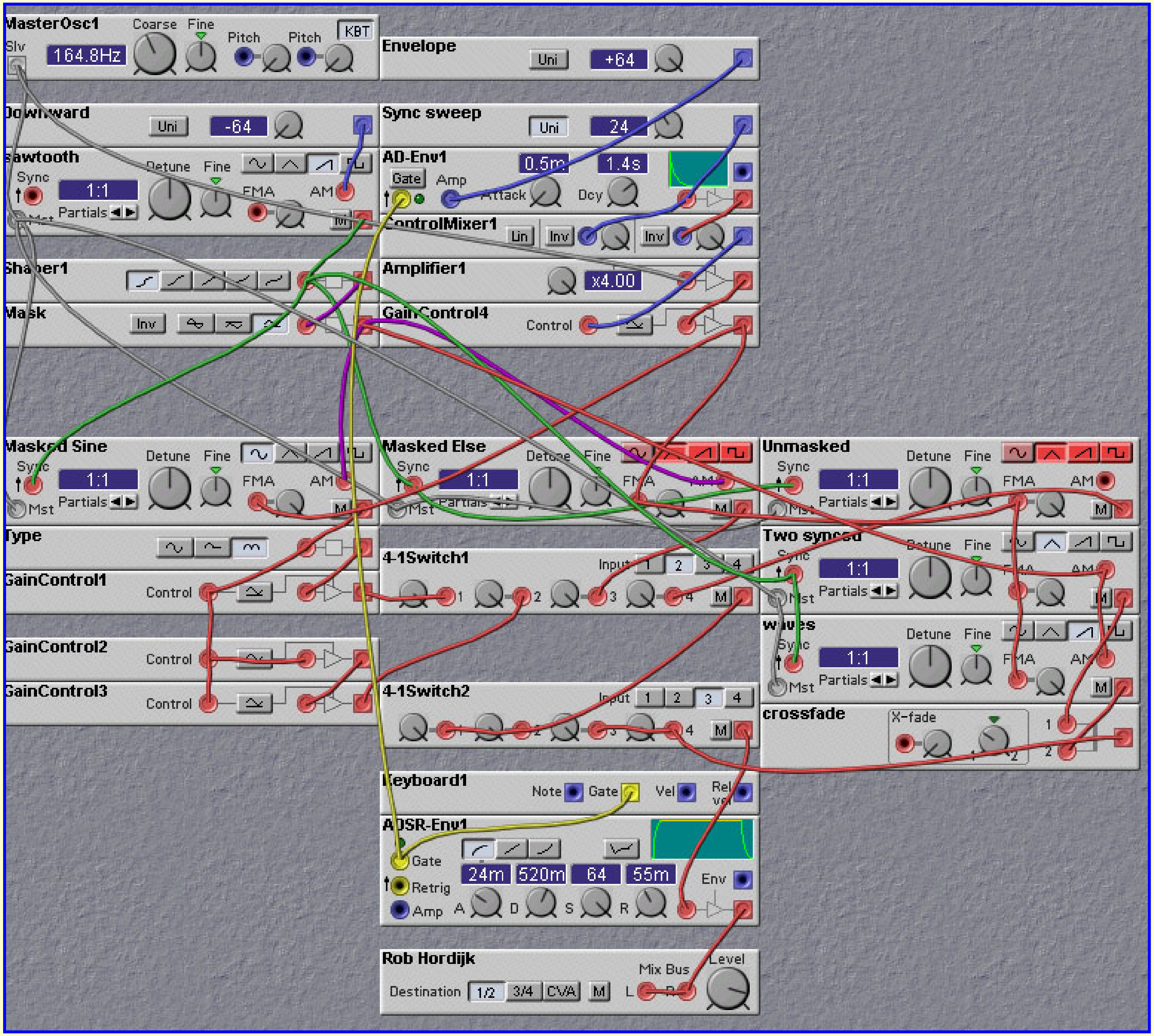


Figure 2.2. Patch illustrating a masked hardsync technique (R. Hordijk).

2.2 Frequency Modulation

Frequency modulation provides an easy way to alter the harmonic content of a sound. The sounds of many instruments are characterized by the way in which their timbre changes over time. To generate similar sounds using FM is often merely a matter of varying the modulation level appropriately, using envelope generators or LFOs.

The simplest FM techniques feed the output of one sine-wave oscillator, called the modulator, into the frequency modulation input of another, called the carrier. If the two sinewaves have relatively harmonic frequencies, then the resulting waveform out of the carrier will be harmonic as well. The presence and amplitude of the carrier's harmonics is determined by the relative frequency of the carrier and modulator. By adjusting the ratio between these two frequencies we can get quite a range of different harmonic structures.

For example, to emulate Brass instruments using FM, set the frequency of the modulating signal to be the same as that of the carrier. Use an envelope generator to set the modulation level, resulting in the characteristic brightening of the timbre during the attack.

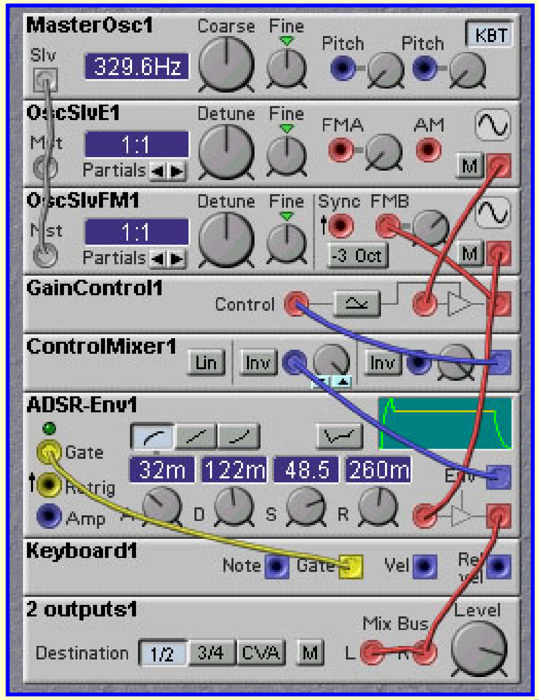


Figure 2.3. Patch illustrating a simple FM approach to obtaining a brass-like sound (J. Clark).

Instruments with cylindrical bores, such as the clarinet, have mainly odd harmonics. To obtain this type of spectra we set the frequency of the modulator to be twice that of the carrier. Conversely to the case of brass instruments, the timbre of the clarinet starts out bright then becomes duller during the attack phase. So use an inverted envelope generator to adjust the modulation level.

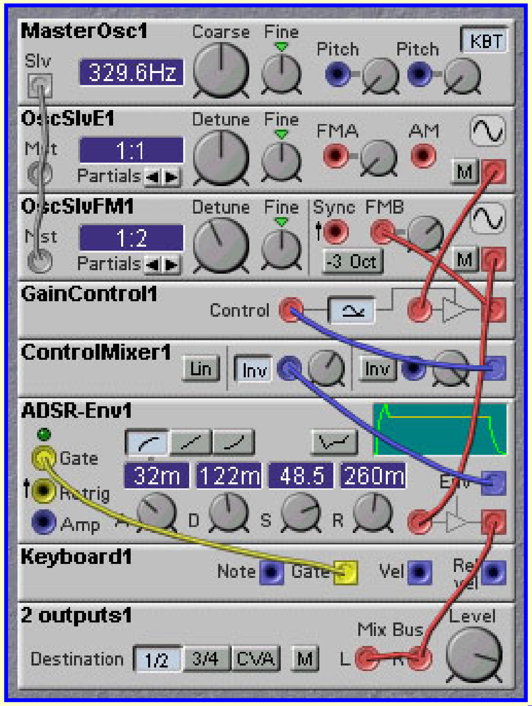


Figure 2.4. Patch illustrating a simple FM approach to obtaining a clarinet-like sound (J. Clark).

To model the sound of the Bassoon and similar instruments using FM, set the frequency of the modulator to a sub-multiple of the carrier frequency. This will weaken the strength of the fundamental relative to the higher harmonics. All harmonics (odd and even) are present in these types of instruments, due to the tapered bore.

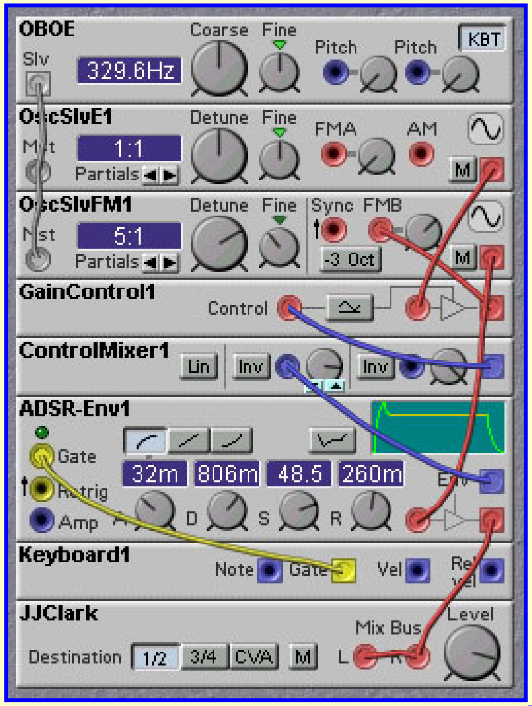


Figure 2.5. Patch illustrating a simple FM approach to obtaining an oboe-like sound (J. Clark).

FM synthesis is capable of much more than the simple examples shown here. The output of one carrier oscillator can be used as a modulator waveform as well, and one can create modulation chains, where the spectrum gets more and more complex as one moves down the chain. Modulating the amplitude of the carrier signals in such a chain using envelope generators or LFOs can produce quite expressive and complex time-varying sounds. The following patch shows a three-stage FM chain. The sound is more lively than the simple clarinet patch given above.

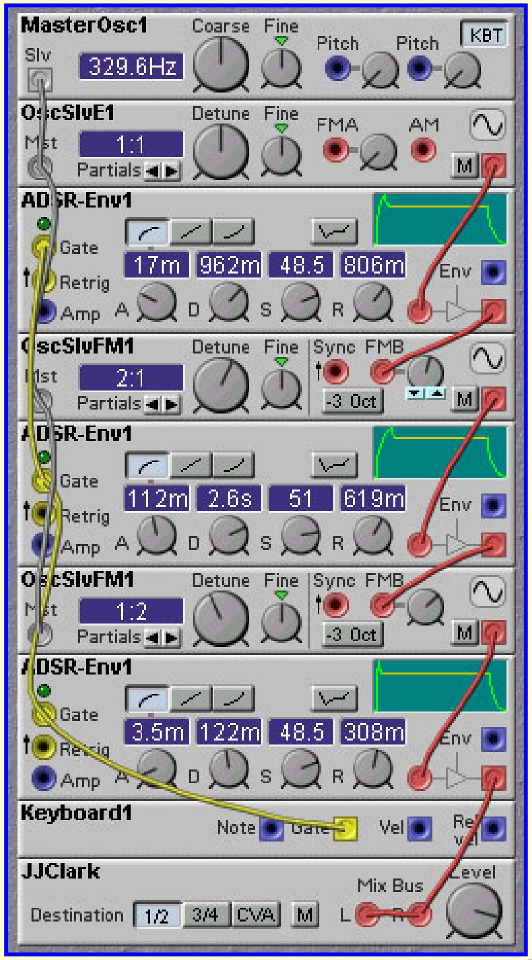


Figure 2.6. Patch illustrating the use of multiple modulation stages in an FM patch (J. Clark).

One can use non-sinusoidal waveforms for the carrier and modulators, providing an even greater variety of harmonic spectra. There is no reason one cannot use modulator and carrier frequencies that are non-harmonic. Using these will result in waveforms that have non-harmonic spectra, good for metallic sounds such as bells and gongs. The following two patches illustrate the use of non-sinusoidal modulator and carrier waveforms. The harmonic structure is more complex than in the simple case of sinusoidal waveforms.

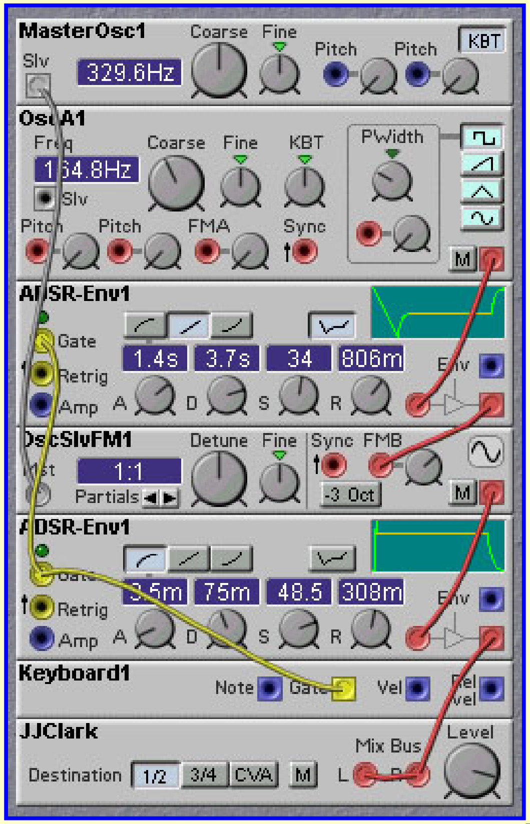


Figure 2.7. Patch illustrating non-sinusoidal modulation waveforms in an FM patch (J. Clark).

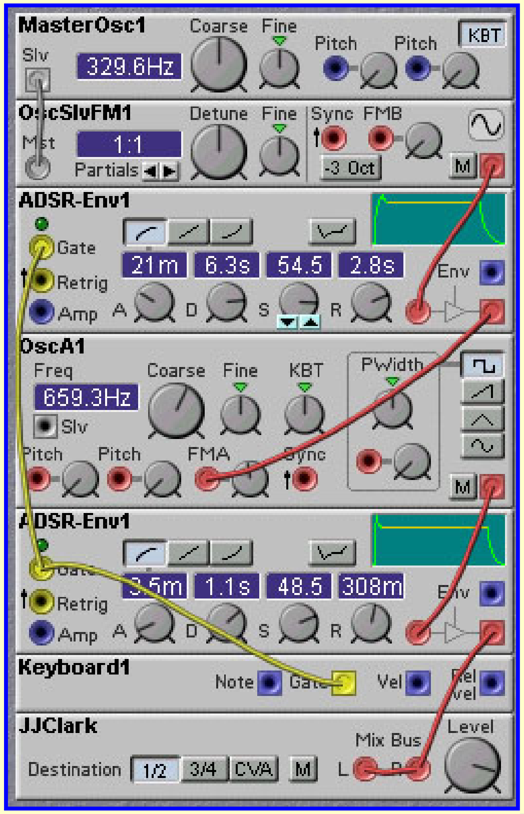


Figure 2.8. Patch illustrating non-sinusoidal carrier waveforms in an FM patch (J. Clark).

Feedback is a technique that always produces interesting, if not musical, results, and this is true for FM synthesis as well. One can modulate the first oscillator in a modulation chain with the output of one of the other oscillators (or with its own output, for that matter). The results of applying feedback can be very difficult to analyze mathematically, even in the case where all oscillator frequencies are nominally harmonic. For this reason it is advisable to experiment, and see what comes about. Some general rules can be found, however. First is that a little bit of feedback goes a long way **-** too much feedback causes instability and even chaotic behaviour. Of course, this might be just what you are looking for! Likewise, feedback from an oscillator output closer to the oscillator being modulated gives a more gentle effect than feedback from an output further down the chain. Inserting time delays into the feedback patch can create very unpredictable shifts in the timbre.

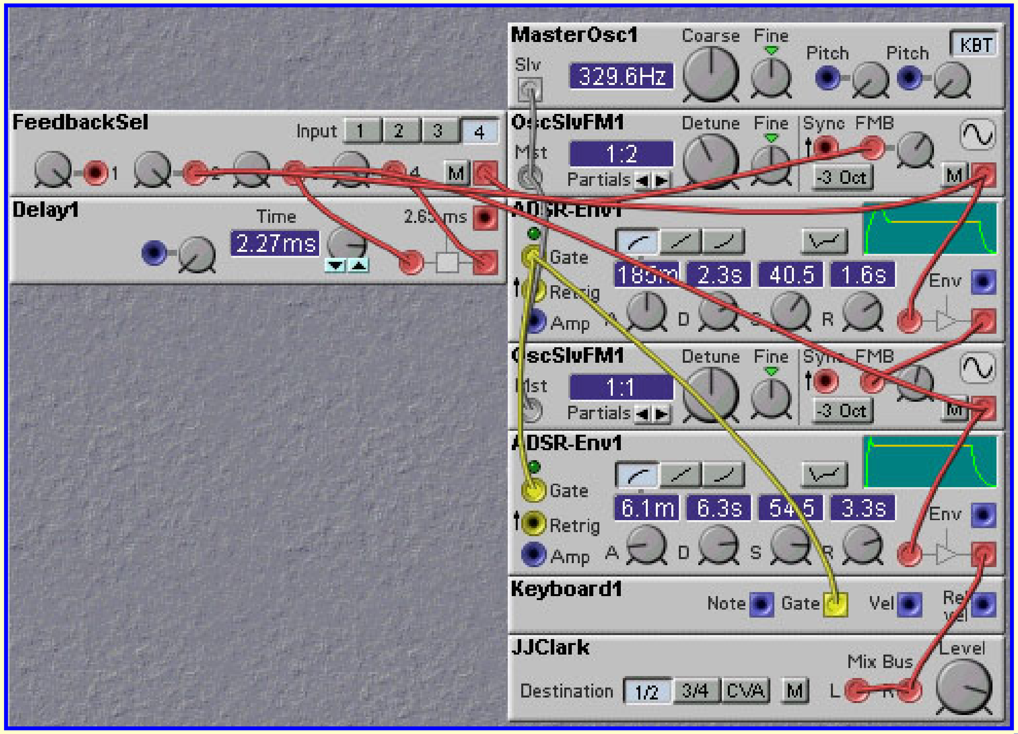


Figure 2.9. Patch illustrating the effects of feedback back to the first oscillator in an FM modulation chain (J. Clark).

In the above patch Knob 2 selects the source of the feedback signal, so that you can compare the effects of each. Selecting position 1 gives no feedback, position 2 gives feedback from the first oscillator (i.e. self-modulation of this oscillator), position 3 gives feedback from the second oscillator in the chain, while selecting position 4 gives time delayed feedback from the second oscillator. Play around with the amount of feedback by adjusting the OscSlave1 FM Amount control (Knob 1). If it is set too high a chaotic noisy waveform will be the result.

The oscillator modules on the Nord Modular are specifically designed to permit useful Frequency Modulation synthesis. There are four types of frequency modulation inputs on these modules - "Mst", "Pitch", "A" and "B". The "Mst" and "Pitch" frequency modulation inputs are logarithmic, while the "A" and "B" inputs are linear. The Pitch input is typically used to set the basic pitch of the oscillator, for example, from a keyboard note value. The "Mst" input is usually connected to the "Slv" output of a master oscillator.

To understand the difference between the different input types, consider a typical approach to FM synthesis, where we have two sources determining the frequency of an oscillator. The oscillator adds these two sources together to determine the actual frequency of the oscillator. Typically, one source value is a `stable' one that controls the basic pitch of the waveform (e.g. the "note" being played). The other source is some waveform that varies at an audio rate. The `stable' input should be logarithmic (=non-linear) to accomodate a keyboardvalue on the NM. If the other modulation input (the one connected to the audio rate signal) is also logarithmic the waveform shape, and therefore the timbre, changes if the pitch is increased. A low pitch will be relatively more modulated (have more harmonics) than a high pitch. Sometimes this is desirable, for example when making a fat bass sound, but it makes it not very useful for synthesizing a `physical instrument'-like sound. When modulating with a linear modulating input the harmonic content will remain stable over the complete pitch range.

But what if we selfmodulate a sine-oscillator with the FMA-input? Opening the FMA-knob will not only change the timbre, but also the pitch of the sound, since the average value of the sine output is non-zero. This means that the pitch of the sound will no longer track the note value being fed into the logarithmic pitch input. Not a good thing for emulating standard instruments! Of course, you might be looking for just that off-key type of sound... The FMB input is designed to prevent such a shift in pitch when a waveform with a non-zero average value is used as a frequency modulation source. This input is designed to mimic the form of frequency modulation used on the Yamaha FM-synthesizers. On these synthesizers, such as the DX-7, the pitch doesn't change when selfmodulating. This is because Yamaha used `phase modulation' rather than frequency modulation. In this technique, it is not the frequency of the oscillator that is modulated but it rather the phase of the wave. If the modulating waveform has a non-zero average value, there will only be an (inaudible) shift in phase, and not a shift in pitch. On the Nord Modular we can obtain Yamaha-like FM patches by means of the FMB input. The FMB input does not directly modulate the phase of the oscillator. Instead, the FMB input is passed through a highpass filter (inside the module) and the filtered result is used to frequency modulate the oscillator. The highpass filter removes the DC component of the modulating signal, thereby ensuring that the average value is zero. Hence no pitch shift will occur. Also, consider that a highpass filter acts like a signal differentiator for signals whose frequencies are on the low frequency cutoff slope region of the filter. Frequency modulation with the derivative of a signal is similar to phase modulation with the signal itself (since frequency is the time derivative of phase). Thus, signals applied to the FMB input result in something similar to phase modulation, as in the Yamaha systems. For modulating signals with frequencies above the cutoff frequency of the highpass filter, the modulation will be of frequency and not of phase, so the character of the sound created by the modulation with high frequencies will be different than that for modulation with low frequencies. One might want to lowpass filter the modulating waveform to eliminate the high frequency components that will adversely affect the sound.

Due to the great success and popularity of the Yamaha DX-7 FM synthesizer, there has been much written on the programming of FM synths. There is also a large number of patches available for study. Wout Blommers has created a vast collection of Nord Modular patches based on Yamaha DX-7 patches. These can be found, along with additional information about FM synthesis in the section on Emulating Classic Synths. Rob Hordijk has also written a nice tutorial on FM synthesis which can be accessed at http://www.clavia.se/nordmodular/Modularzone/FMsynthesis.html.

2.3 Wave Shaping

The timbre associated with a given waveform is determined primarily by the harmonic content of the waveform. One can create new harmonics by passing the waveform through a nonlinear element, normally referred to as a *waveshaper*. The principal Nord Modular modules used for waveshaping are the Signal Shaper, Wave-Wrapper, Quantizer, Diode, Clip, and Overdrive modules. Note that the Ring Modulator is *not* a nonlinear element, but is (bi-)linear. Nonlinearities can be created with combinations of other modules (such as using the gain control and ring modulator modules to multiply a signal by itself) or by (mis-)using the Logic Modules.

Nonlinear elements are defined by the dependance of the element's output on the amplitude of the input. In a linear system, if one increases the input amplitude by a given amount, the output amplitude is also increased by this amount. The shape of the output is not changed due to the increase in input amplitude, just its amplitude. In a nonlinear system this is not the case. It is therefore recommended to apply the nonlinear element after a VCA in a signal chain. As the VCA gain is altered, for example with an envelope or LFO, the amplitude of the waveform is changed. Since the level of the harmonics created by the nonlinear element depend on the amplitude of its input, applying the nonlinearity to the output of the VCA will result in time-varying harmonics. If the nonlinearity was applied before the VCA, the harmonic levels would be static and uninteresting. Likewise you should apply a filter *after* a nonlinear element rather than before (although interesting effects can be obtained by feeding the output of a resonant filter into a nonlinear element). If you are mixing together different waveforms, it is best to apply the nonlinearity before the mixing. This keeps the harmonics pure. If you are using VCAs you will need one for each waveform. Of course, interesting results can be obtained with applying nonlinearities after mixing, but the result is harder to control and the sound can get muddy if you are not careful.

The following patch illustrates the use of nonlinear elements in waveshaping.

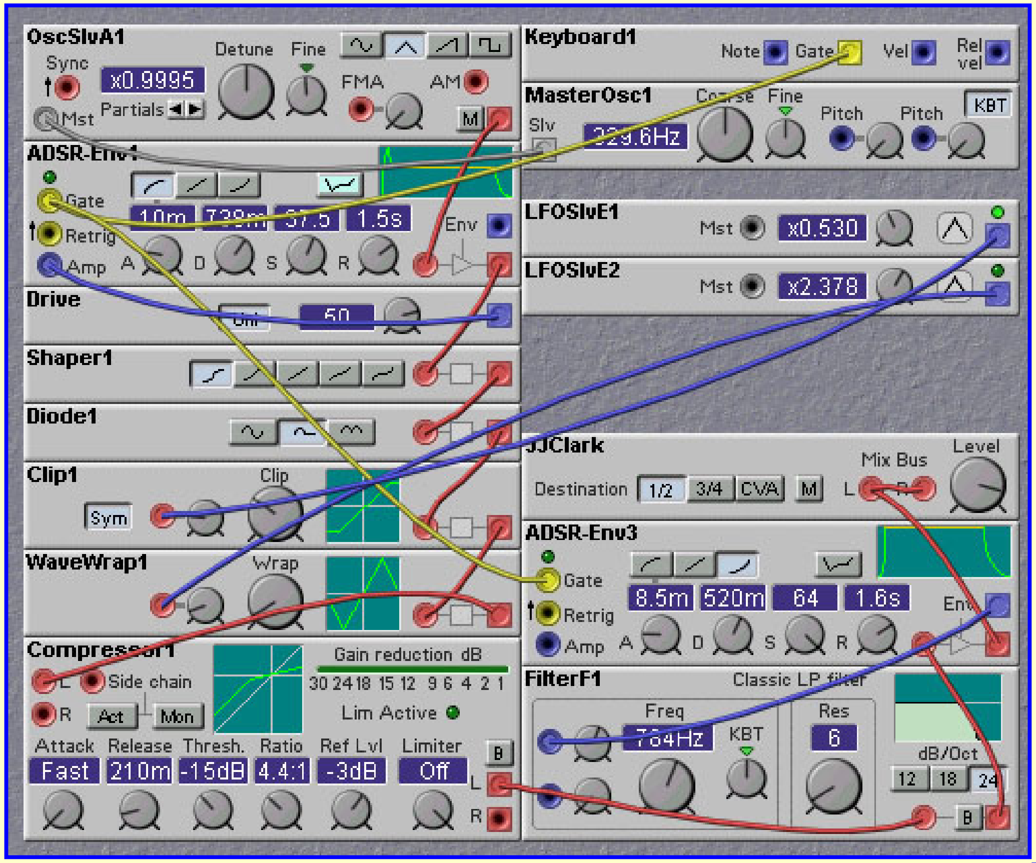


Figure 2.10. Patch illustrating waveshaping with nonlinear elements (J. Clark).

Note the use of a compressor after the nonlinear elements. The reason for this is to even out the changes in overall amplitude that can result from the application of the nonlinearities. Remember, the purpose of the nonlinear elements here is to modify the *shape* of the waveform, but not its amplitude. You want to use a VCA/Envelope Generator after the compressor to provide the dynamics for the patch.

The waveforms that you use as input to the nonlinearities are not too crucial but they do make some difference. In particular, you should avoid using pulse or square waves as input, as their shape is not affected by nonlinear elements, only their amplitude. In this regard, they are much like sine waves with respect to linear filtering operations - the shape of a sine wave is not affected by linear filtering, only its amplitude. In mathematical jargon, sine waves are *eigenfunctions* of linear time invariant systems (which is a fancy germanic way of saying that the shape of sine waves are not altered by linear filters) and pulse (two-level) waves are eigenfunctions of (memory-less) nonlinear mappings. Typically triangle waves or sawtooth waves are preferable, as their linear slopes makes predicting the effect of nonlinear operations easier than for other waveforms. In some synthesizers, many waveforms are derived from easy to produce triangle waves. For example, one can use a nonlinear element that has a gradual decrease in gain with amplitude to round off the corners of the triangle waves, producing an approximation to a sine wave. This is also an example of a case where application of a nonlinear process results in fewer harmonics than exist in the input!

2.4 Vector Synthesis

Vector Synthesis is a waveform modification technique that was developed by Sequential Circuits, and first appeared in their Prophet VS synthesizer. Vector synthesis involves the crossfading of two (or more) waveforms under user or program control. The Prophet VS used four waveforms, visualized as lying on the four corners of a square, and the performer used a 2-D joystick to move within the square. When the joystick moves close to one of the corners, the waveform associated with that corner becomes dominant and the other waveforms are attenuated. When the joystick is in the interior of the square, a linear combination of the waveforms is produced. In the Prophet VS a joystick was used to scan the four waveforms. The following patch illustrates the vector synthesis technique.

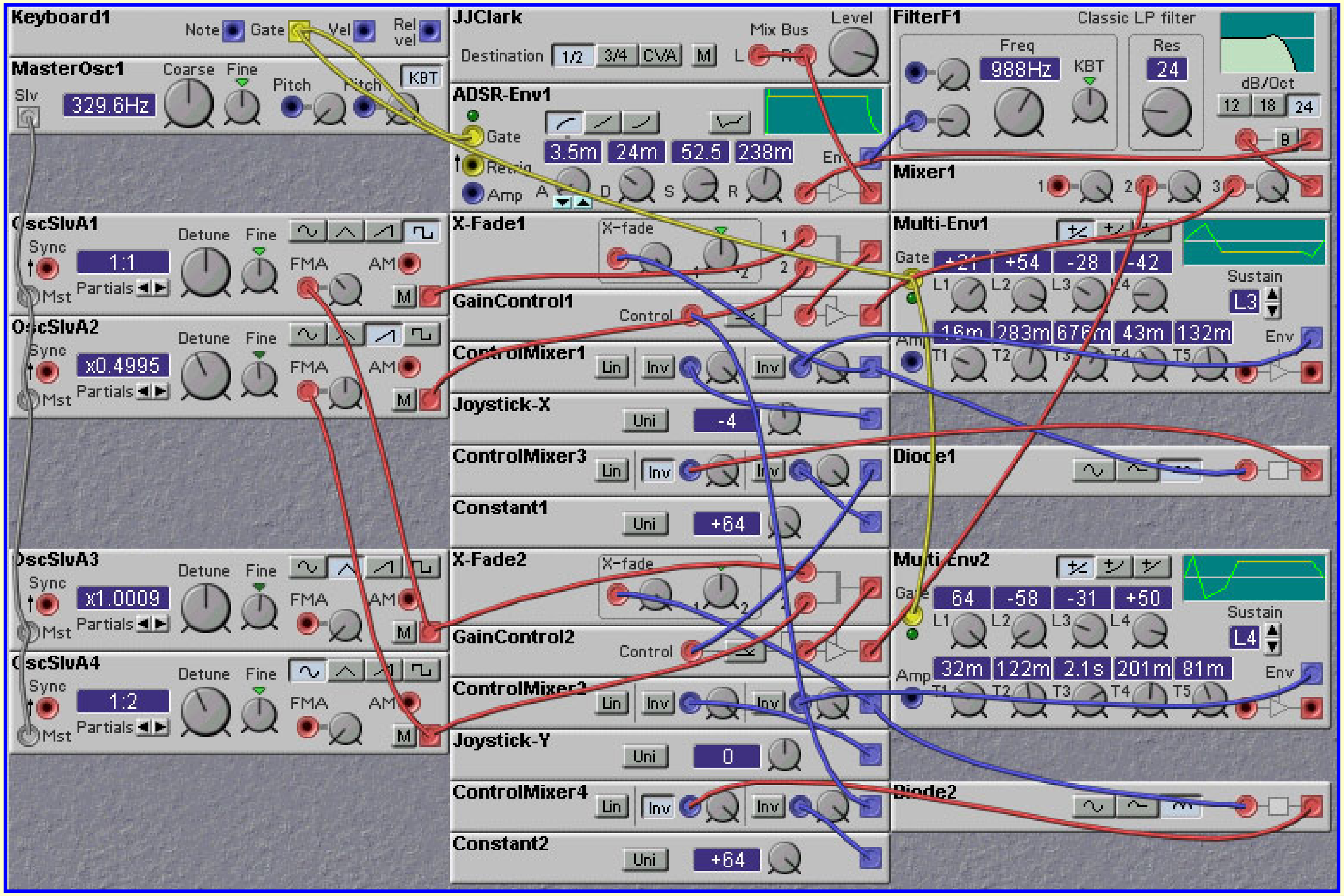


Figure 2.11. Vector synthesis patch (J. Clark).

The crossfading between the four oscillators is done via the mathematical relation:

$$F = ([A*(64-X)+B*(64+X)]*(64-|Y|)+[C*(64+Y)+D*(64-Y)]*(64-|X|))/128/64$$

where A,B,C,D represent the four waveforms and X and Y represent the two crossfade signals, which range from -64 to 64. The absolute value operation (|X| and |Y|) is implemented with the diode modules. This function has the desired properties that for X=64, Y=0, F=B; for X=-64, Y=0, F=A; for X=0, Y=64, F=C; for X=0, Y=-64, F=D; and for X=Y=0, F=(A+B+C+D)/2. Other crossfade schemes that have these properties are possible, of course.

One can iterate the vector synthesis approach, so that the basic waveforms used in these approaches are themselves vector synthesized waveforms. If you have enough computational resources, you can carry this iteration deeper and deeper, leading to fractal waveforms in the limit. The patch shown in the following figure implements a 2-level vector synthesis algorithm, using 16 oscillators (4 groups of 4 oscillators) with five 2-D waveform crossfade controls (4 to do the second level waveforms, and one to do the top level waveform). LFOs and envelopes are used to generate the crossfade controls, but, of course, any source of control signal could be used.

2.5 Wave Sequencing

Wave sequencing is a descendent of vector synthesis. Wave sequencing used up to 255 different waveforms rather than the four of vector synthesis. In wave sequencing various waveforms from the set were sequentially played back and crossfaded, according to some control signal. A precursor to wave sequencing was found in the PPG Wave synthesizer, which used a wavetable of 32 single cycle waveforms which were selected by an index controlled by an envelope or LFO. The waveforms in the set were all related, with little difference between waveforms in adjoining locations in the wavetable. One of the first instruments to employ wave sequencing was the Korg Wavestation. In this machine, the wavetable could have up to 255 different samples. These samples could be wildly different, and were not neccessarily of single cycle waveforms. This flexibility comes at a cost, however, and that is the increased complexity of designing suitable control sequences. One can adjust the rate of crossfading from one wave to the next, as well as the amount of time spent on each waveform. And then there is the agonizing choice of which waveforms to use, and in what order!

In the following patch a wave sequencing algorithm is implemented to generate a single complex waveform. In this patch the order of the waveforms is specified, as well as the amplitude and tuning of each step. The duration of each step is also specified.

One of the useful ways to work with wave sequencing is to arrange the harmonic content (and the crossfade rate) to achieve a given effect. For example, one can place bright sounding waveforms, with fast crossfade rates, early in the sequence, with darker waveforms, with slow crossfades later in the sequence. This will give the typical early-bright-late-dull sound characteristic of many natural instruments. This arrangement is implemented in the patch below. Of course, with the Nord Modular, the range of waveforms that we have to work with is much more limited than on a machine like the Korg WaveStation, so don't expect results to be as spectacular. This patch also lacks some effects that are found on the Wavestation, such as moving backwards and forwards through a set of waveforms, and the specification of a controllable starting point.

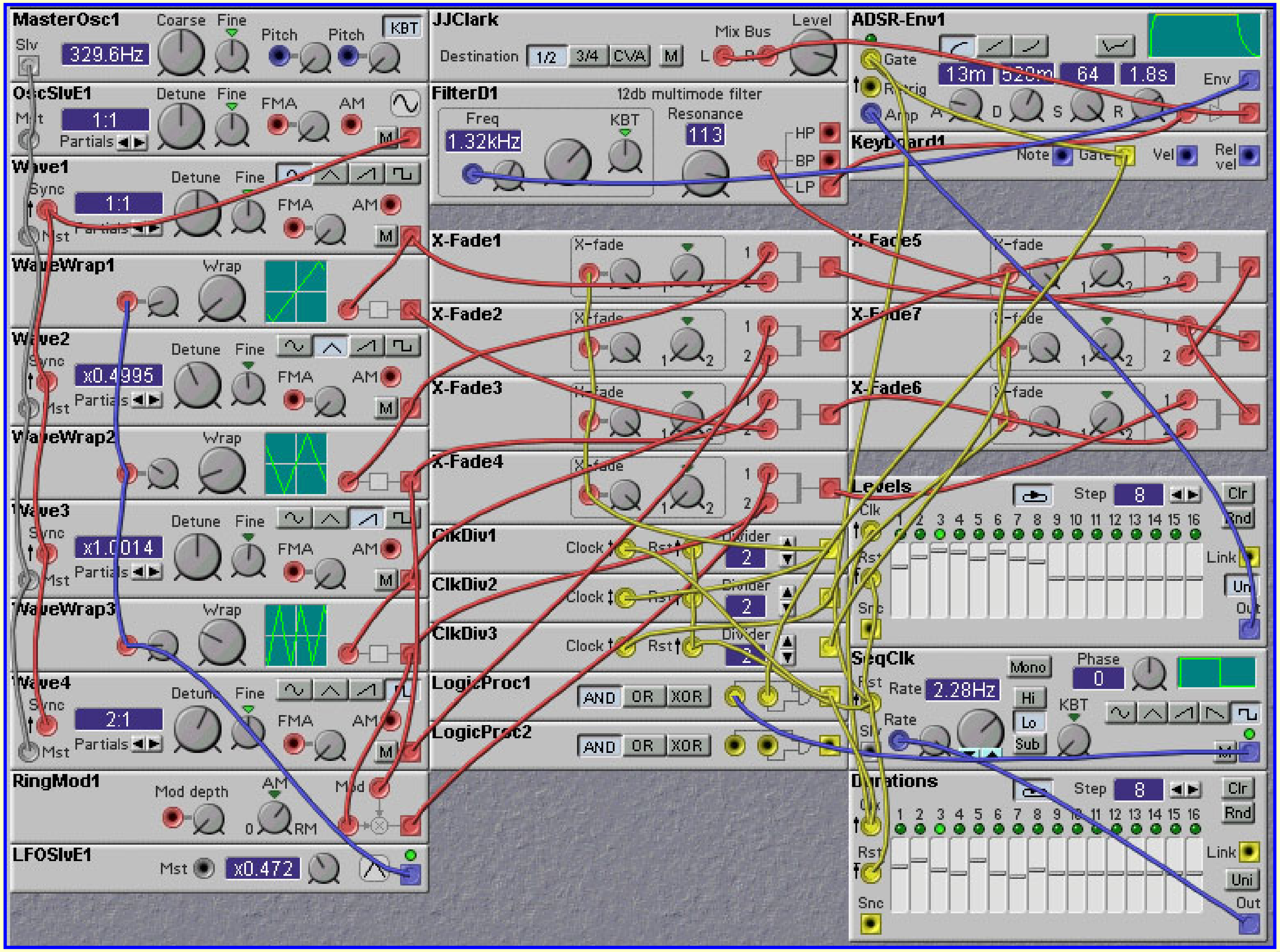


Figure 2.12. Wave Sequencing patch (J. Clark).

Note the use of oscillator sync in the above patch. This is to model the fact that in wavetable instruments such as the Prophet VS and Korg Wavestation, the generation of the waveforms is implemented by scanning of waves in the wavetable, and crossfading between waveforms is acomplished by interpolation between different wavetable entries. Thus each waveform is effectively synchronized, and start at the same point.

In the patch above a 3-bit binary counter is constructed from clock divider modules. This counter controls the crossfading of the waveforms in a binary tree. Unfortunately, this approach requires that the crossfades be done quickly, so there is no possibility for smoothing of the crossfades. The next patch, designed by Rob Hordijk, uses a different scheme for crossfading, which permits smoothing of the crossfades. The amount, or smoothness, of the crossfading from one step to the next can be set by the programmer. An abrupt crossfade will give more high frequency components than a smooth crossfade.

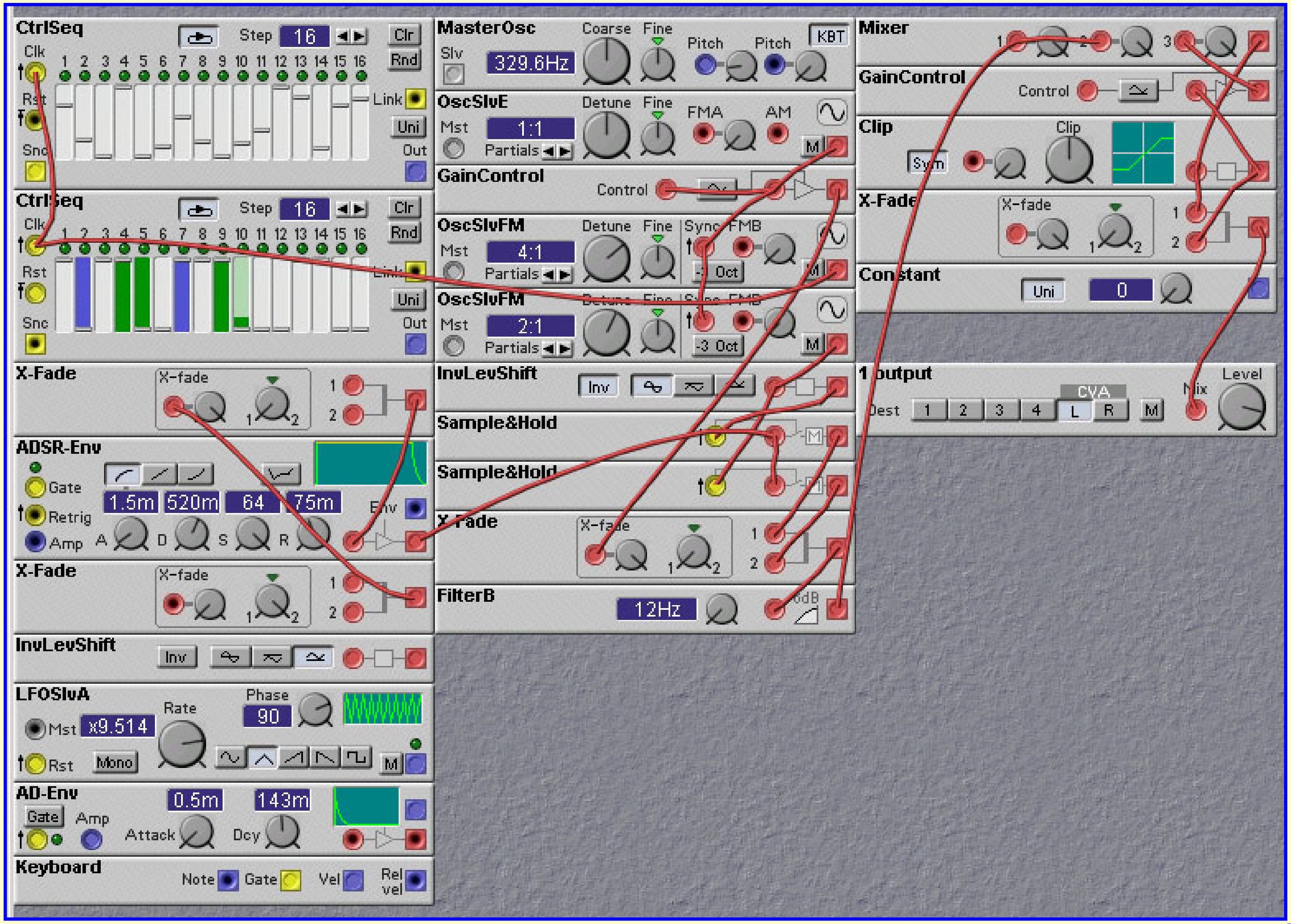


Figure 2.13. Wave Sequencing patch that allows smoothing of crossfades from one wave to the next (R. Hordijk).

The crossfader module can be used to interpolate between a current and a previous output of a control sequencer to create a smooth wavetable oscillator. It proved that using the first quadrant of a sine^2 function to interpolate the transitions sounds the most smooth. Nice, as now only three DSP-cheap sinewave slaveoscs, a gain controller, two S&H's, an inverter, the crossfade mixer and a control sequencer to hold the table are needed to do the trick.

Using two (or more) control sequencers allows for fades between waveforms. Its quite a cheesy sound, so to make the sound a bit grittier a fuzz is added.

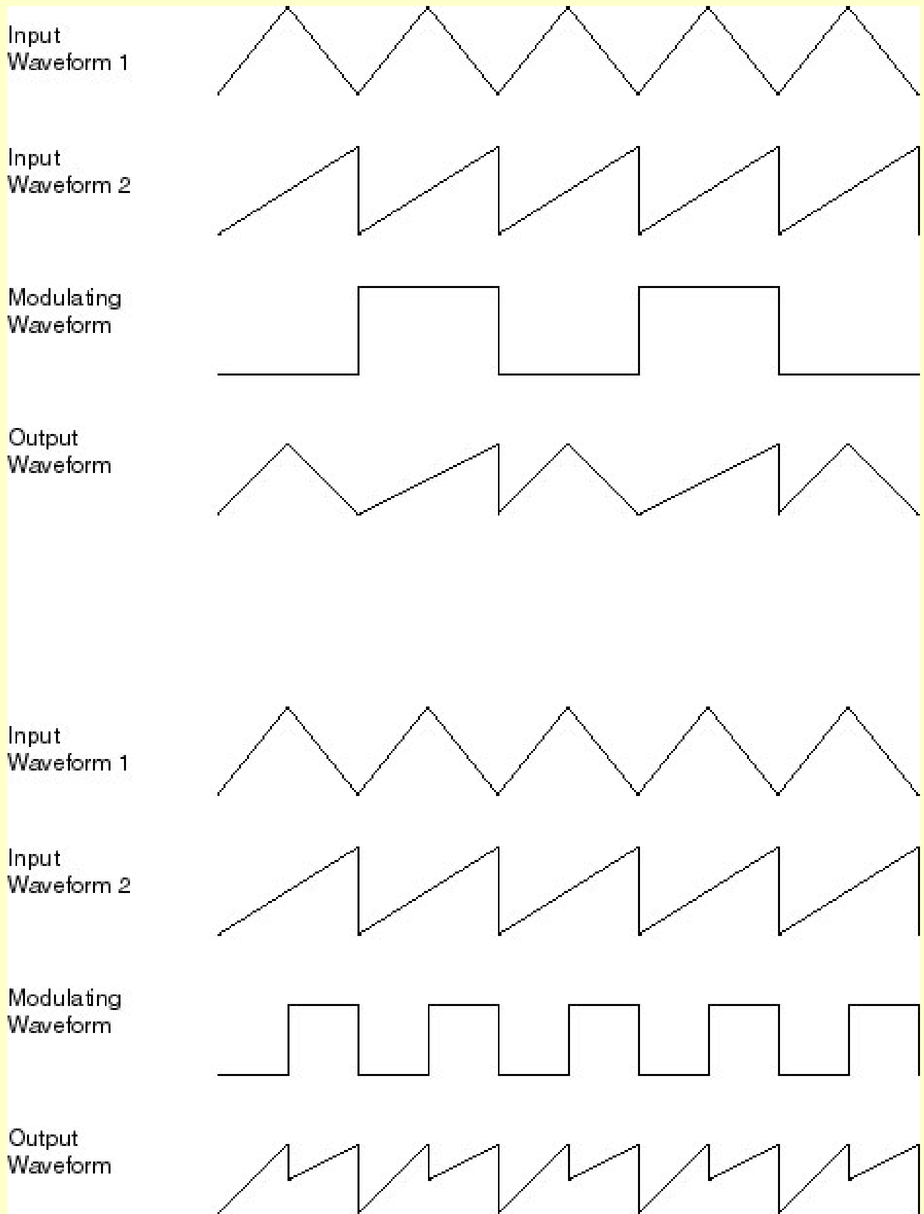
The control sequencers give up around E5 or so, from there up the patch starts to freak out. Which can be quite nice when e.g. controlled from an external sequencer.

2.6 Audio-Rate Crossfading

A useful technique for obtaining interesting timbres involves the use of crossfaders being modulated at audio rates. A crossfader combines two input signals, s_1(t) and s_2(t) through a weighted sum controlled by a modulating signal, m(t), as follows:

s_out(t) = K*[(1+m(t))*s_1(t) + (1-m(t))*s_2(t)] = K*[m(t)*s_1(t) - m(t)*s_2(t) + s_1(t) + s_2(t)]

This is equivalent to amplitude- (or ring-) modulation of each input signal by the modulating signal, inverting one, and summing the results with the unmodulated inputs. One could get the same timbre by using two ring-modulators and a mixer module. The advantage of using the crossfader approach is that it is slightly more efficient, and is often easier to understand conceptually the resulting waveform. This is especially true when the two input signals coming from synchronized oscillators and have frequencies that are integer multiples of the modulating signal's frequency, and when the modulating signal is a square wave. In this case alternate cycles of the output waveform will be copies of single cycles of the two input waveforms. Some examples of the resulting waveforms are shown in the figure below, for the case where one input signal has a triangle waveform and the other a sawtooth.



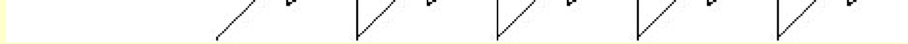


Figure 2.14. Two examples of waveform modification using audio-rate crossfading.

A patch that implements this simple audio-rate crossfading idea is given below. The two input oscillators are synced, so you can alter the harmonic structure by adjusting the frequency of the synced oscillator. Try changing the oscillator waveforms to hear the sound produced in the various cases.

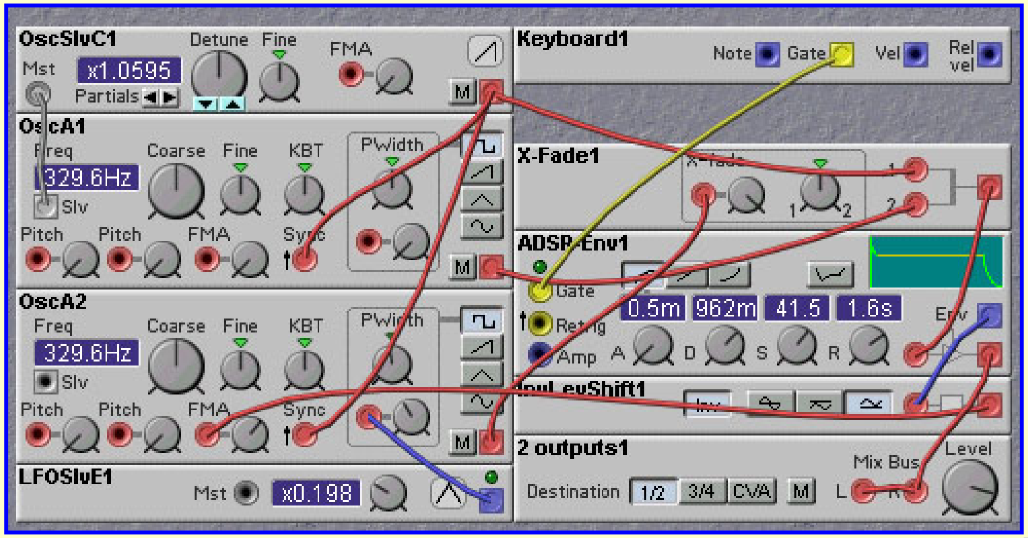


Figure 2.15. Audio-rate crossfading patch (J. Clark).

2.7 Wave Terrain Synthesis

Wave Terrain synthesis is an extension of wavetable synthesis to the use of 2-dimensional wavetables. Now a true 2-D wavetable would require a very large amount of memory, so wave terrain synthesis seeks to save on memory by defining the 2-D wavetable with a surface, called the *terrain*, defined by a mathematical operation called the terrain function. Two 1-D wavetables are then used to define the coordinates on this 2-D surface. If we refer to the two 1-D waves as x(t) and y(t), where t is the wavetable index, then the output of the terrain synthesis is obtained by acting on x(t) and y(t) by the terrain function, z(x,y). The signals x(t) and y(t) define an *Orbit* in the terrain, which selects which values to be output.

Wave Terrain synthesis is described in the book *Computer Music Tutorial*, by Curtis Roads, pp.163-167. In an example of its usage, he implements the following terrain function, and scans it with two signals, x and y:

$$z = (x - y) * (x - 1) * (x + 1) * (y - 1) * (y + 1)$$

A Nord Modular patch based on this example has been created by *Chet Singer*, and is shown below.

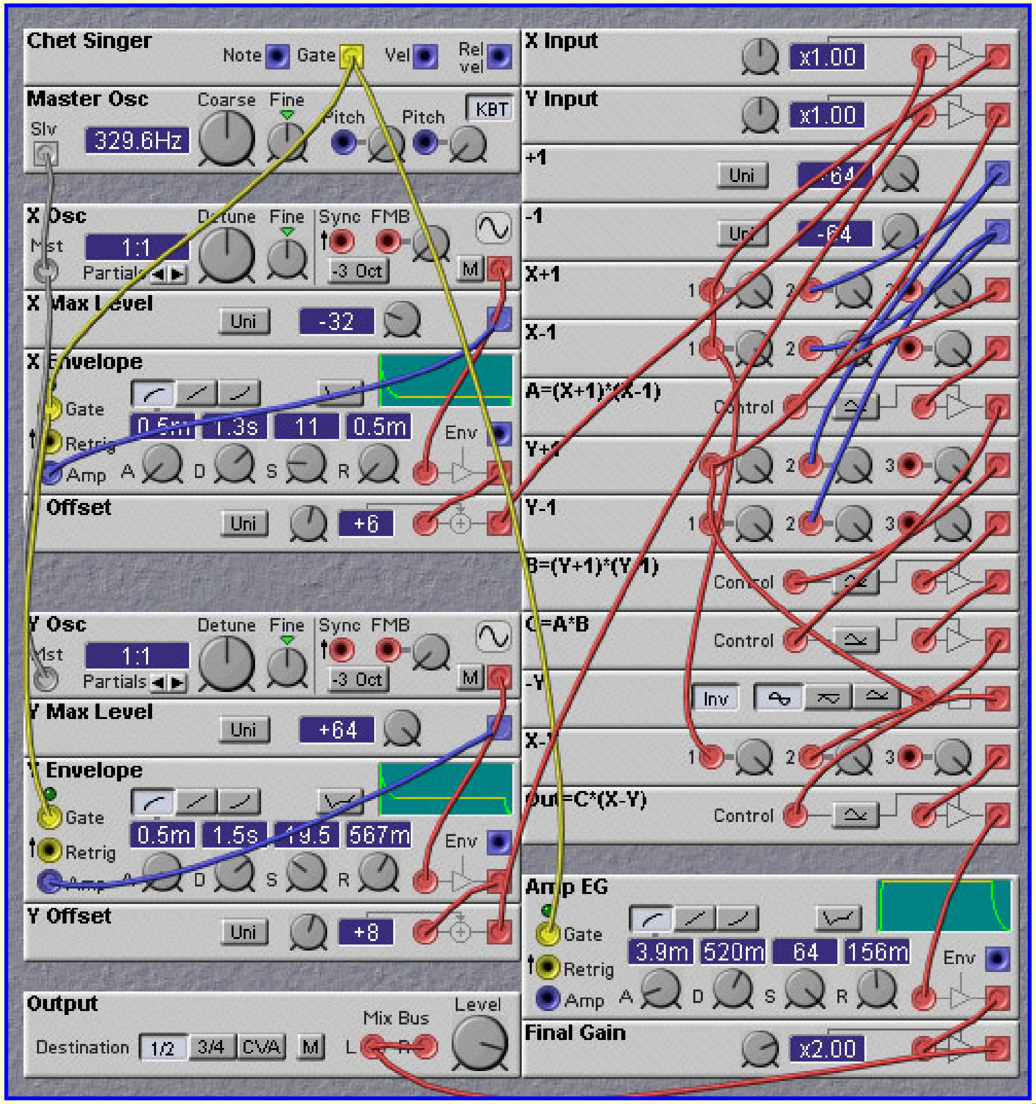


Figure 2.16. A simple Wave Terrain patch (C. Singer).

In this patch the two 1-D waves which define the orbit on the terrain function are enveloped sine waves. The resulting orbit traces out two circular paths connected by a spiral on the terrain surface. At the beginning of the envelope, the path follows the outer circle, then quickly moves to the inner circle during the sustain phase. Upon the release of envelope the orbit moves to the center of the circular paths and stays there. The circles are offset somewhat from the center of the terrain (see the "Offset" control modules in the patch). This adds some assymmetry to the sound and makes it more interesting.

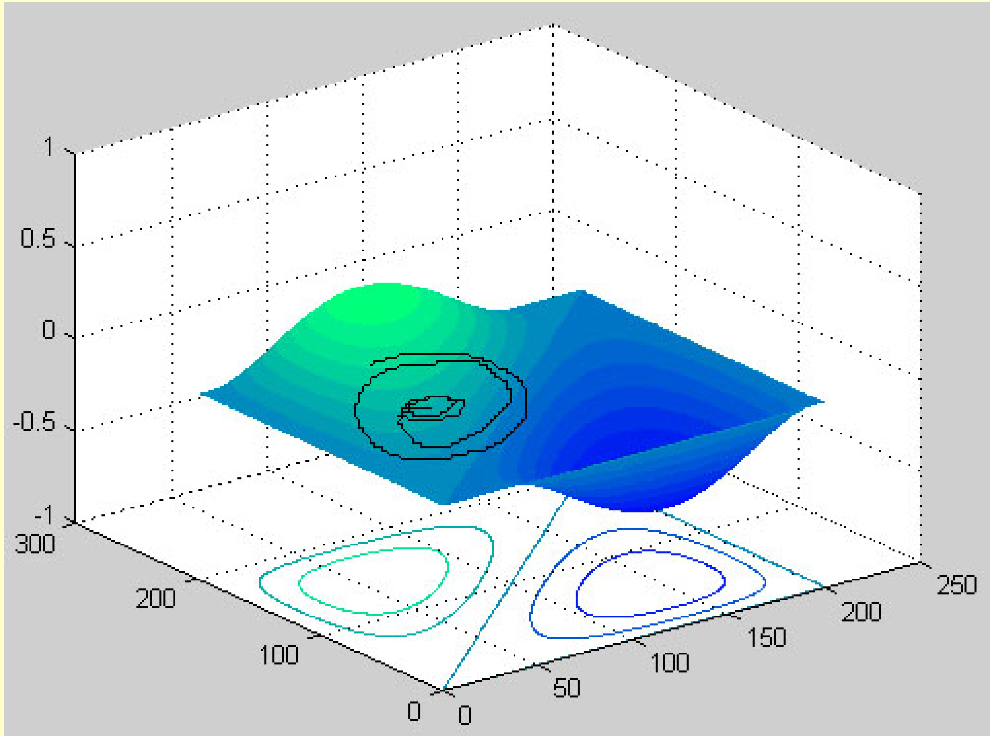


Figure 2.17. The Terrain function used in the above patch, with a typical circular orbit.

2.8 VOSIM

For a description of the VOSIM technique, with Nord Modular patch examples, read Rob Hordijk's tutorial on the subject, http://www.clavia.se/nordmodular/Modularzone/VOSIM.html.

2.9 FOF (Fonction d'Onde Formantique) Synthesis

FOF synthesis is a technique for producing waveforms with a specific spectral shape. It was developed by Xavier Rodet at IRCAM. Similar to the VOSIM technique, sounds are produced from a sequence of *excitation* pulses in the form of enveloped sine waves. The rate of generation of the excitations specifies the fundamental frequency of a formant region. Harmonics of this fundamental are also present, and their amplitude is determined by the shape of the envelope that is applied to each sinusoid in formaing the excitation pulses. The shorter this envelope, the more gradual the falloff in harmonic amplitude in the frequency domain. Lengthening the envelope, on the other hand, reduces the harmonic amplitudes quickly in the the frequency domain, narrowing the width of the formant region. Other details of the local envelope shape, such as the rise and fall rates, affect the shape of the harmonic amplitude curves in subtle ways, allowing for precise control of the harmonic structure. Of course, the shape of the local envelope can be changed over time, permitting dynamic timbres to be produced. Typically, many such excitation waveforms are added together, each providing a different formant region. In this way vowel-like vocal sounds can easily be created, although non-vocal sounds are also possible.

The following patch, designed by *Per Villez*, implements a basic FOF synthesis system. A slightly more complex patch, FoFGen01Cello02.pch by the same designer, shows the ease with which FOF synthesis can model string instruments.

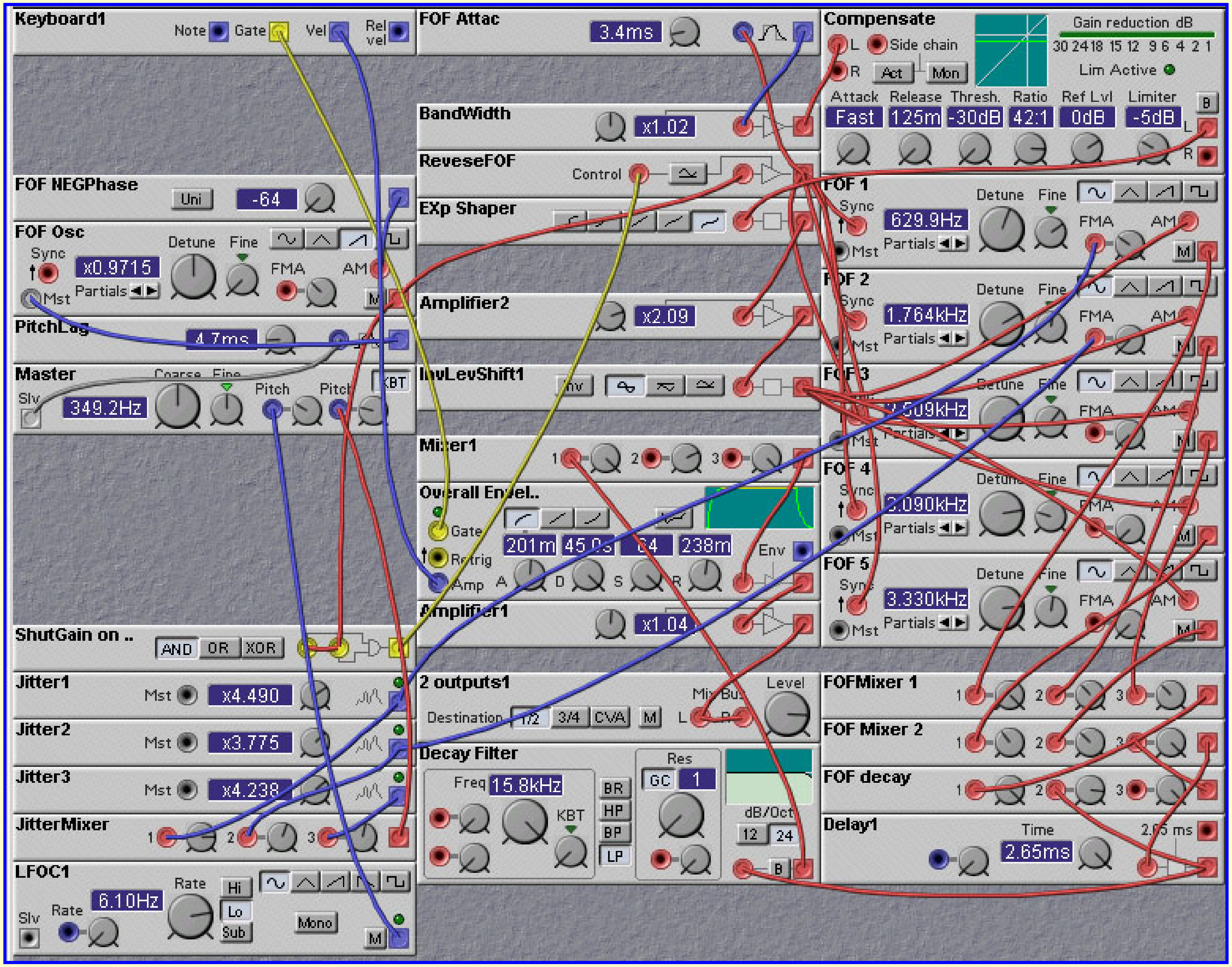


Figure 2.18. A FOF patch (P. Villez).

2.10 Granular Synthesis

Granular synthesis is similar in many ways to the VOSIM and FOF synthesis techniques described above. A grain is a short piece of a waveform (which can come from any source, but is typically a sample stored in a wavetable) which has a short-time envelope applied to it. Each grain is generally 10-100 milliseconds in length. Many individual grains are combined with various relative timings and overlaps to create a composite "cloud" of sound. The individual grains can be modified by altering their envelope (lengthening or shortening, changing the rise and fall time, etc) and changing the frequency of the grain waveform (by changing the scan rate of the wavetable, for example). The relative positions of the grains in the cloud can be altered as well. All of these changes can either be deterministic or random.

The total number of grains, the grain pitch, the inter-grain interval (or overlap if interval is negative), grain envelope shape (attack and decay), and grain length are commonly altered in a granular synthesis algorithm.

The following patch, by *Pelle Dahlstedt*, illustrates how Granular synthesis can be implemented on the Nord Modular.

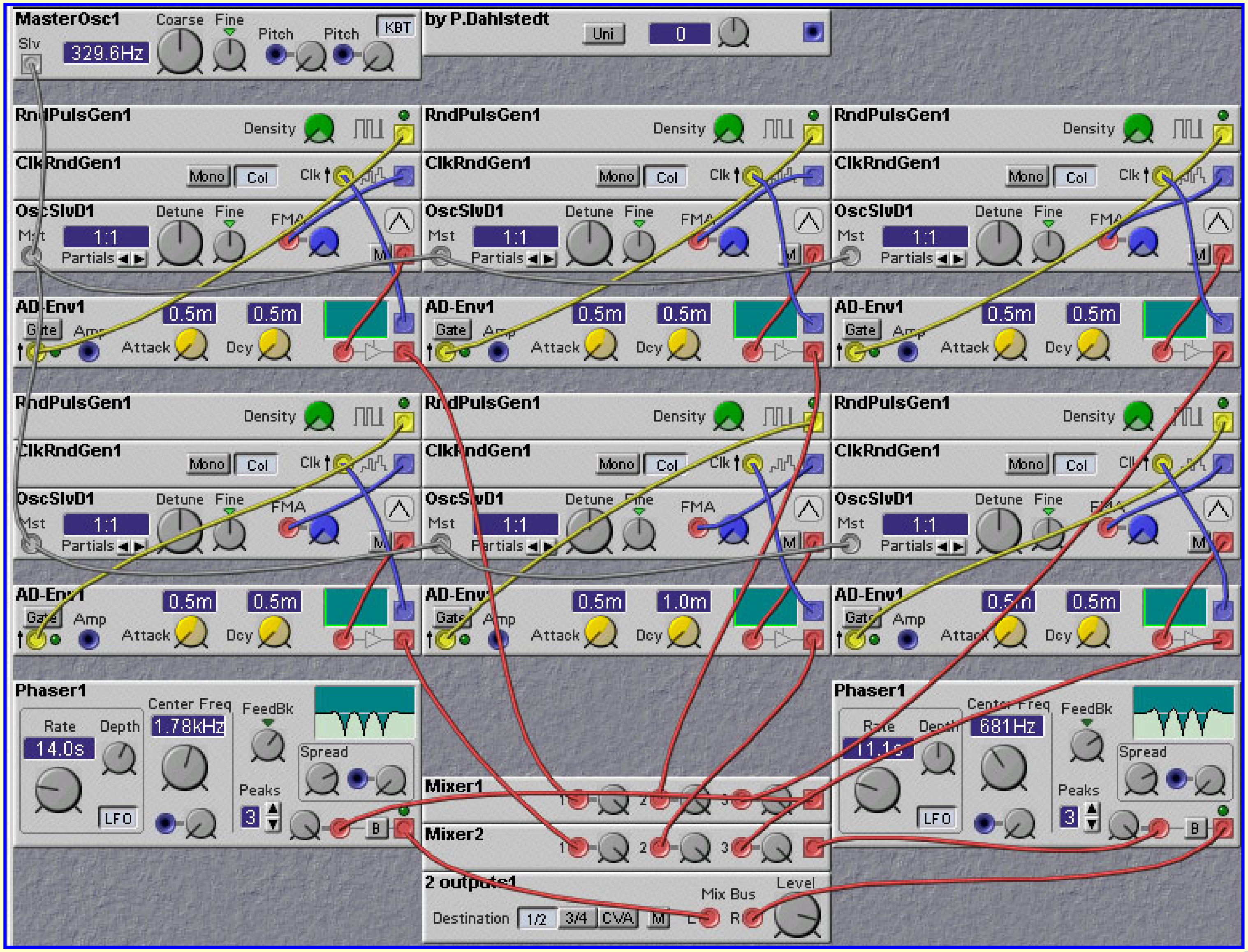


Figure 2.19. A Granular synthesis patch (P. Dahlstedt).

[Prev (Introduction)][Next (Filter Techniques)][Table of Contents]

Chapter 3. Filter Techniques

Much of the character of a given synthesizer sound comes from the filters that are used to sculpt the spectrum of the sound. While the Nord Modular provides a wide range of different filter modules, one can obtain an even wider array of filtering possibilities by applying various patch design techniques.

3.1 Resonant Filters as Oscillators

Filters need not be used only for filtering. They can (in some cases) also be used to generate signals.

The Classic filter on the Nord Modular can be made to oscillate if its resonance setting is high enough. The frequency of the oscillation is the same as the cutoff frequency, and can therefore be controlled with the cutoff frequency knobs, the key-follow, and the frequency control inputs. The resulting sound is a sine-wave. Non-sinusoidal waveforms can be obtained by passing the filter output through a wave-shaper, wave-wrapper or overdrive distortion modules.

With digital filters you have to give the filters a bit of a push to get them oscillating. This is true for analog filters as well, but in analog systems there is always a bit of noise around to do the job. In a digital system you have to explicitly provide this. There are a couple of ways to do this:
- add a small amount of noise to the filter input.
- add a bit of the envelope or note gate to the filter input.
It doesn't take much to destabilize these self-oscillating filters, so these extra inputs can be at a very low level, and hence should not be audible.

The patch shown below uses two self-resonant F-filters (classic filters) with their resonant peaks slightly separated (controlled by an LFO). A wavewrapper is used to add some harmonics to the sinusoidal output of the filters.

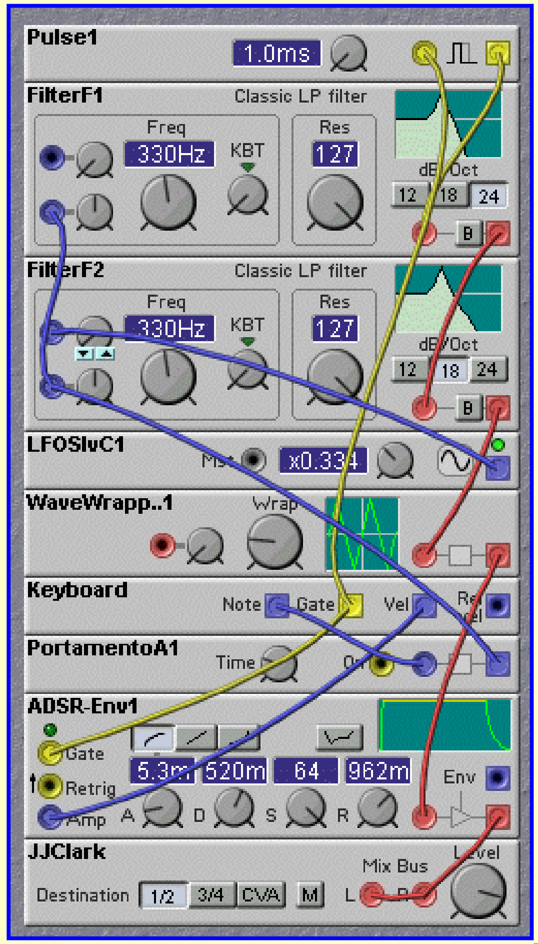


Figure 3.1. Resonant filter oscillator patch (J. Clark).

A soft attack can be generated without an envelope generator by using a filter with a resonance control input, and connecting the note gate signal to the resonance input. When the note gate is low, the filter resonance will be too low to support oscillation. When the gate goes high, the filter will begin to self-oscillate, but will take some time to reach its full amplitude.

The patch shown below implements this idea. We use the E-filter, as this provides a control input for the resonance level. Unfortunately, the E-filter does not self-oscillate, so we add a little bit of extra positive feedback (via the mixer) to make the filter self-oscillate.

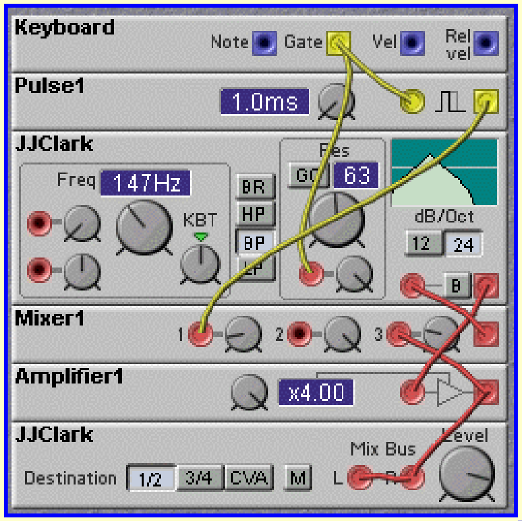


Figure 3.2. Resonant filter resonance controlled soft attack patch (J. Clark).

Resonant filters can also make good sources of drum sounds. They need not be self-oscillating, but just have a high enough resonance level so that they "ring" for a while. The following patch shows this. The resonance is set to a fairly high level and the filter input is connected to a pulse derived from the note gate. The pulse causes the filter to ring. The higher the resonance level, the longer the ringing period. The filter rings for longer times at low cutoff frequencies than at high, so we adjust the resonance level with the note output to compensate.

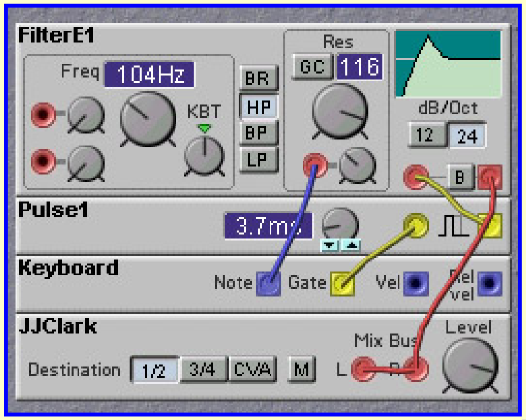


Figure 3.3. Resonant filter used to produce drum sounds (J. Clark).

The same ringing resonant filter technique can be applied to the generation of envelopes.

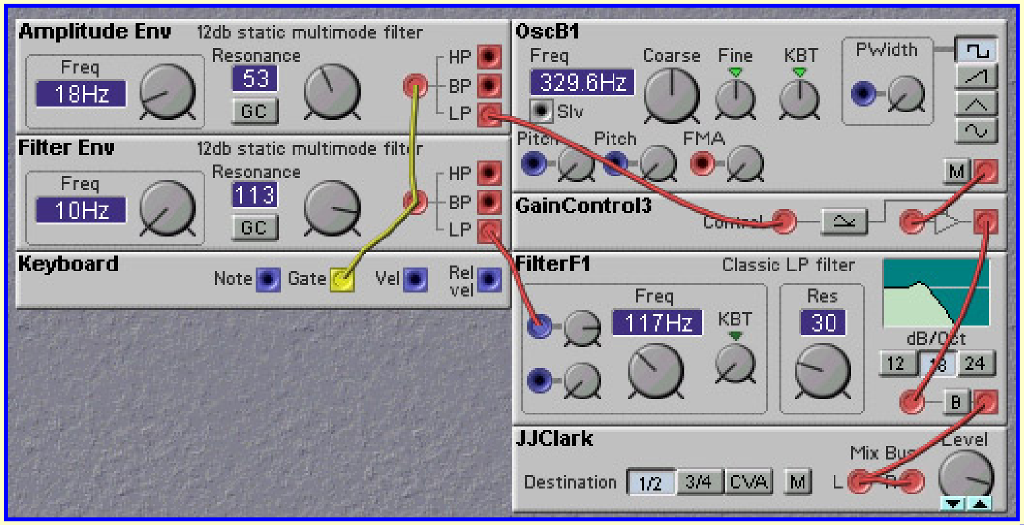


Figure 3.4. Resonant filter used to generate envelope waveforms (J. Clark).

3.2 Serial and Parallel Filter Techniques

There are a small number of variable filter modules provided with the Nord Modular, which might seem to limit the possible ways to filter sounds. Indeed, the Nord Modular filters to a large extent define the distinctive Nord Modular 'Sound'. But, remember that the Nord Modular is a *Modular* synthesizer, and so we do not have to just use a filter all by itself! We can combine filters in many different ways to produce new types of filtering action. In this way we can obtain patches that sound different than the normal ones.

There are two basic structures to be used when combining filters - serial and parallel. We will begin by discussing serial filter structures. In a serial filter structure, filters are connected in chains, with the output of one feeding into the input of another.

We can make filters whose cutoff slope changes with frequency. For example, we can have a filter whose slope starts out as 12dB and then increases to 24dB. This is done by having a 12dB filter followed by a 24dB filter, with the cutoff frequency of the second filter being higher than that of the first. The effect of this type of filter is subtle compared with a normal 12dB filter but can be useful when you want a 12dB filter sound, but also want to reduce the buzziness caused by the very high frequency components.

Perhaps ones of the best uses of serial structures is to alter the character of a filter's resonance. For example, one can construct a 'two pole' 12dB filter with a serial connection of two 6dB filters. Resonance can be obtained by feeding back the output (inverted) to the input. One can also obtain a resonant 12dB filter by stringing together four 6dB filters in series, feeding the output of the fourth filter back to the input, but taking the overall filter output only after the second filter. This will result in a 12dB filter with a resonant characteristic different than the first type of 12dB filter. The resonance characteristics can also be altered by modifying the feedback path. Putting filters, delays, or distortion elements in this path will all change the sound. The following patch allows a comparison between the resonance of the different Nord Modular filters, as well as to a custom filter constructed using four 12dB filters, with a feedback path, and output taken from the first 12dB filter. You can hear that the filters with the feedback taken from the higher order sections (beyond the 12dB point) all have a more pronounced resonance than the simple 12dB filter.

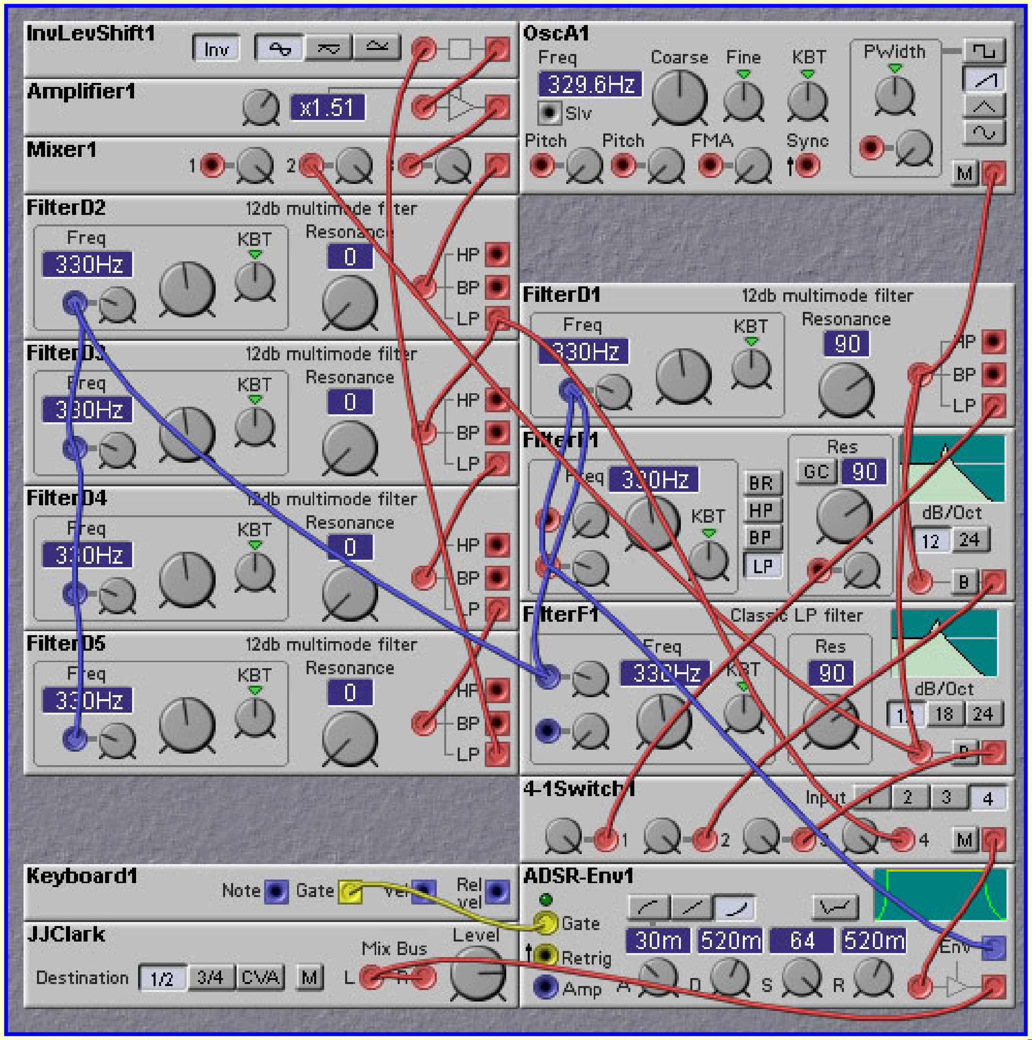


Figure 3.5. A comparison of four different 12dB resonant filters (J. Clark).

Serial structures are also useful for adding in multiple resonances to the filter's frequency response. We can do this by stringing together a number of resonant filters in series. Each filter should have a different cutoff frequency. This will result in a falloff that is more rapid than any filter taken alone, but will also result in multiple resonance peaks. This is useful for implementing 'body' resonances, such as found in physical instruments. Unlike real instruments, we can shift these 'body' resonances dynamically, with LFO or Envelope control signals, for example.

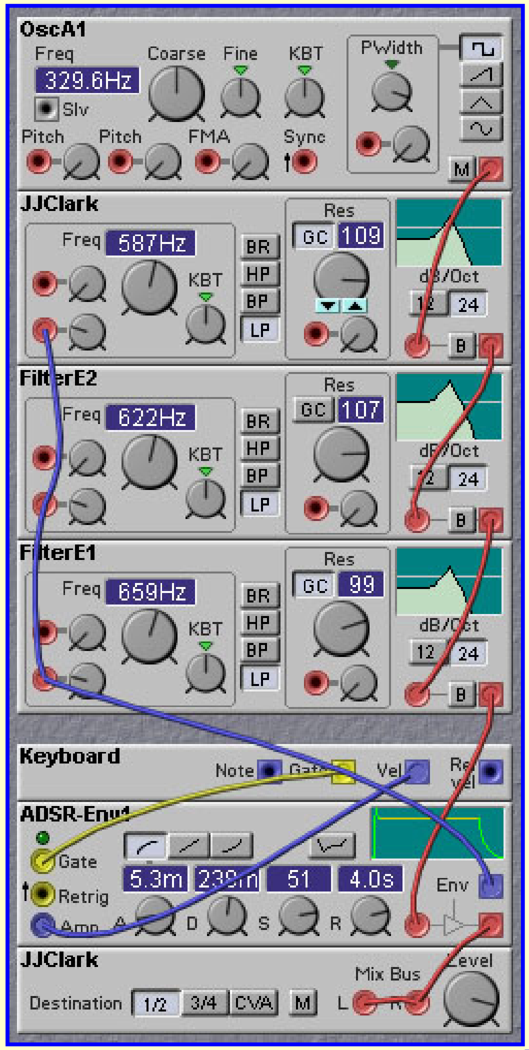


Figure 3.6. Slightly detuned serial resonant filters (J. Clark).

By using a set of filters in parallel (with each filter having the same input, and summing their outputs), one can obtain spectral characteristics different than that of the common second order resonant filter. For example, connect two lowpass second order filters with slightly different cutoff frequencies. Use an intermediate-to-high resonance level on each filter. The slightly different frequency components that are emphasized by the resonant peaks will interfere with each other, producing a beating or phasing sound. This effect is similar to that obtained with serially connected filters, but without the drawback of increasing the cutoff slope. Thus the resonant peaks can be stronger than in the serial configuration.

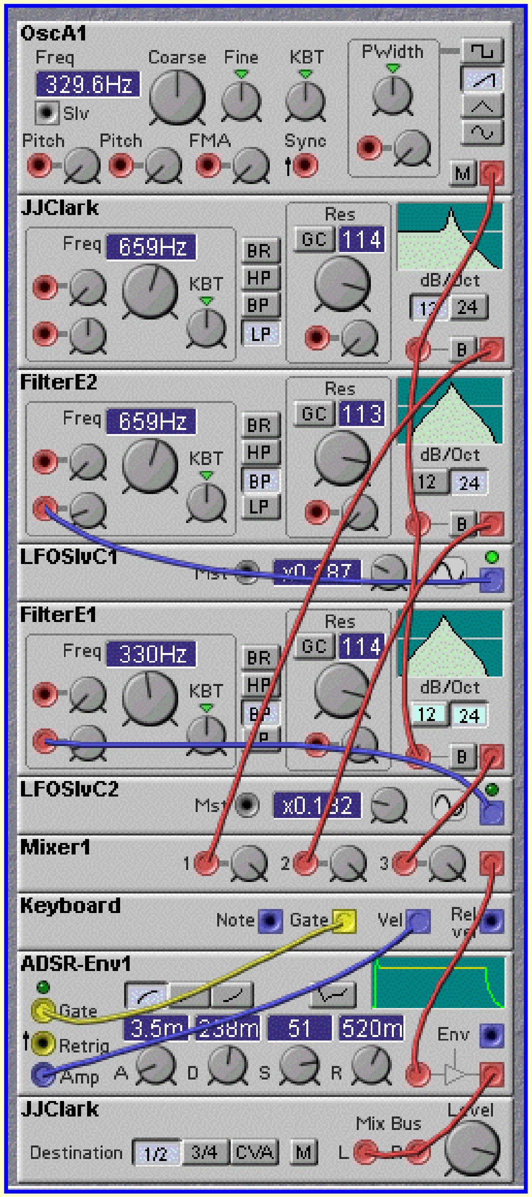


Figure 3.7. Slightly detuned parallel resonant filters (J. Clark).

Parallel structures can be used to adjust the slope of the cutoff region. The non-static filter modules provided in the Nord Modular come in only 12, 18 and 24 dB varieties but we can construct filters with cutoff slopes that fall in between these values. This is done by combining filters in parallel that have different cutoff frequencies. As one filter drops off, there will be a corresponding reduction in the overall spectrum, but some of the other filters will still be passing these frequencies, so that the overall falloff rate is less than the individual filters. An example is shown in the following patch, where a (more-or-less) non-static 6dB filter is constructed from three 12 dB filters. As the input signal frequency passes the cutoff frequency of the filter with the lowest cutoff, the amplitude will start to drop off, but the other two filters will still be passing this frequency. As the signal frequency passes the cutoff of the filter with the next highest cutoff frequency, the signal amplitude will again start to drop, but some of the signal will still be passed by the third filter. Finally, when the signal frequency passes the cutoff of the third filter, the cutoff slope will be that of the third filter, 12dB. Thus the overall effect is of a filter with a reduced falloff rate over a certain range of frequencies. One can get an increased accuracy and increased range of applicability by using more filters, with cutoff frequencies closer together.

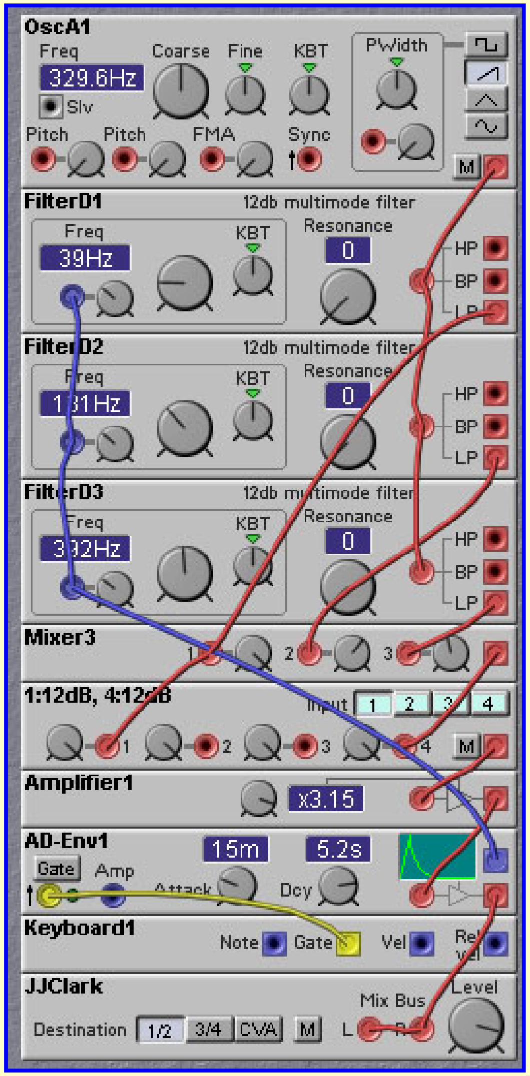


Figure 3.8. Constructing filters with a lower cutoff slope by combining filters in parallel (J. Clark).

One common use of parallel filters is to split a sound into various frequency bands and process them separately, then recombine them. For example, you can distort the separate signals by differing amounts. The Roland SE-70 has a band chorus which breaks the sound into two bands (low pass and high pass filter) and allows different chorus settings to be applied to each band. Another possibility is to use phase shifters for each band. A good way to get a nice wide stereo sound is to direct different frequency bands to the left and right outputs. Some examples of this can be found in the chapter on effects.

3.3 Audio-Rate Cutoff Frequency Modulation

The cutoff frequency control input on the Nord Modular's E-filter can take an audio signal. So, the obvious thing to try is to feed an audio signal into this input! What is the effect of this? Well, it depends on the nature of the audio input and on the filter characteristics. The effect can be seen to be similar to amplitude modulation of the filter input signal. To see this, consider an ideal lowpass filter, with no resonance peak, and an infinitely sharp cutoff. Consider a signal that is rich in harmonics being fed into the filter. The filter will cutoff all the harmonic components above the current cutoff frequency, and pass through all the ones below. Now, suppose we lower the cutoff frequency a bit. Some of the harmonic components that were previously passed through by the filter will now be cutoff completely. If we then increase the cutoff frequency back to where it was, then these components will be passed through once again. If we periodically vary the cutoff frequency in this manner, then it should be clear that the amplitude of the harmonic components whose frequencies lie in the range of variation of the cutoff frequency will appear to be periodically amplitude modulated. Because of the sharp cutoff of the filter, the amplitude modulation will have a `pulse-wave' shape. If we use a filter will a gradual cutoff (such as the Nord Modular filters) the amplitude modulation will still have a pulse-wave shape, but with rounded corners. The more gradual the cutoff, the more rounded the amplitude modulation shape. If there is more than one harmonic component in the cutoff frequency range of variation, each of these components will have some amplitude modulation, but this modulation will be different for each component. Components with frequencies near the far edge of the cutoff frequency range of variation will have modulation shapes that are more pulse like, while components with frequencies near the center of the range will have modulation shapes that are almost square. Harmonics with frequencies outside of the range will have no amplitude modulation. Thus, the sound of the audio-rate cutoff frequency modulation will sound somewhat different than pure amplitude modulation, since in the latter case, all harmonics will have the same modulation shape.

The graph below shows this effective amplitude modulation. The graph shows the output of a second order filter in response to a 1KHz sinusoidal input. The cutoff frequency control is being fed with a 100Hz sinusoidal input. The different plots depict different modulation levels, which translates into different cutoff frequency ranges. The nominal cutoff frequency is set to 1KHz.

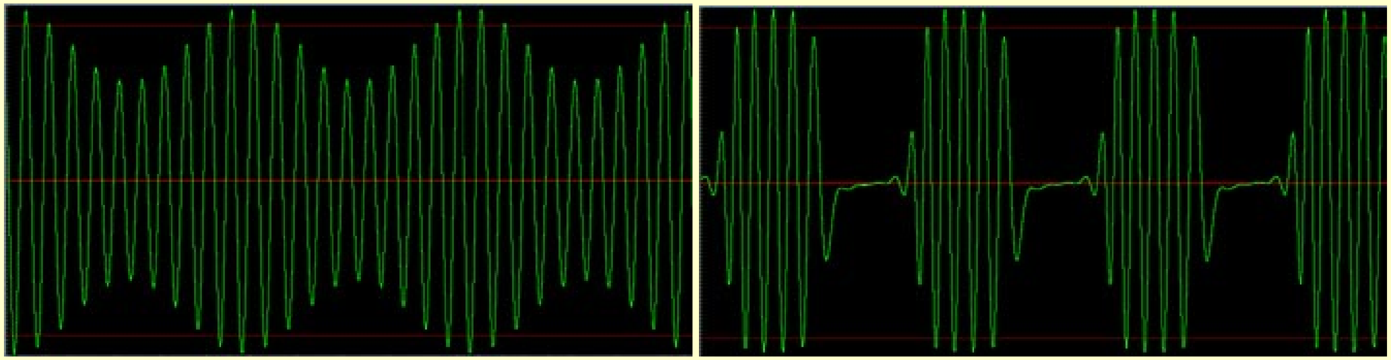


Figure 3.9. Audio rate filter FM. Effect of modulation level. Left - modulation level = 5, Right - modulation level = 50.

In the next graph, the modulation level is fixed, and the frequency of the input sinusoid is varied. As the frequency of this sinusoid moves from the center of the range of variation of cutoff frequency to the edge of the range, the effective amplitude modulation shape changes from a rounded square wave to a pulse wave. The modulation level was set to 25 in producing these graphs.

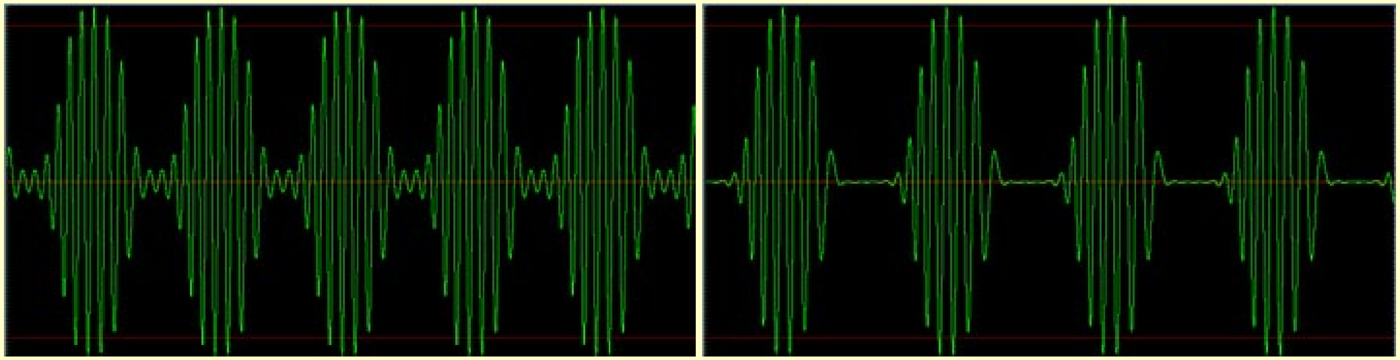


Figure 3.10. Filter FM - Effect of the input signal frequency. Left - input frequency = 1000 Hz, Right - input frequency = 1200 Hz.

Finally, the next graph shows the effect of varying the filter cutoff slope from 18 dB/octave to 24 dB/octave. As the slope gets gentler, the shape of the amplitude modulation becomes rounder. In obtaining these graphs an input frequency of 1KHz was used, with a modulation level amount of 25.

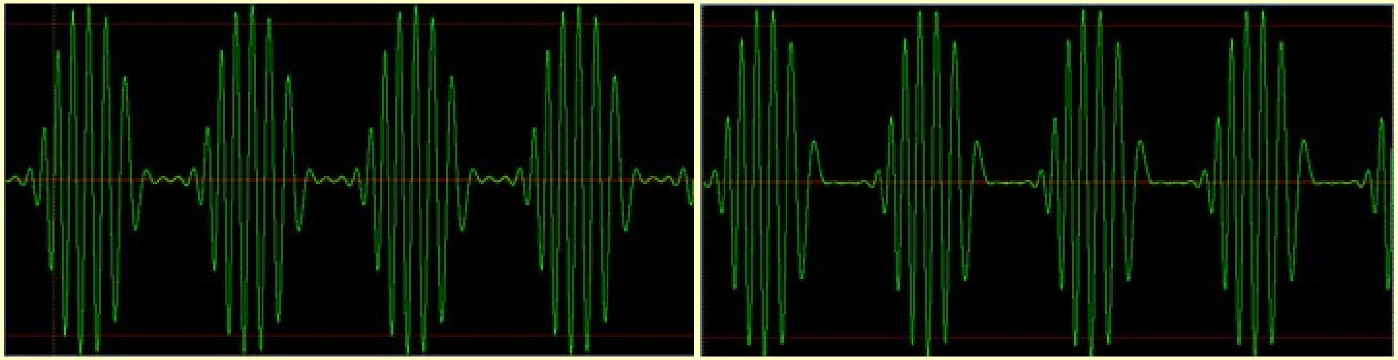


Figure 3.11. Filter FM - Effect of filter cutoff slope. Left - cutoff slope = 18 dB/octave, Right - cutoff slope = 24 dB/octave.

The following patch implements audio rate filter cutoff modulation. Play around with this patch, changing the filter order (and therefore the slope of the filter cutoff), the filter type (from lowpass to highpass), and the filter resonance setting. Also try changing the input waveform type.

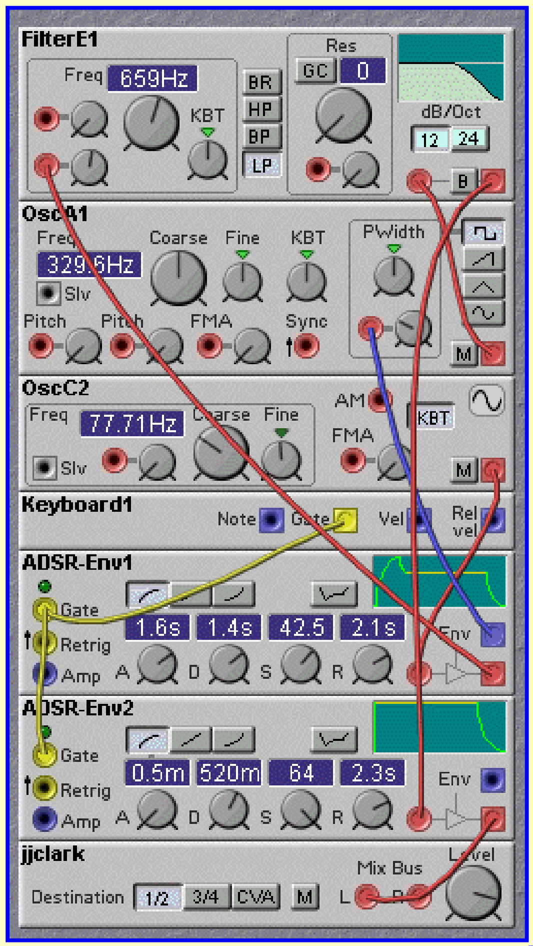


Figure 3.12. Audio-rate filter cutoff frequency modulation patch (J. Clark).

(Those of you with some background in signal processing theory may be wondering how it is that a completely linear system (which the filter is, even if we can modulate its cutoff frequency) can generate harmonics? The reason is that the system is time-variant. A linear time-variant system can generate harmonics, just as a non-linear time-invariant system can. It is only linear time-invariant systems that do not generate harmonics. You should also note that a ring-modulator is also a linear system (or, more precisely, a bilinear system). Doubling the amplitude of (one of) the inputs doubles the amplitude of the output. The ring modulator is a time-variant system, however, from the perspective of one of the inputs.)

3.4 Adding `Analog Feel' to Filters

Some purists argue that digital filters sound lifeless and sterile. While they may be right, there are some things that can be done to improve upon the basic digital filter sound and make them closer to their analog counterparts.

One way in which to provide a better approximation to analog filters is to vary the resonance with the cutoff frequency. For example, the Moog classic filters do not have a constant resonance. Their resonance decreases significantly when their cutoff reaches and goes below about 130Hz. The Modular Moog filters do not self-oscillate below that frequency. Another aspect of true analog filters is that they have nonlinearities (changes in response as the input signal amplitude changes) that can cause distortion, or creation of harmonics, in the output. Finally, the properties of true analog filters change over time, primarily due to variations of component values resulting from changes in temperature. These slow variations cause subtle changes in the response of the filter, and are a big part of the "analog" sound.

The following patch includes an emulation of an analog filter. It uses explicit feedback to implement resonance, rather than using the resonance control of the Nord Modular filter modules. This allows us to insert a clipping distortion in the feedback path, which models one of the most common causes of distortion found in analog filters. This approach to implementing resonance also has the by-product of reducing resonance as the cutoff frequency drops (or increasing resonance as cutoff frequency increases). Clipping is also added after each filter, to emulate the saturation of the analog filter stages. To accentuate the analog feel, we add small random variations to the filter cutoff frequencies, and to the clipping levels. Finally, portamento is added, to give the patch that classic synth sound!

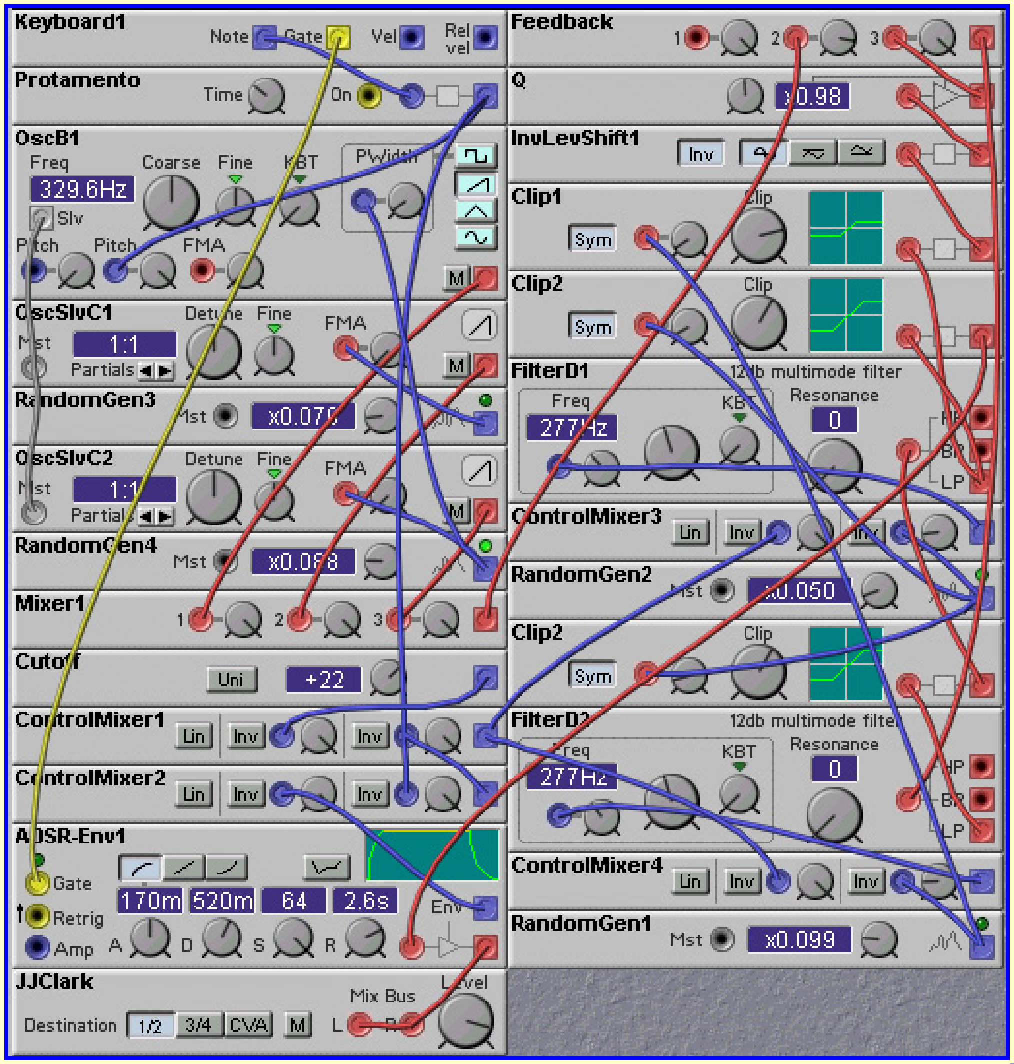


Figure 3.13. An emulation of an analog filter, which includes distortion and variation of resonance with cutoff frequency (J. Clark).

Rob Hordijk has developed a very nice sounding "analog" style filter. His filter avoids the "buzzy" sound produced by many digital resonant filters. The buzziness arises because digital oscillators contain a relatively large amount of energy above 10kHz. Most analog oscillators have less energy in this region, as the analog circuitry itself filters part of it away. Imagine a sawtooth set to 200 Hz, therefore having overtones 200 Hz apart from each other. This means that in the area between 10 kHz and 20 kHz there are 50 overtones present, all crowded together within a single octave! When using, for example, 3 slightly detuned oscillators you're talking about 3 * 50 = 150 overtones all in that one high octave, and all phasing fast with each other. The amplitudes of these overtones are very small, but there are a lot of them and very high sounds are perceived quite well, so there is a distinct buzz in the high. If the cutoff frequency is set to this area the buzziness is increased even more at high resonance levels. The resonance band of a 12 dB filter is a bit broader than that of a 24 dB filter, so the 12 dB filter suffers a bit more from the buzz.

The problem with this buzz is that it can mess up those other sounds that have by nature lots of energy in the same band, notably hihats and cymbals and some dipthongs in the vocals. Thus it is a good practice to filter everything above 10kHz away from all instruments when there are hihats and cymbals in the rhythm track, or if you use vocals from someone with a clear voice. Otherwise these hihats and the s's and t's will drown in the high of the other instruments. Its even worse if the 10kHz+ area gets in a reverb with a very bright tail. That will start to produce lots of noise.

For most synth sounds, especially strings, its not the 10kHz+ area which is important, but the area between 3.5kHz and 10 kHz. So filtering away all above 10kHz but slightly emphasizing the 3.5 to 8 kHz area greatly improves the warmth and depth of stringsounds. A single 6 dB LP filter set to 10kHz won't do the job, the cutoff frequency should be set to 2.5 kHz or less to effectively remove the buzz. Even the cutoff frequency of a 24 dB filter should be set to something like 5kHz. But in both cases you would also lose part of the important 3.5 to 8 kHz area. The most useful solution is to use a dipfilter with a notch around 12 kHz.

The filter is composed of two 12 dB filters that are cascaded to get a 24 dB filter. On the first filter a little bit of the HP output is mixed to the LP output. This is tuned by a MasterOsc module. As it apparently needs some bizarre overexponential control to get everything right, the grey signal is raised to the power of two and mixed with the grey signal to control the amount of HP. This creates a notch at the top end of the spectrum, which does three things:

1) it attenuates the very high end, making the filter less "buzzy ".
2) it reduces the resonance at the top end of the spectrum relative to the rest of the spectrum, especially at high resonance settings. This also makes the sound less buzzy.
3) the notch increases the filter slope slightly.

The messing about with that grey signal is just to keep the notch at the right place, which is tuned to taste by ear.

The second 12dB filter increases the filter cutoff slope to 24 dB. The feedback from the LP output of the second filter increases the bottom end of the spectrum, giving the sound a little bit more guts.

This filter can give good analog bass sounds with even a single sawtooth oscillator.

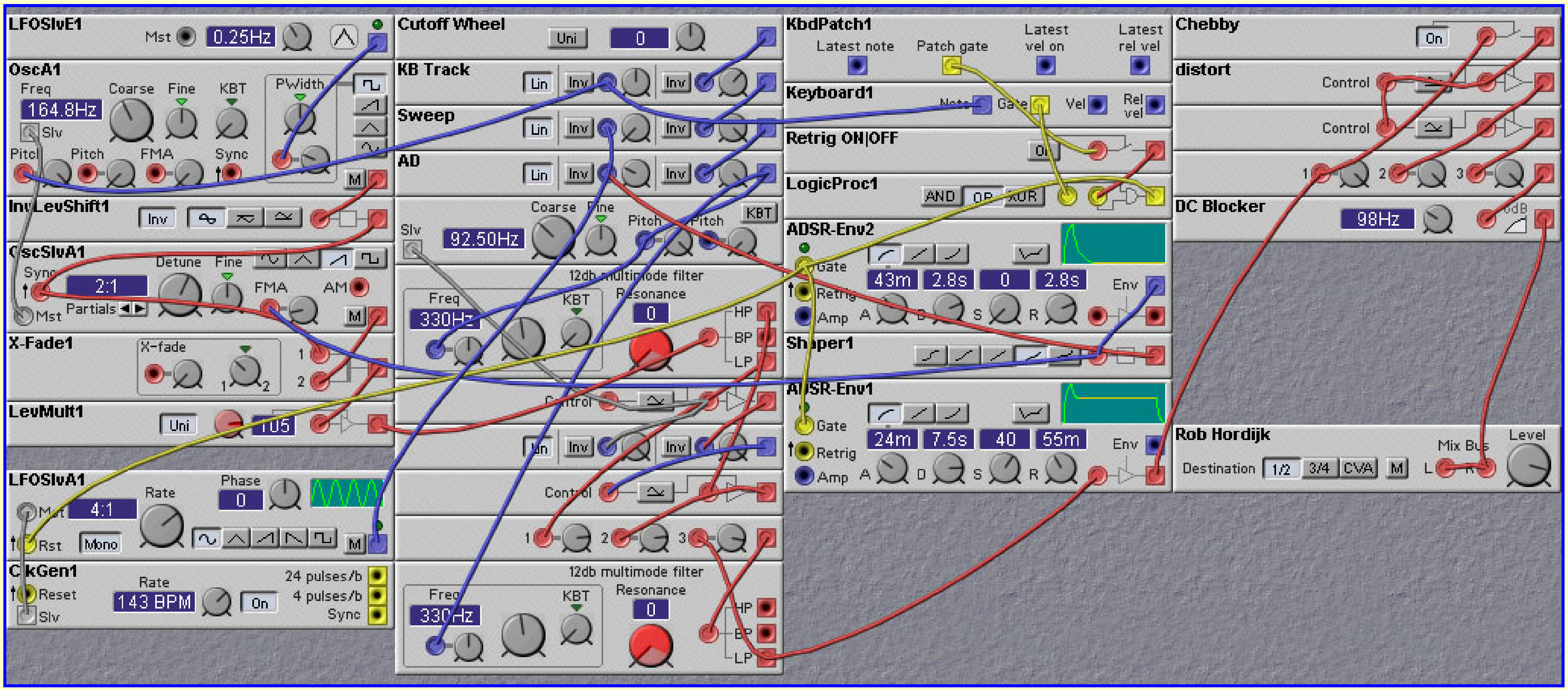


Figure 3.14. A bass patch employing a filter having a high frequency notch to reduce "buzziness" (R. Hordijk).

By filtering away a conflicting band in one of the sounds, especially if that band is emphasized in another sound, you can get much more control. In fact thats what most of postproduction is about and why multiband compressors have become so important. And why professional records sound so good. In well-produced recordings there is hardly ever a moment when a sound is soundwise conflicting with another sound, instead they all seem to emphasize each other to get that snappy drive. Rob Hordijk suggests using the shaper module to make the filter envelope signal reverse exponential. That way the filter stays a bit longer in the higher regions and the env doesn't have to sweep the filter above 10kHz, where all the horrible aliasing noise and hiss lives.

To get a good idea of how the presence of a notch in the filter response improves the sound, listen to the following patch for a couple of minutes. The patch includes a string-like waveform generator and a hi-hat sound. Without the notch on the strings the sound is 'tense' and the HH drowns in the stringsound. With the notch present, the sound is open and the tension is gone.

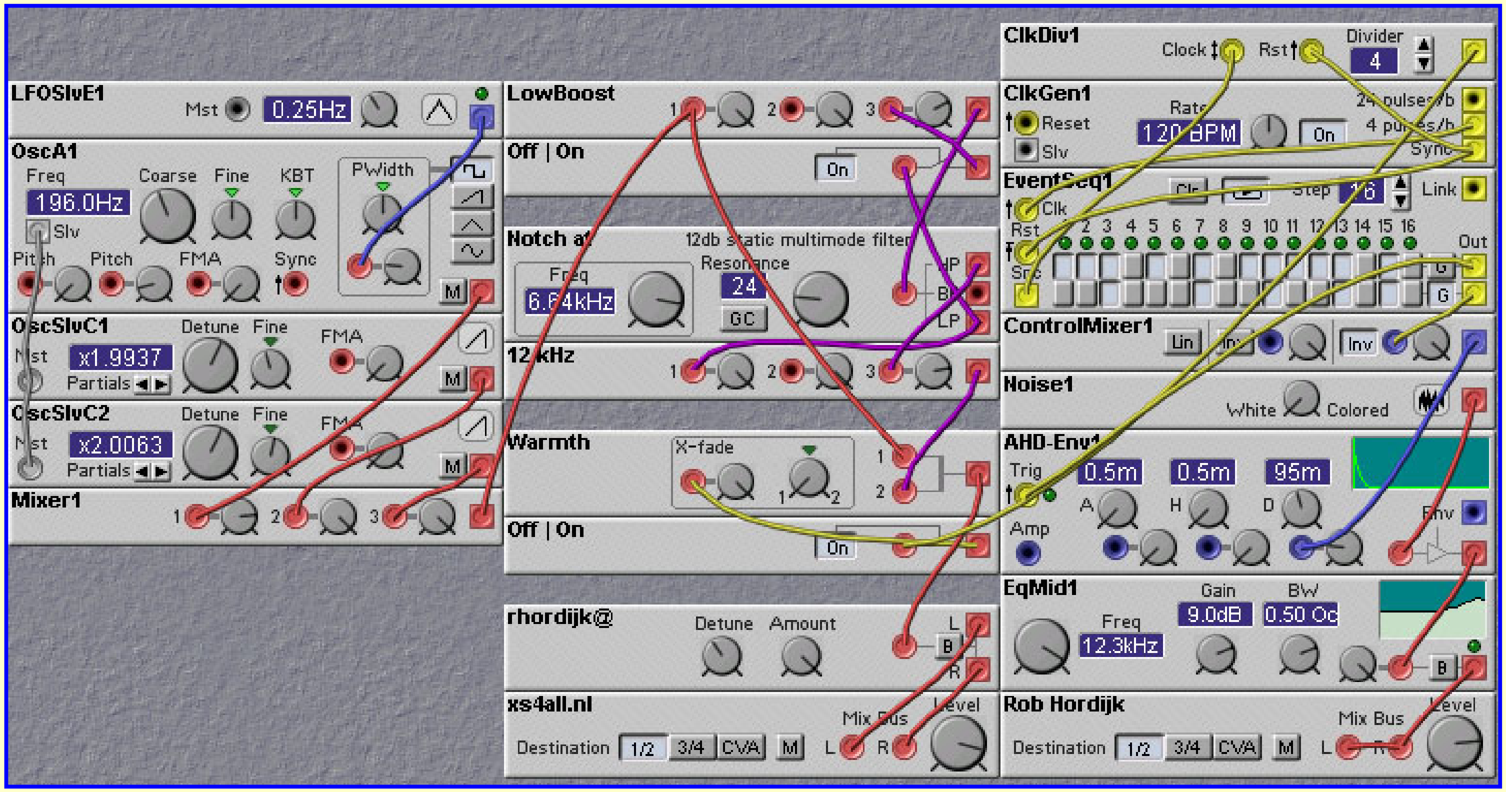


Figure 3.15. A patch demonstrating the effectiveness of high frequency notches in reducing "buzziness" and interference with instuments like HiHats in the mix (R. Hordijk).

In listening to this patch there is also a slight formant effect that you can hear. If the sound is slightly more like 'aah' or 'uuh' it is generally felt as pleasant, whereas if it sounds more 'ooh' or 'iih'-ish then it has a bit more tension. The 'uuh' formant makes the sound neutral, an 'aah' effect asks for attention, 'ooh' makes it severe and 'iih' is like a conclusion. This has much to do with harmonicity, an obscure subject at first, but basically if too many overtones of different sounds and notes start to conflict with each other it starts to feel tense. One can use this to good effect in constructing string patches. Add a vocal filter module to the output and use it to add expressiveness or "emotion" to the sound. For example, use key velocity to modulate the vocal filter from "AA" through "UU" to "OO". Thus, at low velocities the sound will be mellow, while at high velocities the sound will be more tense. This is illustrated in the following patch:

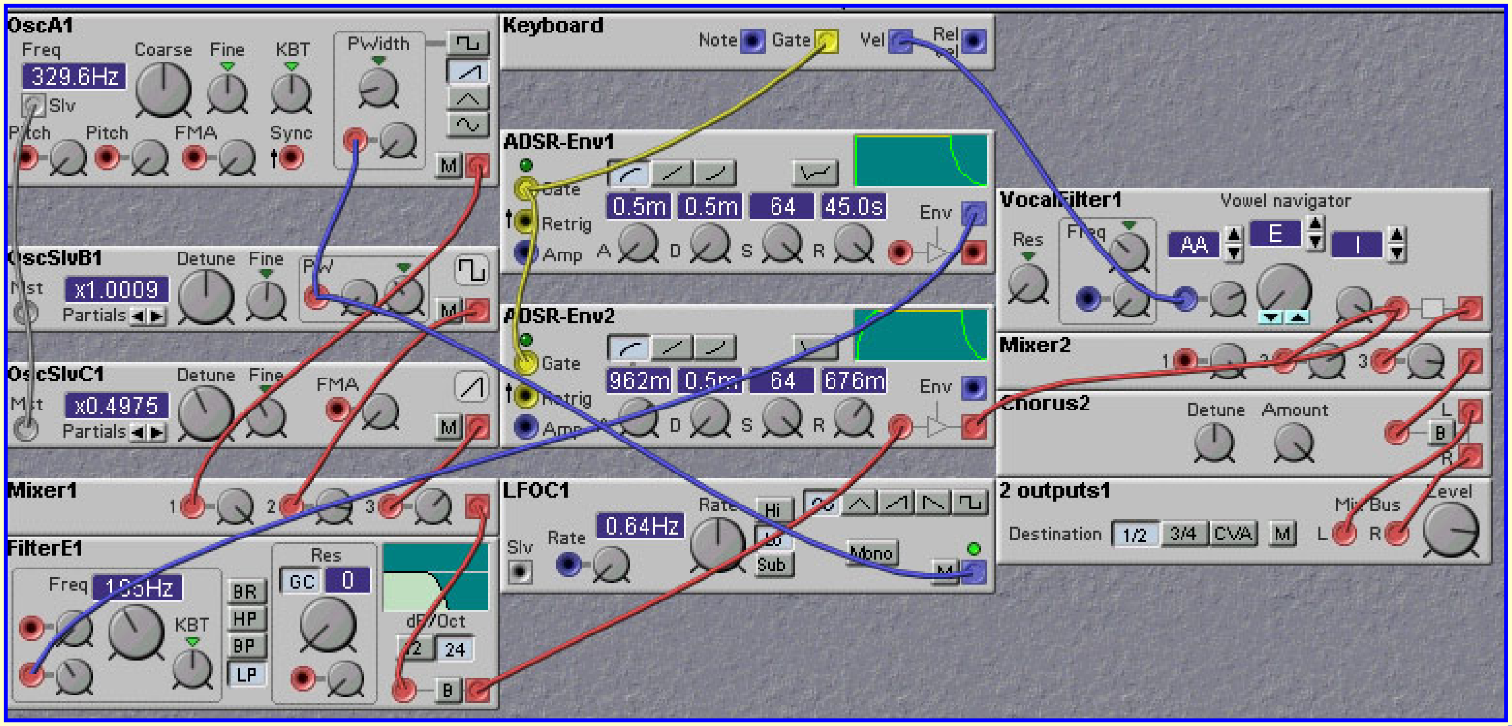


Figure 3.16. A string patch using a vocal filter to provide an expression of emotion to the sound (J. Clark).

Much of the 'analog' sound of classic synthesizers was created by small differences in the filters' pole frequencies due to component tolerances. A big feature of the Nord Modular is that it is no problem at all to combine a number of filters in all sorts of combinations. For example, the following patch illustrates the use of a custom filter structure. Four 12dB lowpass filters are strung together in a serial arrangement. A feedback path is formed from the output of the final filter in the chain back to the first. A static lowpass filter is inserted at the end of the feedback path to limit the resonance at high frequencies that would otherwise cause the sound to become very buzzy. The outputs of three of the serial filters are fed into static lowpass filters, which provide some additional filtering and also introduce some phase shift. These three signals are then summed together to provide the final filtered result. The input waveform comes from a synced sawtooth. The resulting sound is reminiscent of a bassoon.

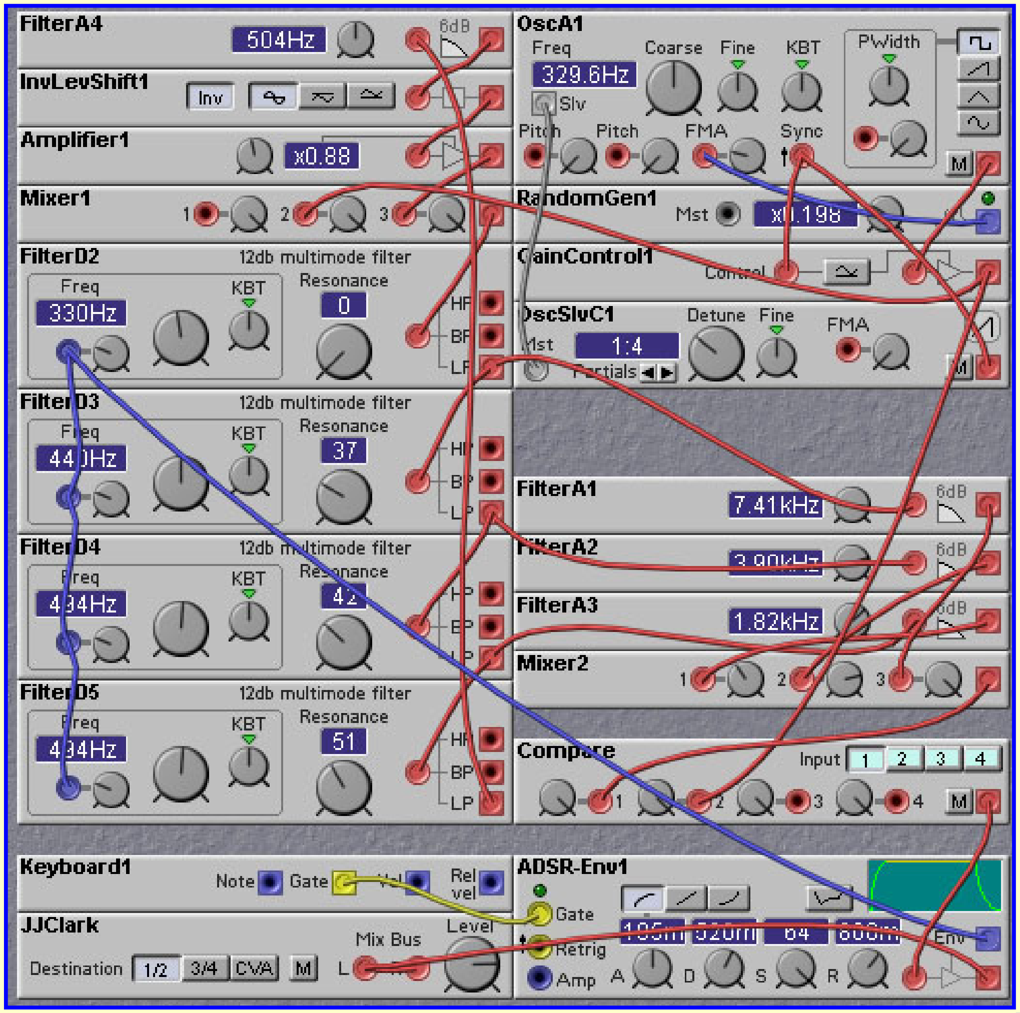


Figure 3.17. A bassoon patch using a custom filter arrangement (J. Clark).

3.5 Wet-Sounding Filters

In my studies of sonic alchemy I have come to the conclusion that the wetness of wet filters comes from 'dispersion' which is a technical term for different frequencies having different delays. (the term comes from the effect of a prism on white light in dispersing the lights spectrum). Adding in frequency dependent delays definitely makes sounds 'wet'. Typically delays up to 100 msec or so sound good and wet, but the Nord Modular can only do delays up to about 10msec. Nonetheless, this is enough to moisten the sound somewhat. Check out the patch shown below, which dampens the sound of a percussion module. Twisting knob 2 takes the sound from 'dry' to 'wet'.

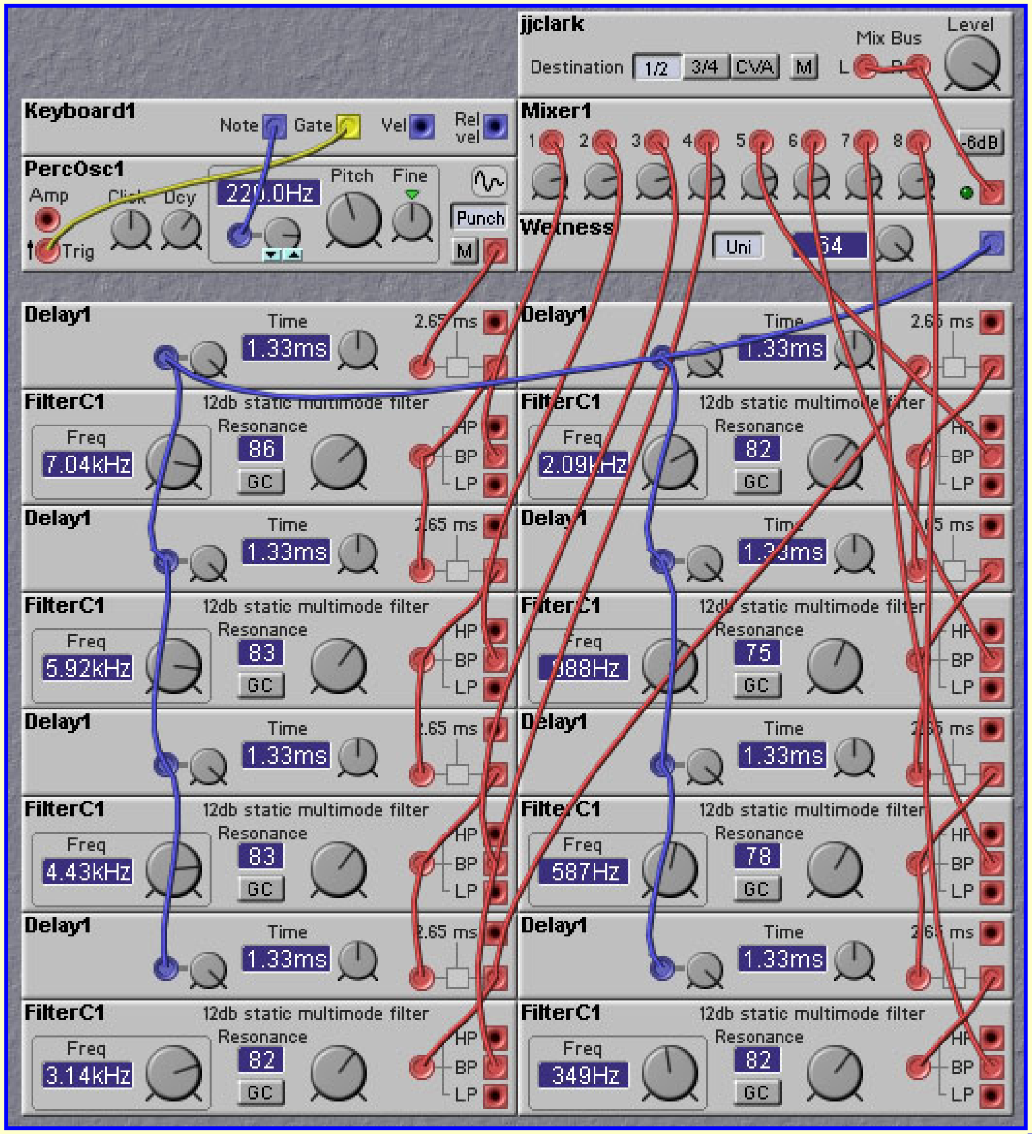


Figure 3.18. Obtaining a "wet" filter using frequency dispersion (J. Clark).

Chapter 4. Noise Generation

My mother used to say that the music I listened to was just a bunch of noise. That was even before my Einsteurzende Neubauten phase! But on reflection, there really was a lot of noise pouring out of my speakers. Of course, most of it was noise harnessed to serve the needs of the composition. In this chapter we look at different types of noise that are used in synthetic music, and how they can be implemented on the Nord Modular.

4.1 White Noise

White Noise

In its oscillator section, the Nord Modular contains a *white noise* module. The waveform produced by this module has a flat spectrum, out to about 20KHz. That is, the waveform contains frequency components of equal amplitude over this range. The spectrum is shown in the following figure.

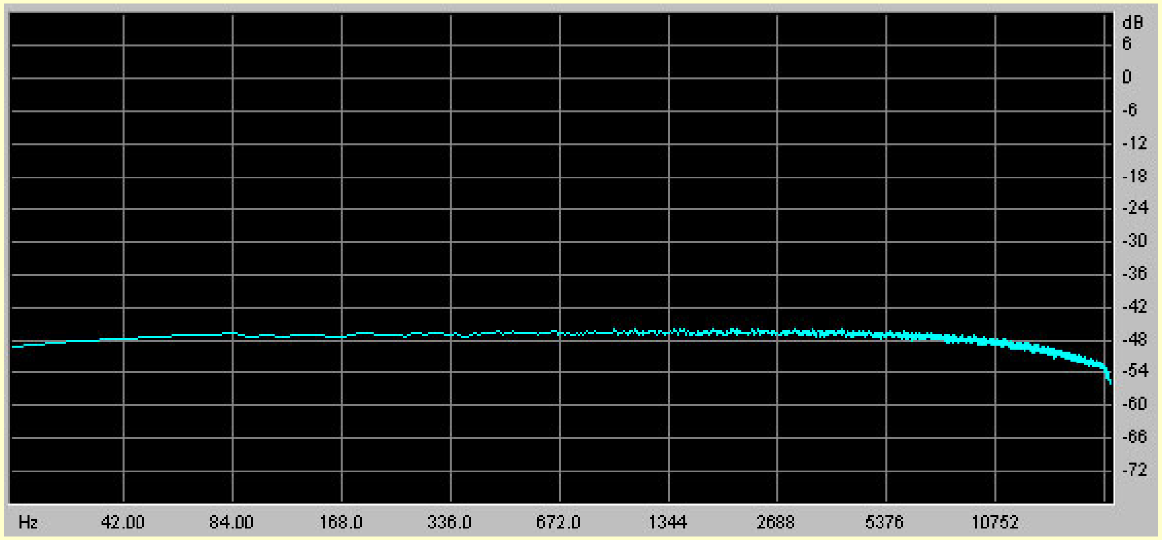


Figure 4.1. White noise spectrum.

(Note: this spectral plot, and other plots which follow, was obtained by digitizing the output of the Nord Modular feeding through a Mackie 1402 mixer, with a 20-bit Event Gina audio card. The spectrum was computed by CoolEdit Pro, in 32-bit mode.)

4.2 Brown Noise

The sound produced by the white noise generator is quite harsh, too much for many applications. Thus one would like to generate somewhat smoother noise signals. This can be easily done by filtering the output of the white noise generator with a lowpass filter. In fact, the Noise module in the Nord Modular has a `color' control, which adjusts the cutoff frequency of a first order lowpass filter applied to the output of the noise generator. At the minimum setting of the color control the cutoff frequency is set to 20KHz (or above), essentially giving no effect. As the color control is increased the cutoff frequency decreases, filtering out more and more of the high frequencies. At the maximum setting of the color control, the cutoff frequency is about 20Hz, and the spectrum of the noise output has a falloff of 6dB/octave. Noise with this rate of falloff in its spectrum is commonly known as `brown noise'. This name comes from `Brownian motion' which produces random signals with this spectrum.

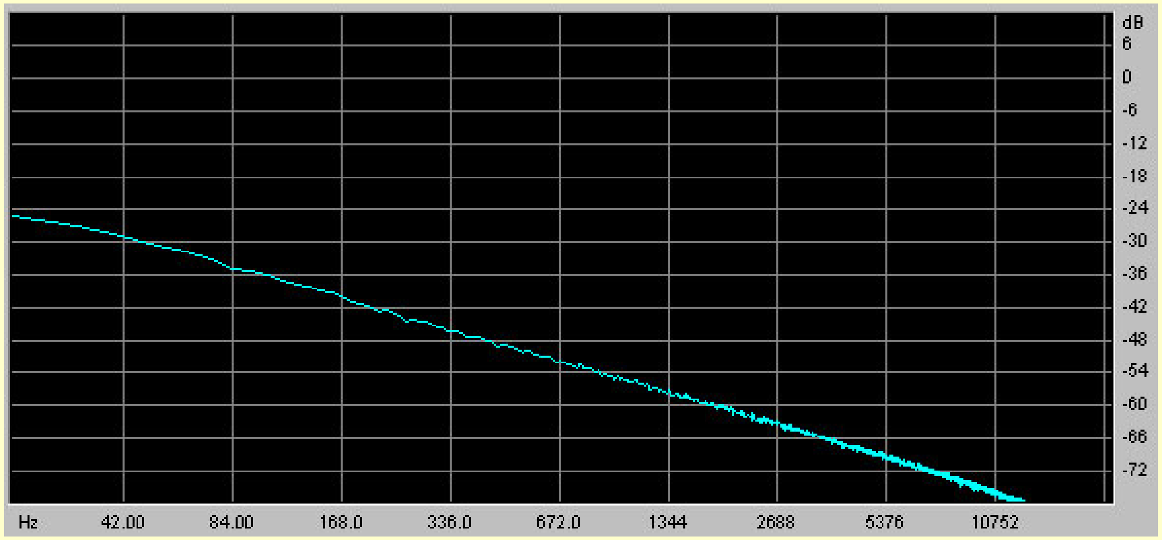


Figure 4.2. Brown noise spectrum.

4.3 Pink Noise

White noise is characterized by having equal power over frequency ranges of equal width. That is, the white noise signal has the same power in the range of 10-20 Hz as it does over the range of 100-110 Hz. Our ears, however, tend to perceive frequency ranges in a logarithmic manner. To us, the interval from 10-20 Hz is equivalent in range to the interval from 100-200 Hz. Because of this a white noise signal is perceived to have much more energy in high frequencies than at low. A noise spectrum which would be perceived to be more evenly spread would be one which had equal powers over equal frequency ranges *in a logarithmic scale*. Such a noise is referred to as *pink* noise. This type of noise is also called *1/f* noise, which refers to the fact that the spectral falloff of such noise is 3dB/octave, or a factor of 2 in power for every factor of 2 increase in frequency. Thus, the power in a frequency band is inversely proportional to its frequency.

Pink noise is `self-similar' in that if you increase its time scale by a factor of 2, it looks the same (in a statistical sense). This self-similarity property has inspired many applications to random compositions, where the micro-structure of the piece (e.g. individual notes) are mirrored by its macro-structure (e.g. thematic variations). For this reason, as well as the fact that it sounds more uniform than white noise, pink noise is very useful in music.

At first glance, it may appear simple to generate pink noise. All we have to do is filter white noise with a lowpass filter having a falloff of 3dB/octave. Well, yes, but you might notice that the Nord Modular does not include a filter with a 3dB/octave falloff. In fact, the first order filter has a 6dB/octave falloff. We need a 1/2 order filter! Or so it seems. Fortunately, there are a couple of tricks that we can play to get a filter with a 3dB/octave falloff, or to generate a noise directly that has the desired spectrum.

The first approach is due to R. Voss, and was described in a Scientific American article by Martin Gardner (M. Gardner, "White and Brown Music, Fractal Curves and One-Over-f Fluctuations", Sci.Amer., 16 (1978) p.288). The idea behind this approach is to sum the output of a sequence of *clocked* white noise generators, each being clocked a factor of two slower than the preceding one. The clocking can be thought of as a sample/hold operation on the output of a continuous white noise generator. The holding process has the effect of lowpass filtering the noise signal. The lower the clocking rate is, the lower the effective cutoff frequency of these lowpass filters. Summing the outputs of the sample/hold units then yields an overall frequency falloff with the desired 3dB/octave value.

The Nord Modular has exactly the module that we need for this process, the **ClkRndGen** module. This module acts like a sample/hold being applied to the output of a white noise generator. The pink noise generation patch using these modules is shown below:

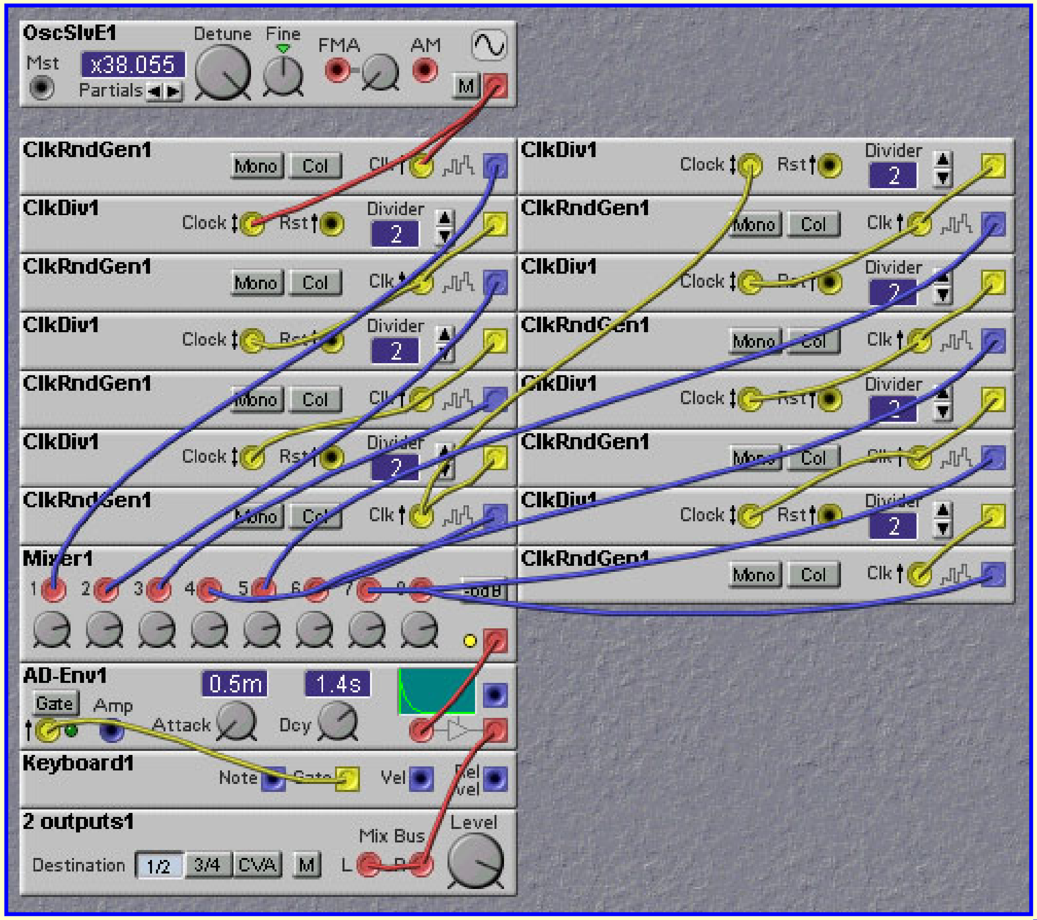


Figure 4.3. Clocked pink noise patch (J. Clark).

The spectrum of the noise generated by this patch is shown in the following graph:

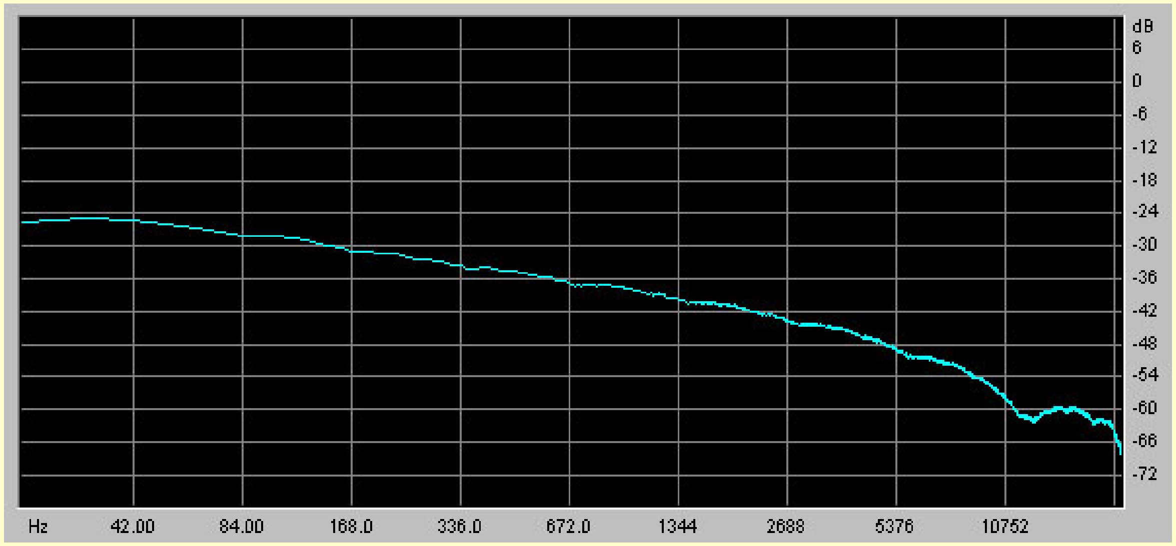


Figure 4.4. Pink noise spectrum with summed ClkRndGen module outputs

The spectrum is not perfectly straight, as you can see flat spots. The average falloff is approximately 3dB/octave, over the range of 30-3000 Hz or so.

A second approach to obtaining pink noise is to sum the weighted outputs of a set of first order (6dB/octave) lowpass filters being fed white noise. If the cutoff frequencies and weights are chosen properly, then the overall spectrum will have the desired 3dB/octave falloff. A Nord Modular patch based on this approach is shown below.

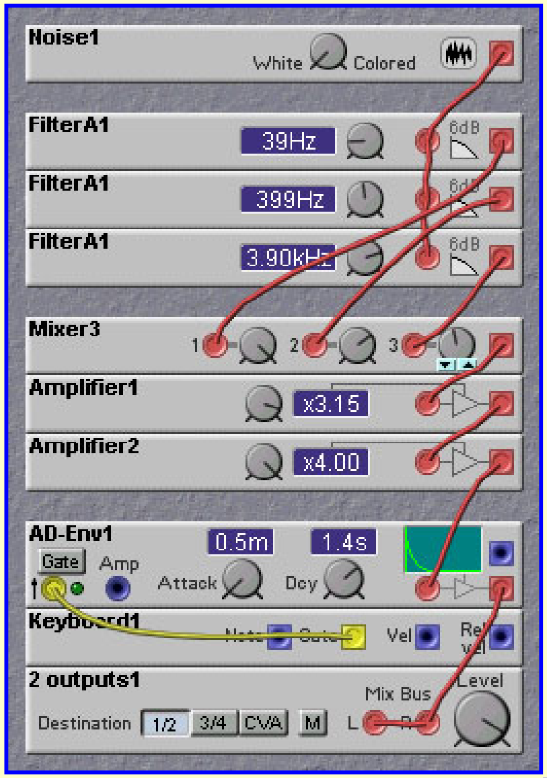


Figure 4.5. Multi-filter pink noise patch (J. Clark)

Only three filters are used in this patch. More accurate results (with less ripple in the falloff) can be obtained with more filters. The cutoff frequencies of the filters are set to be multiples of 10 apart (30 Hz, 300 Hz, 3000 Hz), and the weights of the summing operation were chosen by trial and error to give the desired response. The spectrum of the noise generated by this patch is shown in the following graph:

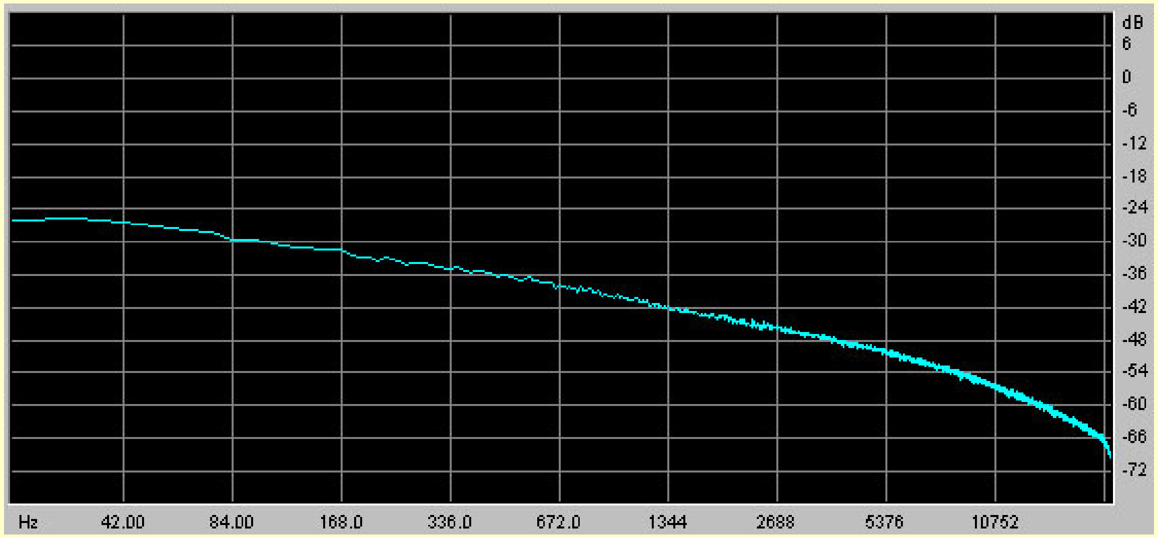


Figure 4.6. Pink noise spectrum with summed LPF outputs

Note that the falloff is smoother than that obtained with the summed ClkRndGen approach. The range of frequencies where the falloff is about 3dB/octave is about the same, however. The falloff rate begins to increase past about 6 KHz.

4.4 Pitched Noise

Many musical uses of noise requires *pitched noise*, that is, noise that has a definitely perceptible pitch. For example many percussive sounds, such as cymbals, have a definite pitch but are still rather noisy.

There is a wide range of different approaches to generating pitched noise. We will cover a few of these here.

One way of producing pitched noise is to modulate an oscillator with a noise signal. The modulation will spread the spectral peaks of the oscillator. The following patch uses frequency modulation of a sinewave oscillator by a white noise signal.

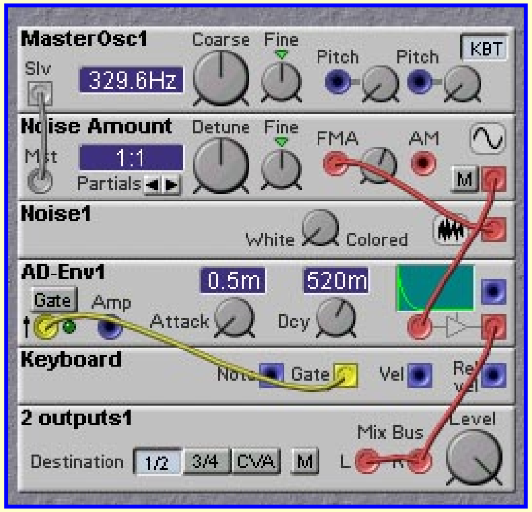


Figure 4.7. Patch for creating pitched noise with frequency modulation (J. Clark).

The spectrum of the noise generated by this patch is shown in the following graph:

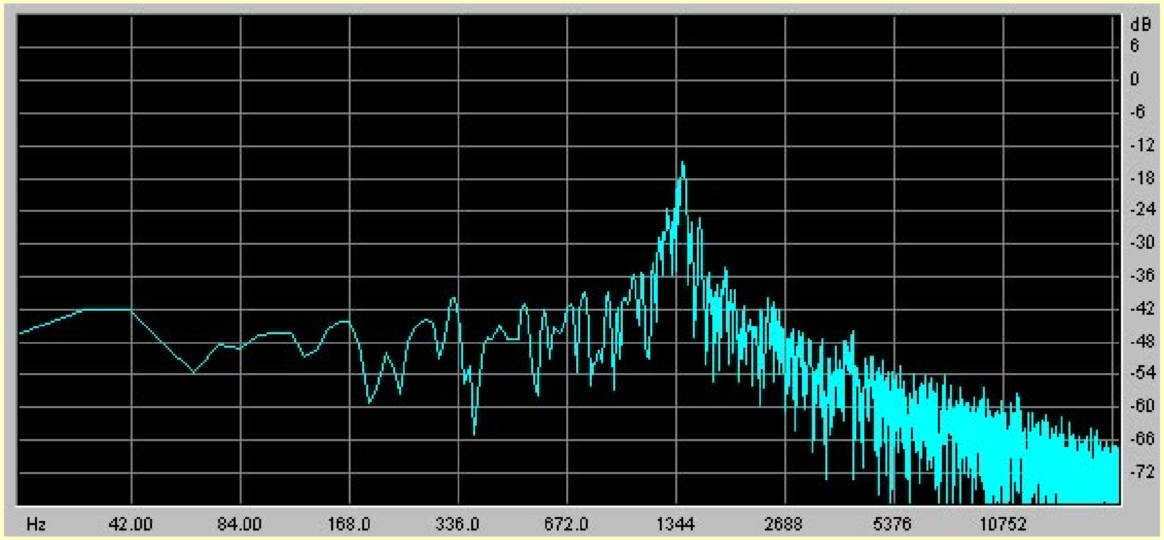


Figure 4.8. Spectrum of the pitched noise generated using FM.

Note the presence of one main peak that is quite wide. The noise modulation has broadened the very sharp peak of the sinewave oscillator. As one increases the modulation amount, the width of the peak increases and the sound becomes less pitched.

Another approach to generating pitched noise is to pass a noise signal through a resonant bandpass filter. The filter will preferentially pass frequencies near the filter's resonant peak, causing a noise spectrum similar to the one obtained with the FM technique shown above. The width of the peak can be adjusted with the filter's resonance control. The higher the resonance, the narrower the peak, and hence the more pitched the sound becomes.

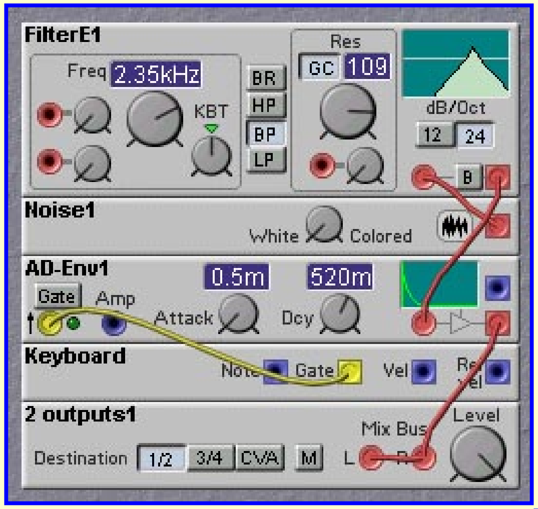


Figure 4.9. Patch for creating pitched noise with a resonant filter (J. Clark).

Chaotic systems can also be used to create pitched noise. We won't go into the details of chaos here, but you can read the chapter on chaos to learn the ideas underlying chaotic systems. The following patch uses a chaotic system of the *Henon* type. The oscillator is used to clock the dynamic system to go from one state to the next. This clocking gives the sound its basic pitch. The chaotic behaviour of the system gives the sound its noisy aspect. Adjusting the level of the Parameter1 setting causes the system to move between chaotic and non-chaotic (period) regimes. Different types of sounds are obtained in this way, so experiment!

Note carefully that in this patch there are no noise source modules at all, just an oscillator and some multipliers and adder modules. Even so, noise appears! Such is the power of chaotic systems - even simple systems can exhibit very complex behaviour.

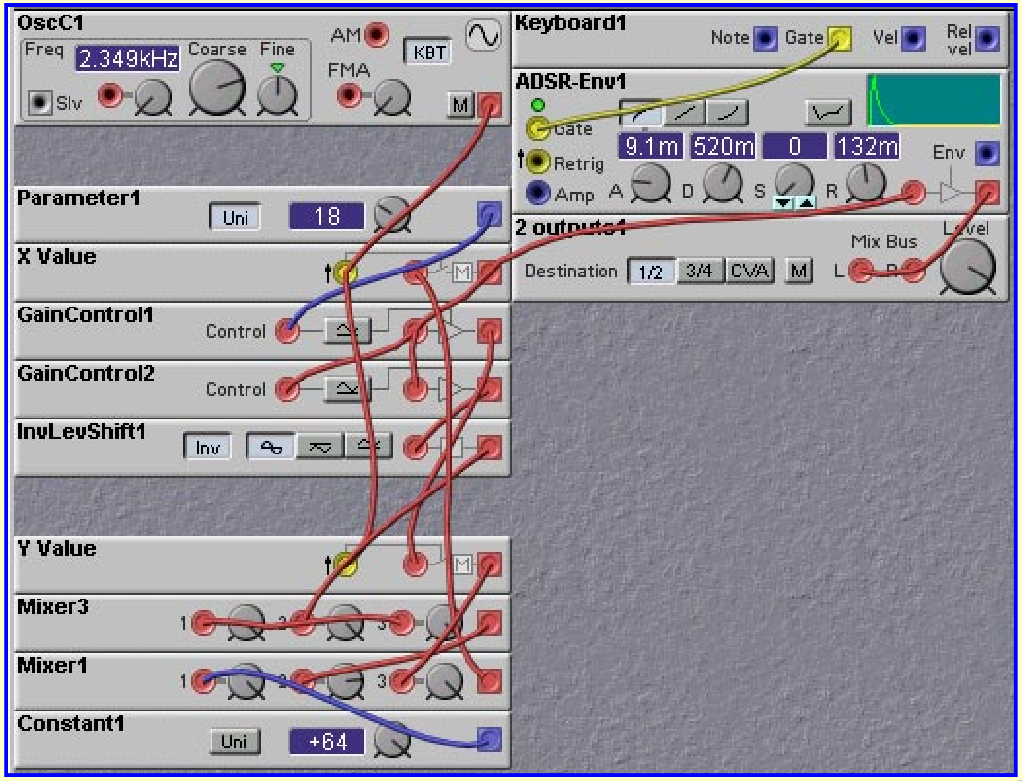


Figure 4.10. Patch for creating pitched noise with a chaotic system (J. Clark).

The spectrum of the noise generated by this patch is shown in the following graph:

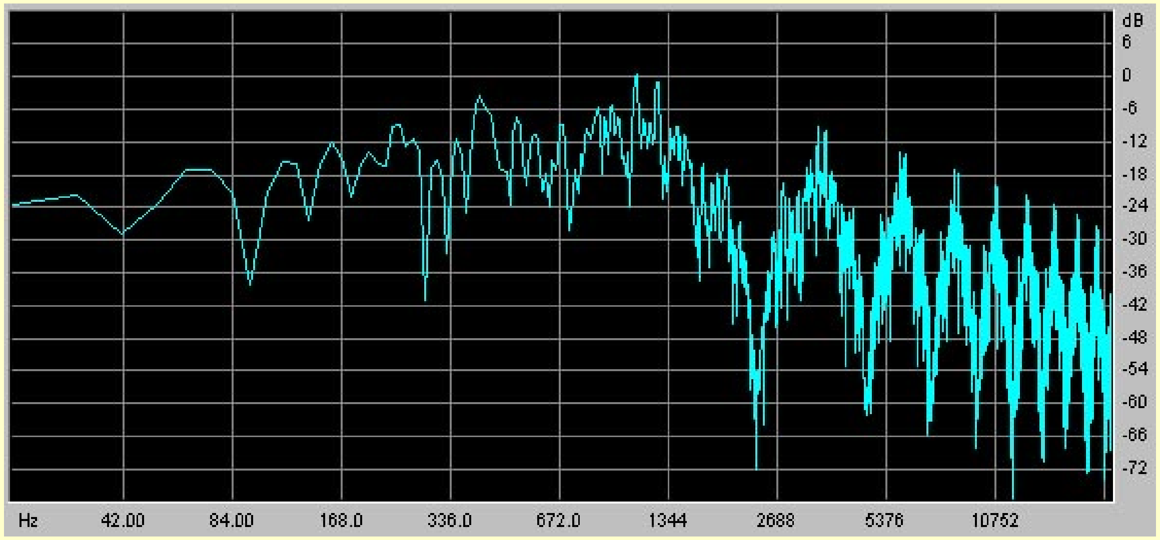


Figure 4.11. Spectrum of the pitched noise created by a chaotic system.

The spectrum is more or less flat until the frequency approaches that of the clocking oscillator, at which point the spectrum drops rapidly to a minimum. After this point, the spectrum exhibits a shape similar to a noisy square wave oscillator.

Another useful technique for altering the spectrum of noise is to pass white noise through a nonlinearity such as the shaper module. If the shaper is set to exponential (curving up) it enhances the peaks and suppresses the valleys in the noise waveform. This makes the noise signal more `spiky'. Try using such an expanded noise signal to excite a filter module. Using a single filter with high resonance only excites a single particular frequency. The phaser module and the vowel filter can resonate on more frequencies at the same time. The phaser seems to create the most interesting result, a bit like stroking a hard brush at the side of a metal sheet. An example of this is shown in the following patch which implements a 'shaker' or maracas.

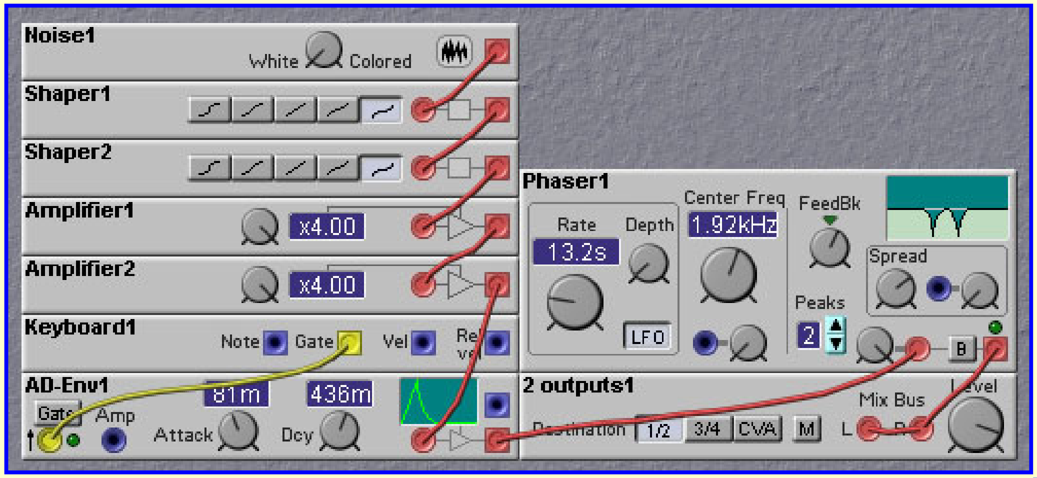


Figure 4.12. Putting white noise through an expansive nonlinearity to make it more spikey. A phaser then selects certain frequencies (J. Clark).

[Prev (Filter Techniques)][Next (Percussion)][Table of Contents]

Chapter 5. Percussion Synthesis

Every kid wants to be a drummer before they want to be a guitar player (or keyboard player). And every synth seems to want to do percussion, even if it wasn't designed to do so. The Nord Modular is no different, and this chapter explores ways in which the Nord Modular can generate percussive sounds (beyond the percussion modules that are already provided).

5.1 Bass Drum Synthesis

Bass drums may appear to be rather simple instruments, just producing a simple "thump". The simplest bass drum patch consists of a resonant filter set just below the self-oscillation resonance point. A fast, narrow pulse fed into the filter input will cause the filter to ``ring", providing a nice exponentially decaying sinusoid. Adjusting the resonance setting will adjust the decay rate, and adjusting the filter cutoff will set the pitch of the drum.

Figure 5.1. A simple bass drum patch (J. Clark).

But bass drum sounds are actually somewhat more complex than those that this patch produces. The spectrum of a bass drum sound is harmonic at low frequencies (i.e. it consists of components with frequencies that are integer multiples), perhaps slightly shifted in frequency by 5-10 Hz or so, depending on how the drum is tuned. At higher frequencies the spectrum of the bass drum sound becomes more and more inharmonic. The sound produced by the bass drum consists of two parts - the vibrating drum head, and the sound of the mallet striking the drum head. This latter sound is very short and impulsive. The pitch of the drum's vibration increases when the drum membrane is struck, due to the increase in tension of the membrane. This causes an initial rapid rise in pitch, followed by a slower decay in pitch as the membrane relaxes.

The following patch shows a rather complicated implementation of a bass drum. The low frequency components are produced by a master sine oscillator and a sinebank module containing six slave oscillators. The frequencies of these oscillators are adjusted to more or less correspond to the spectral components of a typical bass drum. The higher frequency, inharmonic, components are created by a pair of oscillators each frequency modulating the other. A short click is created by passing the keyboard gate signal through a pulse module. This produces the sound of the mallet hitting the drum. The overall amplitude of the sound is controlled with an AD envelope, with a very short attack and a relatively short decay. The envelope is also used to alter the pitch of the drum to mimic the change in pitch due to the stretching of the drum membrane when struck. Oscillator sync is used to ensure that the oscillator waveforms always start at the same phase when the attack begins.

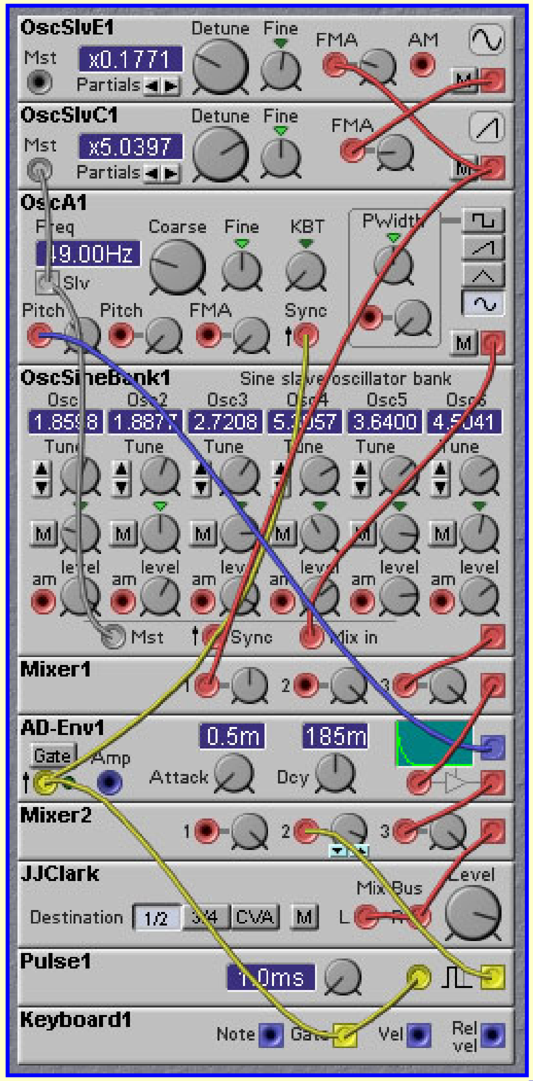


Figure 5.2. A rather complicated bass drum patch (J. Clark).

The following patch uses distortion to create harmonics from a low-frequency sinewave. A filter is used to tailor the spectrum. Distortion is applied before and after the filter for greater flexibility in the setting of harmonic levels. Separate envelopes are used for amplitude and pitch.

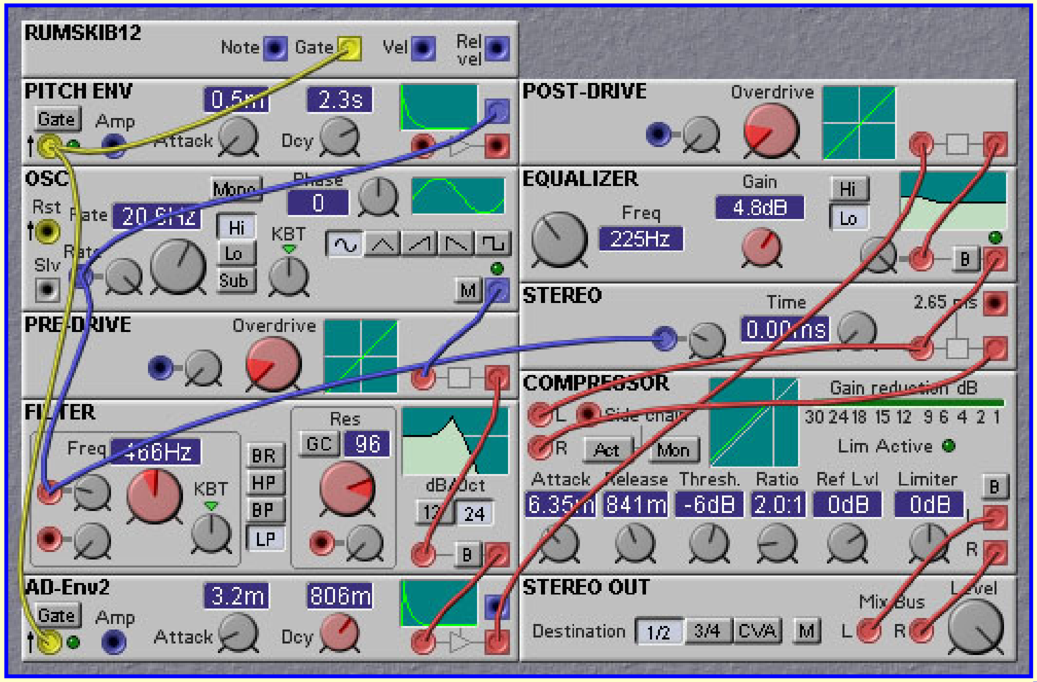


Figure 5.3. Another bass drum patch, which uses filter, distortion and compression modules (P. G. Christensen).

5.2 Snare Drum Synthesis

A snare drum has two heads or vibrating membranes. The lower one (the one which you don't pound on) has a number of cables stretched across it (making up the "snare" which gives the drum its name). If we ignore the effects of the snare cables for the moment, the acoustic coupling of the two vibrating membranes by the air between them, and by the shell of the drum, causes a complex set of resonant modes to arise. Apart from the lowest two resonant modes, the resonant frequencies are seen to form two groups, each of which are nearly evenly spaced in frequency. Each of these groups is similar to that of the bass drum sound.

If we just synthesized these resonant modes, our "snare drum" would sound sort of like a bass drum. It wouldn't sound much like a snare at all. To get the snare drum sound we have to also simulate the effects of the snare cables. Basically the snare cables act as resonant structures, which interact with the vibrating heads of the drum, through contact with the lower head. The cables are effectively one-dimensional structures and behave quite differently than the two dimensional drum head membranes. Thus the resonant modes of the snare cables are very different than those of the drum head membranes. The transfer of energy between the lower drum head and the snare cables will therefore be very complicated, almost random. To make matters even more complicated, the lower drumhead and the snare cables make physical contact in a nonlinear fashion (i.e. sometimes they are in contact, sometimes they are not). This nonlinearity causes all manner of inharmonic frequencies to be generated.

The upshot of the physical analysis of the snare drum is that it is very complicated, impossible to model exactly and impractical to implement even an approximate physical model on the Nord Modular. Instead, we take an empirical approach and try to capture essential features of the sound. There are two main parts to the sound - the basic (in)harmonic tone of the two vibrating drumheads, and the noisy sound caused by the snare cables. We could implement the drumhead sound in much the same way that we did for the bass drum. However, higher frequency components are mainly masked by the snare noise, so we can get away with just synthesizing the lowest frequency components. The following patch emulates the snare drum circuit in the Roland TR-909 drum machine. It uses two triangle oscillators to provide the basic drumhead vibratory modes. LFOs are used for this to save on processing power. The inaccuracies caused by the lower sampling rate of the LFO oscillators add a bit to the dirtyness of the sound. As in the TR-909, each oscillator has its own envelope. The lower frequency oscillator is given a slightly longer decay. The effect of the snare cables is emulated with a white noise source passed through a lowpass-highpass filter chain. The outputs of the low and high pass filters are passed through their own envelopes and then mixed together with the drumhead mode oscillator signals. The overall effect is rather plain, but definitely recognizable as a snare drum sound. The patch could be livened up by adding LFO control over the filter cutoff frequencies and applying distortion to the output. The mixing ratio between the snare noise component and the drumhead mode vibration component could be altered by key velocity, to emulate the natural effect of increased snare noise with striking force (since there is more interaction between the snare cables and the lower drumhead when the upper drumhead is struck more forecefully). I will leave it to you to try these enhancements!

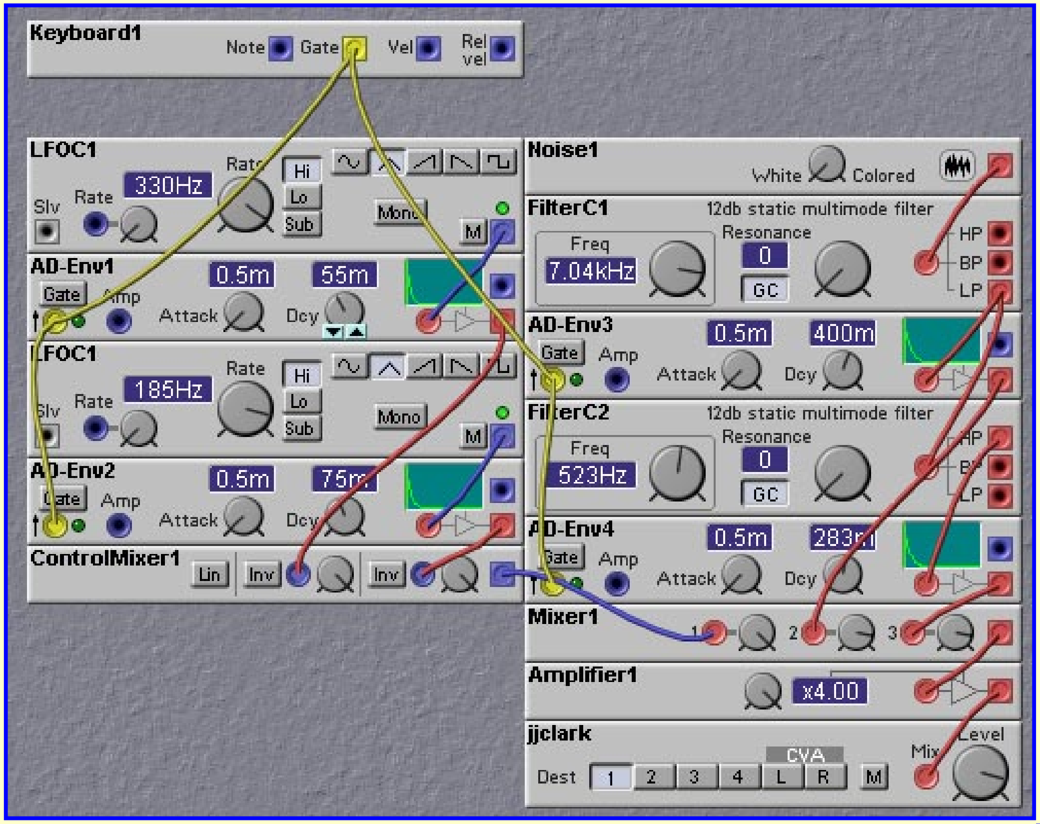


Figure 5.4. Snare drum patch based on the Roland TR-909 snare circuit (J. Clark).

5.3 Synthesis of Gongs, Bells, and Cymbals

One of the most difficult of all instruments to synthesize accurately are bells, gongs and, especially, cymbals. Part of the difficulty lies in the fact that the sound of these instruments are `inharmonic'. That is, their frequency spectrum does not consist entirely of harmonically related sine waves, but instead includes components whose frequencies are not integer multiples of the fundamental frequency. These inharmonic frequency components result in the `metallic' sound characteristic of gongs, bells, and cymbals.

Additive Synthesis

All periodic waveform generators produce harmonically related frequency components. One cannot, therefore, create inharmonic sounds directly using a single periodic waveform generator. There are a number of techniques for creating inharmonic waveforms. The most commonly used technique is *additive synthesis* which linearly adds two (or more) periodic waveforms that are detuned relative to each other. If the fundamental frequencies of the two waveforms are incommensurate (that is, they do not have any harmonics in common) then the resulting waveform will be aperiodic (it will never repeat exactly). An example would be with a wave of fundamental frequency 1, and another with fundamental frequency 1*square_root(2). In general, it is difficult to tune two oscillators to incommensurate frequencies. It is more likely that the two oscillators will have some rather high frequency harmonic in common. To make a convincing metallic sound, one needs to combine a large number of detuned waveforms, each tuned to frequencies incommensurate (or nearly so) with the others. For this reason, making inharmonic instruments via additive synthesis of this sort is tricky, but not impossible.

An example of using additive synthesis to get inharmonic sounds on the Nord Modular is shown in the figure below:

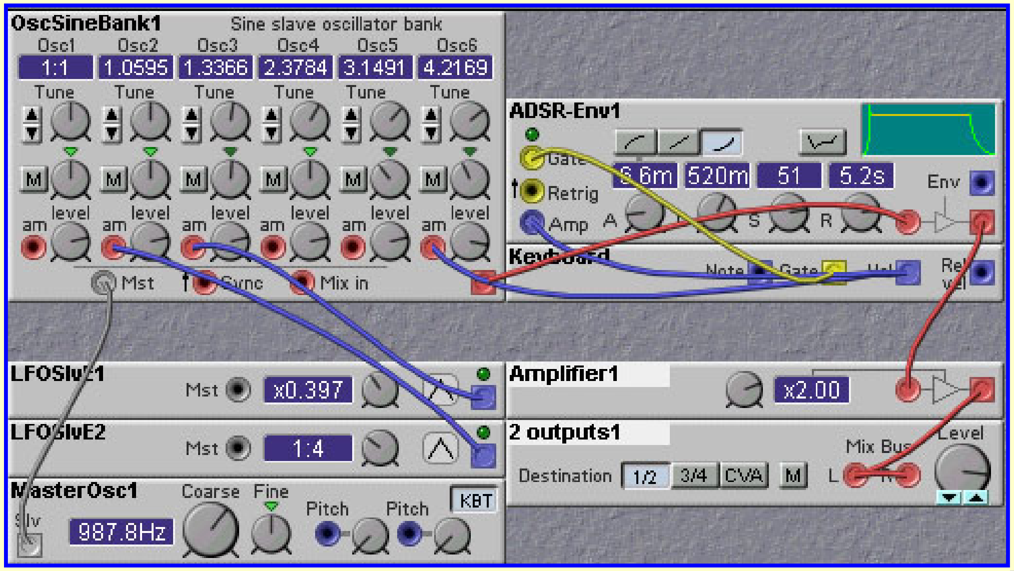


Figure 5.5. An additive synthesis approach to obtaining inharmonic sounds (J. Clark).

In this patch we use the sinebank slave oscillator module, which contains 6 separate sinewave oscillators. We set the relative frequency of the six oscillators to values which result in inharmonic frequencies. The amplitude of the second and third oscillators are modulated by two LFOs. This modulation provides some variation to the sound, making it more realistic. The envelope generator provides the bell-like amplitude dynamics.

Ring-Modulation

An easier way to get a lot of inharmonic frequencies is to use a nonlinear operation, instead of merely adding waveforms together. `Ring-modulation' is probably the most frequently used classic approach to generating inharmonic sounds. A ring-modulator essentially multiplies the two voltage waveforms together. This results in frequency components being generated which have frequencies equal to the pairwise sum and differences of each of the frequencies components contained in one waveform with each of those present in the second waveform. The original frequency components found in the two waveforms are suppressed and not found in the output. A modified version of the ring-modulator called the Amplitude modulator retains the original frequency components in addition to the new components. For example, if waveform 1 has components with frequencies of 1KHz and 1.2 KHz, and waveform 2 has components with frequencies of 2.7KHz, and 3.3 KHz, then the ring-modulated waveform will have components with frequencies of 1.7KHz (2.7-1), 3.7KHz (2.7+1), 2.3KHz (3.3-1), 4.3 KHz (3.3+1), 1.5KHz (2.7-1.2), 3.9KHz (2.7+1.2), 2.1KHz (3.3-1.2) and 4.5KHz (3.3+2.2).

If the fundamental frequencies of the two waveforms input to a ring-modulator are incommensurate, or nearly so, the set of sum and difference frequencies that are generated will be inharmonic. The advantage of using a ring-modulator over merely adding two waveforms is that the number of distinct frequency components that are generated is much greater. For example, if the two waveforms have 3 harmonics each, adding the two waveforms will yield 6 components, while ring-modulating will yield 18 components. For more complex waveforms, with richer spectra, the difference will be even more pronounced. This efficiency in generating inharmonic spectral components is the main reason ring-modulation has found so much use in generating metallic sounds in synthesizers.

In the Nord Modular, ring-modulation is easy to do. There is now a ring-modulator module, which can also be adjusted to provide amplitude modulation (which leaves the original spectral components). One can also use the controllable amplifier to do ring-modulator (and this is how it was done before the OS upgrade that gave us the ring-modulator module).

An example of using ring-modulation to create inharmonic sounds on the Nord Modular is shown in the following figure:

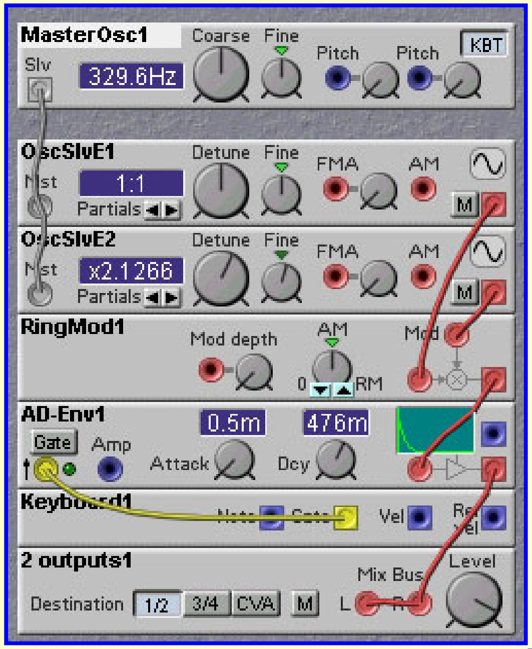


Figure 5.6. Using ring-modulation to obtain inharmonic sounds (J. Clark).

Even though a single ring-modulator can give us a very rich inharmonic spectrum, we need not stop there! Connecting two ringmodulators in a row with only slightly detuned sinewaves gives a nice detuning effect like the big double gamelan gongs in addition to the ring-modulation effect. To make that a stereo effect we can use four. The detuning can be done by feeding only a slight constant value to the FMA inputs, one sine oscillator a negative value and the other a positive value. The advantage of using the FMA inputs is that the detune amount stays constant over the frequency range. To make a more realistic sound, one can follow the ring-modulators with a resonant filter, or delay line based `tube'. The frequency of the oscillators in combination with the frequency of the 'resonating body' controls the timbre of the metallic sound. Keeping the frequencies not too high is best for replicating the bronze sound of gongs. Using sawwaves or squarewaves instead of sinewaves can also provide richer sounds..

FM Synthesis

The other most commonly used technique for obtaining inharmonic waveforms is frequency modulation (FM). Frequency modulation is a technique for non-linearly combining two waveforms. One waveform is used to vary the frequency (or the phase) of the other. The resulting spectrum is very complex and generally contains an infinite number of harmonics.

Implementing FM on the Nord Modular is very straightforward, thanks to the frequency modulation inputs on the slave oscillator modules. A simple FM bell patch is shown below:

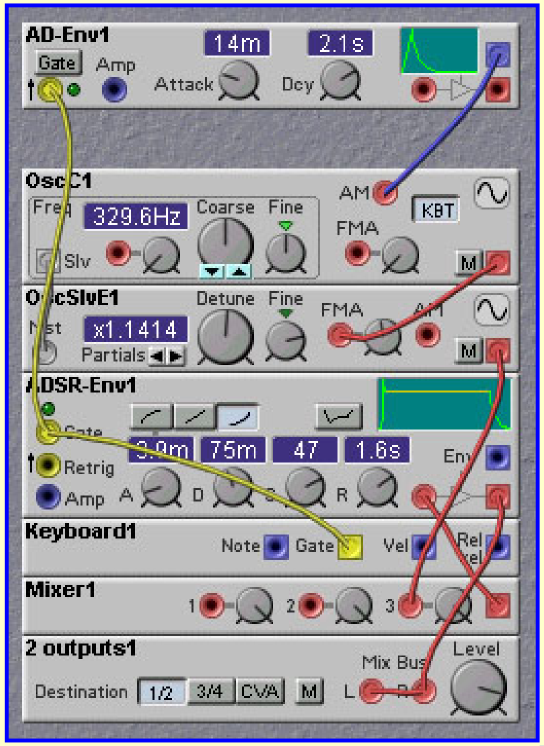


Figure 5.7. An FM synthesis approach to obtaining inharmonic sounds (J. Clark).

In this patch a master oscillator is used to provide both a fixed frequency reference for the slave oscillator as well as a frequency modulation signal for the slave oscillator. The nominal ratio of the slave oscillator frequency to that of the master is set to a non-integer value, which will result in inharmonic frequency components in the output of the slave oscillator. The first envelope generator modulates the amplitude of the master oscillator output. As the master output is used to frequency modulate the slave, increasing its amplitude results in a brighter sound for the slave output, with more frequency components. Thus, this envelope generator acts very much like a filter envelope in a standard subtractive synthesis patch. The second envelope generator is used to provide the overall amplitude dynamics associated with bell-like sounds. The initial `blip' or transient models the striking of the bell.

Cymbal Synthesis

As mentioned earlier, perhaps the most difficult instrument to get right is the cymbal. Its sound is very complex. It contains inharmonically related frequency components, but unlike other metallic instruments, these components are not pure sinusoids, but rather components that are somewhat spread in frequency. These "wide-band" frequency components give the cymbal its noisy sound, and in fact noise can be used to spread the frequency components of a sinusoidal oscillator. To see how we can begin trying to synthesize a cymbal, examine the following patch:

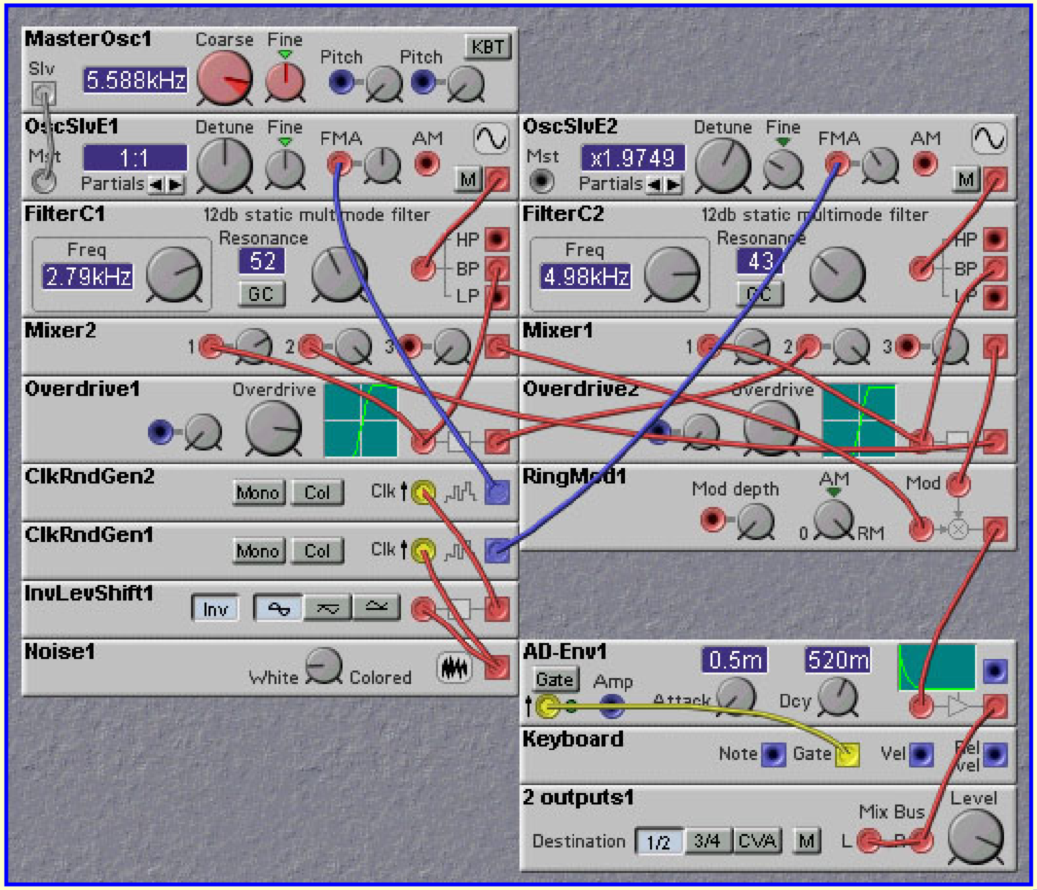


Figure 5.8. Cymbal synthesis using noise-FM (J. Clark).

In this patch, we use two sinusoidal oscillators which are frequency modulated by noise sources. This modulation spreads the bandwidth of the single frequency component of the sine wave oscillators, producing a form of "pitched noise". We want the variation of the two oscillators to be somewhat uncorrelated, so we produce two different noise signals. To make sure the noise sources are different we use two clock noise sources and clock one with a noise signal and clock the other with the inverse of the noise signal. In general, the timing of the rising and falling edges of a noise signal will be uncorrelated, so this will give us what we want. (to make this need for de-correlation obvious, try modifying the patch to connect the same noise signal to both oscillators - the sound is much less random).

The nominal frequencies of the two oscillators are set to inharmonic intervals. The master oscillator should be set to a fairly high frequency (5-7 Khz works well). The outputs of the oscillators are sent through bandpass filters and then through overdrive modules which distort the waveforms, thereby generating additional harmonics. The outputs of the overdrive module are fedback to mixers, which combine with the opposite pre-overdrive signals. The outputs of the mixers are then ring-modulated. This ring-modulation produces even more inharmonicity.

The cymbal patch given above is far from realistic, and perhaps is only useful for a hi-hat buried in the mix. There are many analog drum machines, such as the Roland TR-808 that do a decent job of synthesizing cymbal sounds. Robin Whittle (the maker of the TB-303 Devilfish mod) posted a message on the music-dsp mailing list on how the TR-808 cymbal sound is created. He writes:

"The Roland TR-808 drum machine uses six audio square-wave oscillators, all mixed together with no precise tuning. These are mixed together and used as the input of two separate bandpass filters. The output of one bandpass filter drives one "gating" circuit. The output of the other drives two "gating" circuits.

Each gating circuit is an inspired concoction of transistors, diodes, resistors and capacitors which distorts the signal - imagine limiting a wide ranging signal between two limits, so it is basically a square- wave, and then changing the limits to control the volume. The three gate circuits have various time-constants for their volume.

Their outputs go into three separate high-pass filters, whose outputs are summed to produce the final signal."

The Nord Modular patch shown below is a (perhaps not entirely accurate) implementation of this approach.

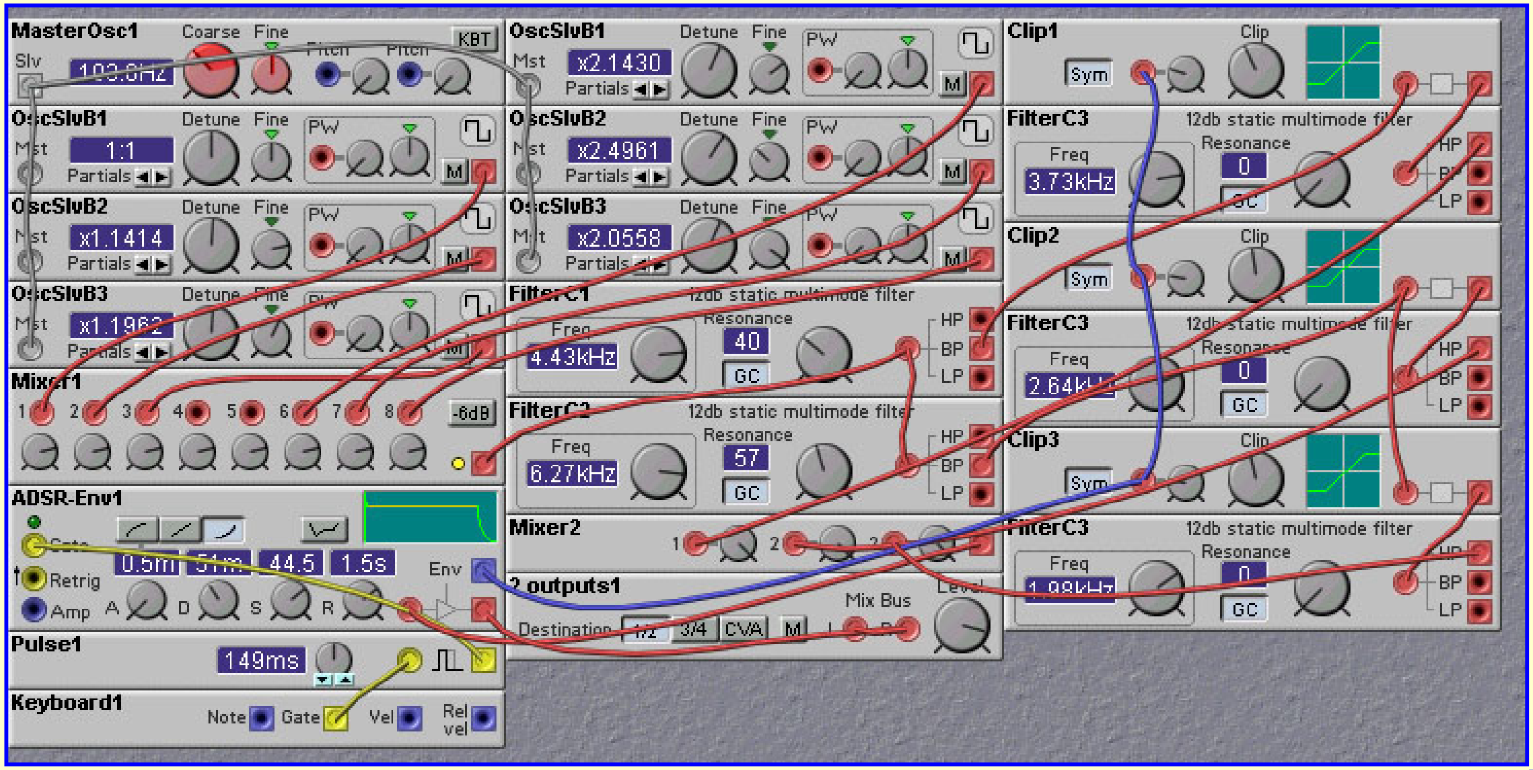


Figure 5.9. Cymbal synthesis using many detuned square-wave oscillators (J. Clark).

This patch produces a much more realistic cymbal sound, although still far from perfect. One of the limitations of this patch has to do with the limited frequency range of the Nord Modular. Even though we can't hear frequencies above 20KHz (or not even that high!) the cymbal vibration includes many frequencies above this limit. These high frequency components can then interact nonlinearly with each other which can generate sum and difference frequencies just as in ring-modulation. While the individual frequency components might be too high in pitch to hear, their difference frequencies may be audible. The Nord Modular cannot replicate these effects, and so the complex variation of a real cymbal will not be captured.

Physical modeling has also been suggested as a way to synthesize cymbal sounds. In contrast to the physical modeling approach to synthesizing woodwind and stringed instruments, which use one-dimensional digital waveguides, physical modeling of cymbals requires the use of two- (or even three-) dimensional digital waveguides. While, one could conceivably implement two-dimensional waveguides with the Nord Modular (by constructing two coupled one-dimensional waveguides), accurate synthesis of cymbals would require a very large number of elements. The distinctive sound of the cymbal comes in large part from the grooves and ridges embossed on its surfaces. These influence the propagation of and reflection of waves along the surface. The propagation velocity, which determines the frequencies of individual waves, varies nonlinearly with the stress at each point in the cymbal. To model accurately, one would need nonlinear waveguide segments for each of these grooves and ridges. Thus hundreds, if not thousands, of delay line elements would be needed, something which is not possible with the current version of the Nord Modular.

5.4 Synthesis of Hand Claps

Hand claps are very popular in many musical styles. Getting a rather artificial clap sound is not too hard. The basic idea is to generate a brief percussive burst that is quickly repeated. These repeats simulate the echoes and reverberations produced in an enclosed environment from the slapping together of the hands. An example is shown in the following patch (by *Grant Ransom*).

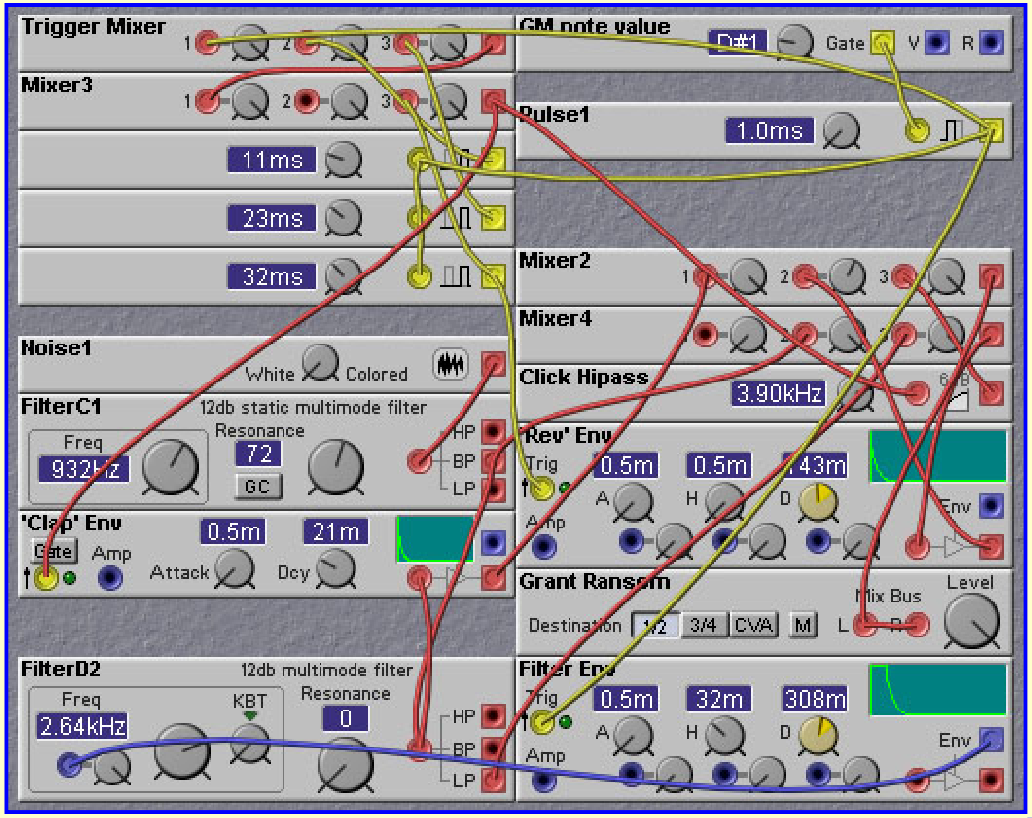


Figure 5.10. A patch for generating hand clap sounds (G. Ransom).

In this patch, white noise is amplitude modulated by a fast envelope to create the direct hand clap impulse. The envelope is triggered by a series of four pulses, each about 11 msec apart.

I will leave it as an exercise for the reader to synthesize the sound of one hand clapping!

[Prev (Noise Generation Techniques)][Next (Additive Synthesis)][Table of Contents]

Chapter 6. Additive Synthesis

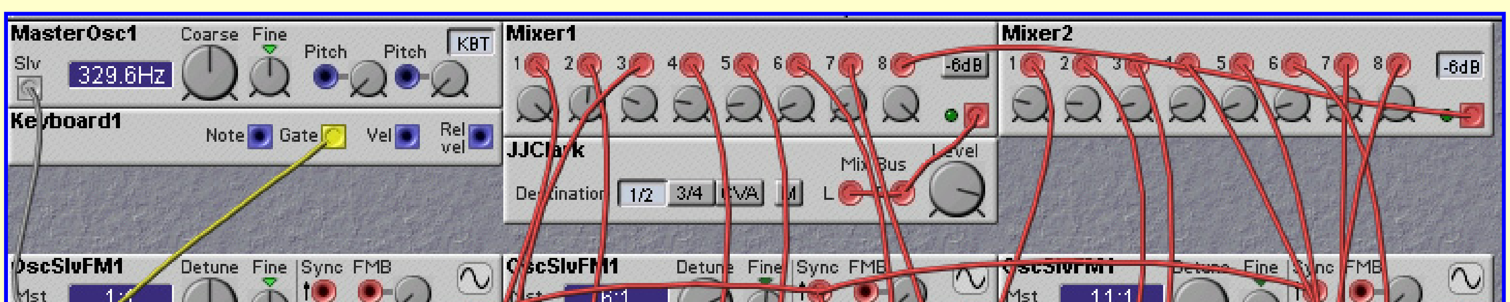
In the usual subtractive synthesis approach, a complex waveform is incrementally whittled away to produce the wanted sound. To use an artistic analogy, subtractive synthesis is like sculpting marble - the marble is chipped away to reveal the statue beneath. Another approach would be like sculpting with clay, where the clay is molded and clumped together to create the desired shape. Such a way of doing sound synthesis is known as additive synthesis.

6.1 What is Additive Synthesis?

Additive synthesis is a technique which builds sounds from the bottom up, by incrementally adding simple waveforms together to achieve the desired resultss. Additive synthesis can be used to very accurately model almost any musical instrument, given enough computational resources. Computational resources are limited on the Nord Modular, however, implying that perfect emulations of instruments will not be achievable. But, very good results can be obtained in some cases!

Besides the high demand on computational resources, additive synthesis has some implementational quirks which can affect the quality of the synthesized sound. One of these arises from the fact that a large number of sound sources are being added together. Since each of these sound sources has some noise, adding them together will neccessarily increase the noise level. Compare this to the situation in subtractive synthesis where the output noise can be less than the noise level of the input waveform, due to the filtering. Thus, instruments created using additive synthesis can be noisy. Another factor to consider is the relative fragility of the harmonic structure of the additive sound. If the user has some control over some aspect of the additive process, say in the amplitude or frequency of some overtone, the complex harmonic structure that provides a certain type of sound can be destroyed, leading to a thin sound. In fact, one of the main criticisms of additive synthesizers is that they produce rather thin sounds. This is not an problem with additive synthesis per se - additive synthesis does have the capability to make rich powerful sounds. The problem is that the characteristics of the overtones must be carefully and precisely set in order to achieve such sounds. It may be difficult to do this just by manually adjusting the sound parameters. Resynthesis based on mathematical analysis of the target sound is usually required to obtain good results. Another problem with additive synthesis is that transient sounds are very difficult to synthesize. This is because transients require a large number of rapidly varying overtones to obtain accurate reconstructions. The phase relationships between the various oscillators must also be carefully controlled to get a sharp attack. Even though the harmonic structure of a sound does not depend on the relative phases of the individual oscillators (except in rare cases of complete cancellation), temporal events such as rapid attacks or decays are very much dependent on the oscillators phases. Noisy instruments, such as drums, flutes, and cymbals, are also hard to synthesize, again due to the large number of overtones required. One can overcome this difficulty somewhat by modulating the overtone frequencies and amplitudes with noise signals. This has the effect of spreading the sinewave frequency, which is itself just an impulse in the frequency domain, to a broader range of frequencies. In general, however, additive synthesis is at its best when synthesizing sounds that are quasi-periodic.

The basic additive synthesis patch is shown below.



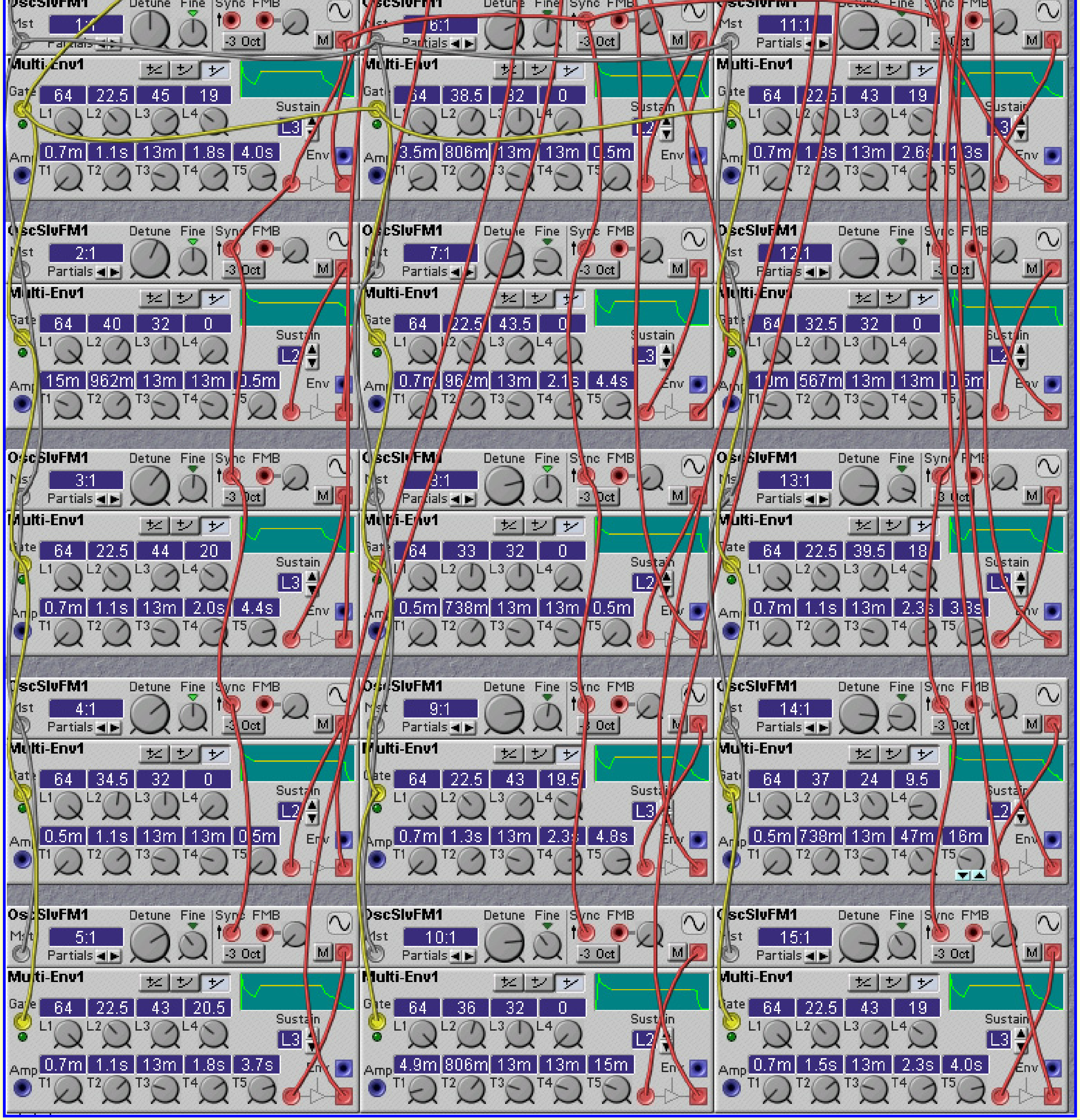


Figure 6.1. A basic patch that can be used as a template for additive designs (J. Clark).

6.2 Re-Synthesis

If you play around with the patch shown in figure 6.1, trying to adjust the parameters to achieve a certain sound, you quickly realize that it is not so easy to get what you want. You can learn quite quickly how filters affect the sound of square waves and sawtooth waves, and so using subtractive synthesis it is not too hard to make something close to the sound you are thinking of. Even getting FM synthesis using a fixed patch setup such as in the Yamaha DX-7 to do what you want can be learned after a bit of practice. But Additive synthesis never seems to get any easier no matter how long you work at it. Doing additive synthesis by hand is always a matter of tedious setting of partial frequencies and sculpting partial envelopes. Having a table of partial frequencies and partial amplitude time variations constructed from spectrographic measurements of a real sound (such as a clarinet or piano) can help remove the trial and error part of the additive synthesis design process. This sort of design process is probably better left to automated, computerized, *re-synthesis* methods. In re-synthesis, a real instrument is sampled, and the measured waveform subjected to a time-varying spectral analysis. For example, spectra could be computed using an FFT algorithm over moving time windows of 20 milliseconds width, over the extent of the instrument's sample. Peaks in each of the spectra are detected and localized, and the strong ones identified as partials. These peaks are tracked from time-window to time-window. The amplitude and frequency variation of each tracked spectral peak are then measured. From these variations the parameters for the partial frequency and amplitude envelopes are generated.

Re-synthesis is not a simple process, even for computers. It is often difficult to detect spectral peaks, as the peaks may widen and merge with others, or be weak. Tracking peaks from one time segment to the next is also a difficult problem, as peaks may be created, disappear, or merge with other peaks. Nonetheless, quite good results can be obtained. This is not something that the Nord Modular is capable of doing itself, at least not with sufficient accuracy. One could write a computer program that would take in a wave file containing an instrument sample and spit out a Nord Modular patch (similar to that of figure 1) which would give an additive synthesis approximation to the sound. If you, or someone close to you, ever writes such a program, let me know!

If you are the patient type, you could try to do re-synthesis by hand, from a table or chart of partial frequencies and amplitudes. Since partial amplitudes in such charts are usually expressed in decibels (dB), it is neccessary to know the attenuation of the mixer and/or sine-bank level controls in dB. Ico Doornekamp has tabulated a list of gain values for the Nord Modular mixer level control in dB. Apparently (although I have not confirmed this) the same conversion factor can be used for the level control in the sinebank oscillator. The values are as follows:

127 1.00000 0.000	126 0.97852 -0.094	125 0.94617 -0.240	124 0.91984 -0.363	123 0.89383 -0.487	122 0.86836 -0.613	121 0.84320 -0.741	120 0.81852 -0.870
119 0.79414 -1.001	118 0.77023 -1.134	117 0.74680 -1.268	116 0.72370 -1.404	115 0.70109 -1.542	114 0.67883 -1.682	113 0.65703 -1.824	112 0.63570 -1.967
111 0.61480 -2.113	110 0.59422 -2.261	109 0.57406 -2.410	108 0.55445 -2.561	107 0.53523 -2.715	106 0.51641 -2.870	105 0.49805 -3.027	104 0.48008 -3.187
103 0.46250 -3.349	102 0.44539 -3.513	101 0.42867 -3.679	100 0.41237 -3.847	99 0.39648 -4.018	98 0.38102 -4.191	97 0.36602 -4.365	96 0.35133 -4.543
95 0.33711 -4.722	94 0.32328 -4.904	93 0.30984 -5.089	92 0.29680 -5.275	91 0.28431 -5.462	90 0.27183 -5.657	89 0.25787 -5.886	88 0.24391 -6.128
87 0.23461 -6.297	86 0.22530 -6.472	85 0.21600 -6.655	84 0.20425 -6.898	83 0.19249 -7.156	82 0.18466 -7.336	81 0.17682 -7.525	80 0.16898 -7.722
79 0.15923 -7.980	78 0.14948 -8.254	77 0.14298 -8.447	76 0.13648 -8.649	75 0.12998 -8.861	74 0.12204 -9.135	73 0.11409 -9.427	72 0.10880 -9.634
71 0.10350 -9.851	70 0.09820 -10.079	69 0.09186 -10.369	68 0.08551 -10.680	67 0.08128 -10.900	66 0.07705 -11.132	65 0.07282 -11.377	64 0.06784 -11.685
63 0.06286 -12.017	62 0.05953 -12.252	61 0.05621 -12.502	60 0.05289 -12.766	59 0.04907 -13.092	58 0.04525 -13.444	57 0.04270 -13.696	56 0.04015 -13.963
55 0.03760 -14.248	54 0.03517 -14.539	53 0.03273 -14.850	52 0.03030 -15.186	51 0.02820 -15.498	50 0.02609 -15.835	49 0.02399 -16.200	48 0.02189 -16.598
47 0.02049 -16.885	46 0.01908 -17.193	45 0.01768 -17.524	44 0.01618 -17.909	43 0.01468 -18.332	42 0.01368 -18.639	41 0.01268 -18.969	40 0.01168 -19.326
39 0.01063 -19.734	38 0.00958 -20.185	37 0.00888 -20.514	36 0.00818 -20.870	35 0.00748 -21.258	34 0.00678 -21.689	33 0.00607 -22.168	32 0.00560 -22.519
31 0.00513 -22.900	30 0.00466 -23.320	29 0.00419 -23.774	28 0.00373 -24.281	27 0.00342 -24.657	26 0.00311 -25.067	25 0.00280 -25.521	24 0.00251 -25.999
23 0.00222 -26.539	22 0.00202 -26.939	21 0.00183 -27.380	20 0.00163 -27.871	19 0.00145 -28.377	18 0.00128 -28.945	17 0.00115 -29.375	16 0.00104 -29.847
15 0.00092 -30.383	14 0.00081 -30.919	13 0.00070 -31.520	12 0.00063 -31.987	11 0.00056 -32.499	10 0.00049 -33.079	9 0.00044 -33.590	8 0.00038 -34.170
7 0.00034 -34.677	6 0.00030 -35.274	5 0.00025 -35.940	4 0.00022 -36.478	3 0.00019 -37.128	2 0.00017 -37.768	1 0.00015 -38.285	0 0.00000 -infinity

Table 6.1. Gain levels for different audio mixer level settings. The first number in each cell of the table refers to the mixer setting, the second to the linear gain level, and the third to the gain level in dB (I. Doornekamp).

6.3 Group Additive Synthesis

Having a separate frequency and amplitude envelope for every sinusoid in the set of overtones is very resource intensive. This is a serious problem in systems like the Nord Modular in which these resources are limited. One way in which to increase efficiency is to group sets of overtones and associate just a single amplitude/frequency envelope pair to each group. This grouping makes the design process more difficult, as the designer must now identify those overtones which have similar frequency and amplitude dynamics. One commonly used grouping technique is to identify groups consisting of overtones that are harmonically related. The advantage of such groups is that they can be generated using single, filtered, non-sinusoidal, waveforms. This approach can be thought of as a hybrid of additive synthesis and subtractive synthesis, where the subtractive process controls the spectral properties of the individual groups, and the additive process combines different groups to obtain the overall desired result.

Figure 2 shows an additive synthesis patch that divides 24 partials into 5 groups, each with its own amplitude envelope. A manual re-synthesis approach was taken for the design of this patch. A piano sample was examined using the spectral analysis capabilites of Cool-Edit. Spectra were obtained at 50 millisecond intervals over the length of the sample (to give 20 spectra for a one second long sample). First, a spectrum from near the middle of the sample was examined to determine the location of the strongest spectral peaks. The relative amplitudes (in dB) of these peaks were measured. These amplitudes were used to set the levels of the sine-bank oscillators, using the conversion factor given in table 6.1. Then, the peaks were tracked (manually!) in the other time window spectra by a nearest-neighbor heuristic (the peaks don't shift very much in this sample) and their amplitudes roughly plotted. The partial groups were assigned from the amplitude plots by grouping together spectral peaks whose amplitude curves looked qualitatively similar. Multi-stage envelope modules were used to implement amplitude curves that more-or-less matched the general shape of the partial amplitude plots. The resulting sound is something like a piano, at least more like a piano than a saxophone. The sound is rather organ-like, a common result of additive synthesis.

6.4 Morphing

Additive synthesis may be used to great effect to synthesize a single type of sound created by an instrument, but has difficulty in synthesizing changing sounds. Most musical instruments are *expressive*, which means that their sounds can be altered by the performer. Many synthesis techniques, such as FM synthesis, can be controlled with a few parameters. These parameters affect the sound in a global fashion. In additive synthesis, the parameters determining the sound are the partial amplitudes, phases, and frequencies. Changing a single one of these will have little effect on the sound, unlike the changing of a single parameter in an FM synthesised instrument. In order to get an expressive change in the sound of an additive synthesized instrument, a large number of parameters need to be changed at once - a difficult task for the performer!

A solution to the problem of adding expressiveness to additive synthesis is *morphing*, where the performer controls a morph, or smooth transition, between two different sets of additive synthesis parameters. One could also use an approach similar to wave-terrain synthesis, in which the performer specifies a one-dimensional *trajectory* or curve in the high-dimensional space of additive synthesis parameters. Different points in this space correspond to different sounds, and in this way the performer can move from one sound to the next.

Morphing can be accomplished in many ways. Two of the most common are cross-fading of two different waveforms, and varying, in a concerted manner, a number of parameters between two different settings. In the Nord Modular the cross-fade module can be used to implement the first type, while the Morph controller groups can be used to implement the second type. It is greatly preferable to use the Morph controller group than cross-fading, as cross-fading requires duplication of most of the patch. Besides using the Nord Modular morph groups, morphing of voltage-controllable parameters can be implemented using cross-fading of the parameter values between two different levels. This approach is more tedious to implement than the morph groups but is very flexible, and can control a large number of parameters, whereas the morph groups are limited to controller a maximum of 25 parameters in the Nord Modular.

The patch shown in figure 6.2 is an example of such a morphing approach. The morph group is assigned to knob 1 in this patch, but could also be assigned to a MIDI controller such as key velocity or aftertouch. The sound morphs from the (re-synthesized) piano sound to a rather clangourous sound. Note the switch (controlled by knob 2) that enables syncing of the oscillators. When the morph knob is fully clockwise (to play the piano sound), the sync switch has little effect. This is because most of the partials are harmonic. When the morph knob is fully clockwise, so that the clangorous sound is being played, the sync switch has a significant effect.

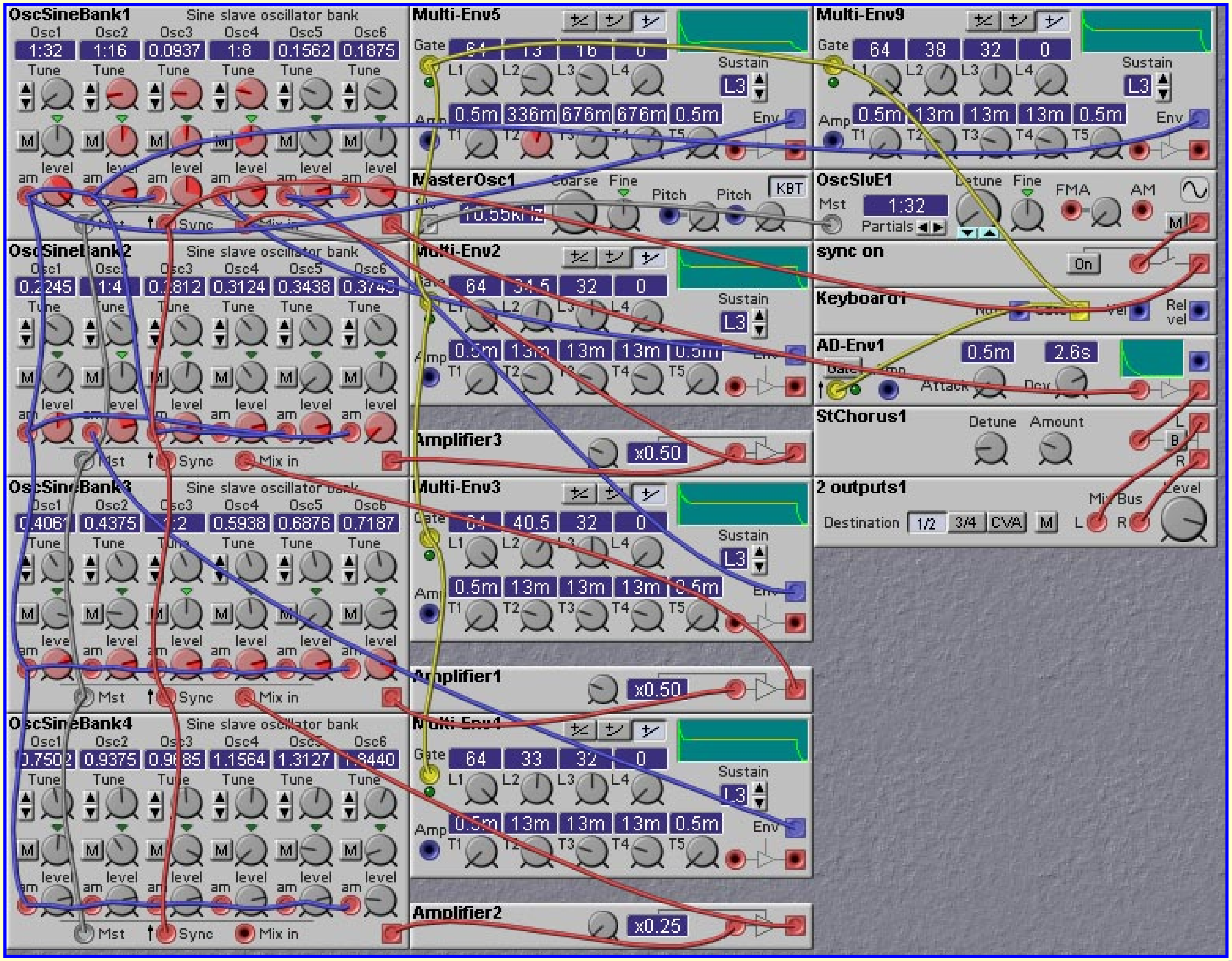


Figure 6.2. An additive synthesis patch illustrating grouping of partials, and morphing of parameter settings (partial amplitudes and frequencies) (J. Clark).

6.5 Adding Transients

It is difficult to create sounds having percussive attacks just with the addition of sine-wave partials. Taking a page from the Roland D-50 synthesizer, which combined subtractive synthesis steady tones with PCM samples of transient sounds, we can combine additive synthesis sound generation with subtractive synthesis generated transients. This is illustrated in the following patch:

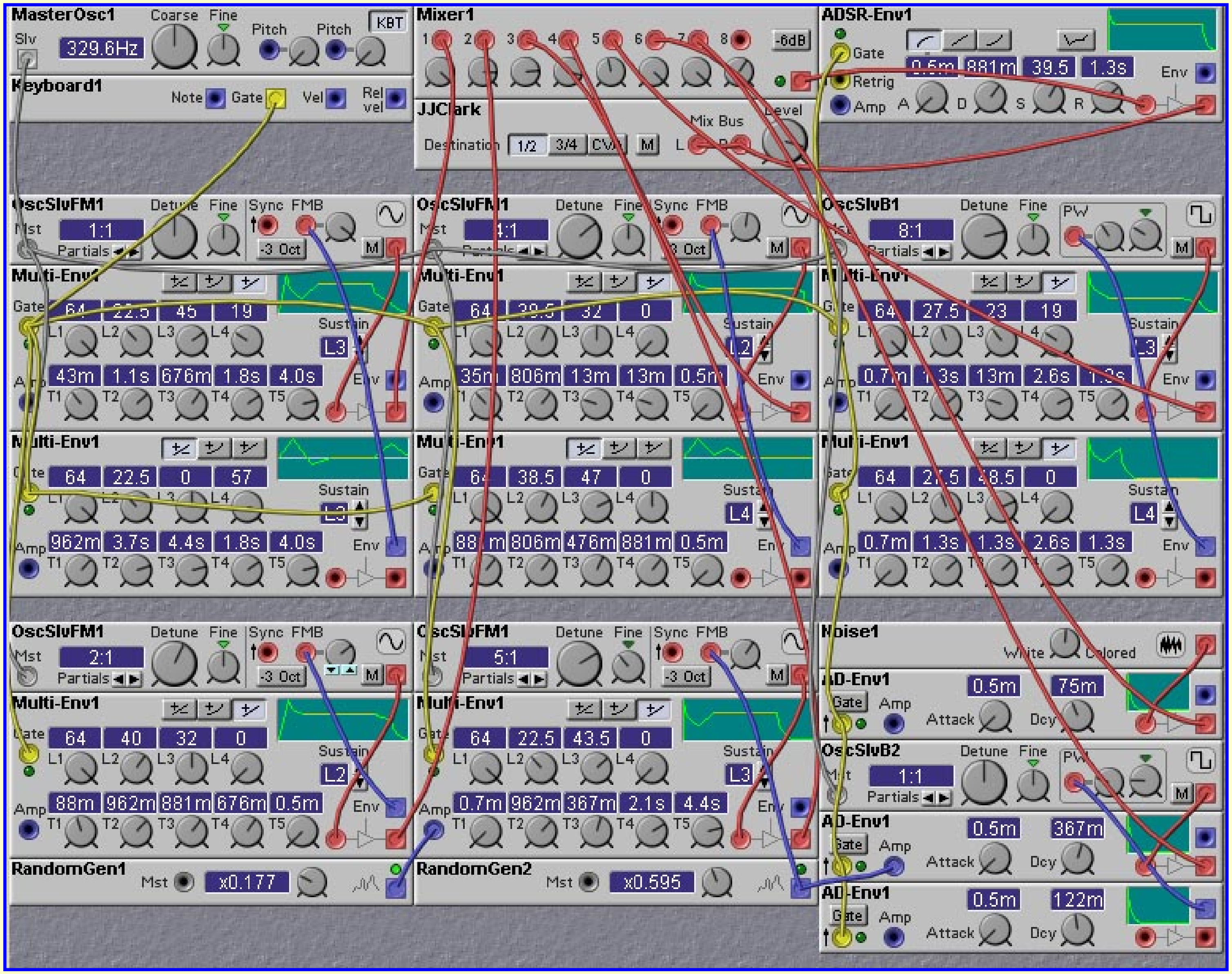


Figure 6.3. Adding transients to an Additive synthesis patch (J. Clark).

In this patch, an impulsive noise burst and a short pulse-wave transient are mixed with a simple 5-voice additive synthesis sound. The fifth partial of the additive part is implemented with a pulse generator rather than with a sine-wave. This is a useful trick when you have a limited number of partials to work with - let the highest partial be a non-sinusoidal wave, with lots of harmonics. This non-sinusoidal partial then fills in the higher harmonics. Otherwise, the sound would be dull and lacking in sparkle. A couple of LFOs are added to give some extra motion to the sound.

6.6 Which Oscillator to Use?

In the Nord Modular there are a wide range of oscillator modules to choose from. Which of these modules are best for implementing additive synthesis?

The slave sinewave oscillator is good for additive synthesis for a number of reasons:
- It produces a single sinusoidal harmonic.
- It is the single oscillator that uses the least DSP cycles (3%), and uses just a bit more DSP resources than 1/6 of the sine-bank.
- It has an FM input, which can be used to provide dynamic variation of the pitch of the partial.
- It has an AM input, which can be used to control the amplitude of the partial.

The slave FM sinewave oscillator is also very good for additive synthesis. It is slightly more expensive in terms of DSP usage than the non-FM slave sinewave oscillator (3.2% vs. 3.0%) but has one important advantage, that is, it has oscillator sync. This can be used to ensure that all harmonic partials will have a fixed relative phase. Be careful in using oscillator sync on non-harmonic partials, as this will introduce extra harmonics which will be hard to control (of course, these extra harmonics might just give you that sound you are looking for!). The FM sine slave oscillator also has an FM input, which can be used to provide dynamic variation of the pitch of the partial. Use this carefully if you are also using oscillator sync, as the FM then will not cause a change in pitch, but will create extra harmonics.

The number of oscillators that we can use in a Nord Modular additive synthesis patch is limited by the available DSP resources. But what if your grand scheme for the greatest patch ever requires more oscillators than this? Well, one way to get a larger number of oscillators is to go cheap and dirty and use LFO modules as the partial generators. An example is shown in the patch below, which uses 64 slave LFOs. You could probably squeeze in more oscillators, but I think you get the point. Triangle oscillators are used as they use the least DSP cycles of any of the LFO slave oscillators. Triangle waves have some low-amplitude higher harmonics, which limits the types of sounds that can be achieved. This slight drawback is insignificant in face of the other drawbacks of using the LFO slave oscillators - the aliasing that is present at higher frequencies, and the limited control one has over the frequency of the harmonics. This patch seems best for generating noisy clangourous sounds! One might question the use of audio mixers in this patch. Wouldn't it be cheaper to use control mixers? Yes, it would, but it would not make any difference to the overall number of oscillators or polyphony that we can achieve, since this is set by the zero-page usage rather than by DSP cycles. And using 8-input audio mixers is much more convenient and simpler to wire up than the 2-input control mixers that are available. You need 7 of these control mixers to construct an 8-input mixer, and the amount of wiring is considerable. Note that group synthesis is used in this patch - the oscillators are partitioned into groups of eight, and each group is amplitude modulated by a single envelope generator.

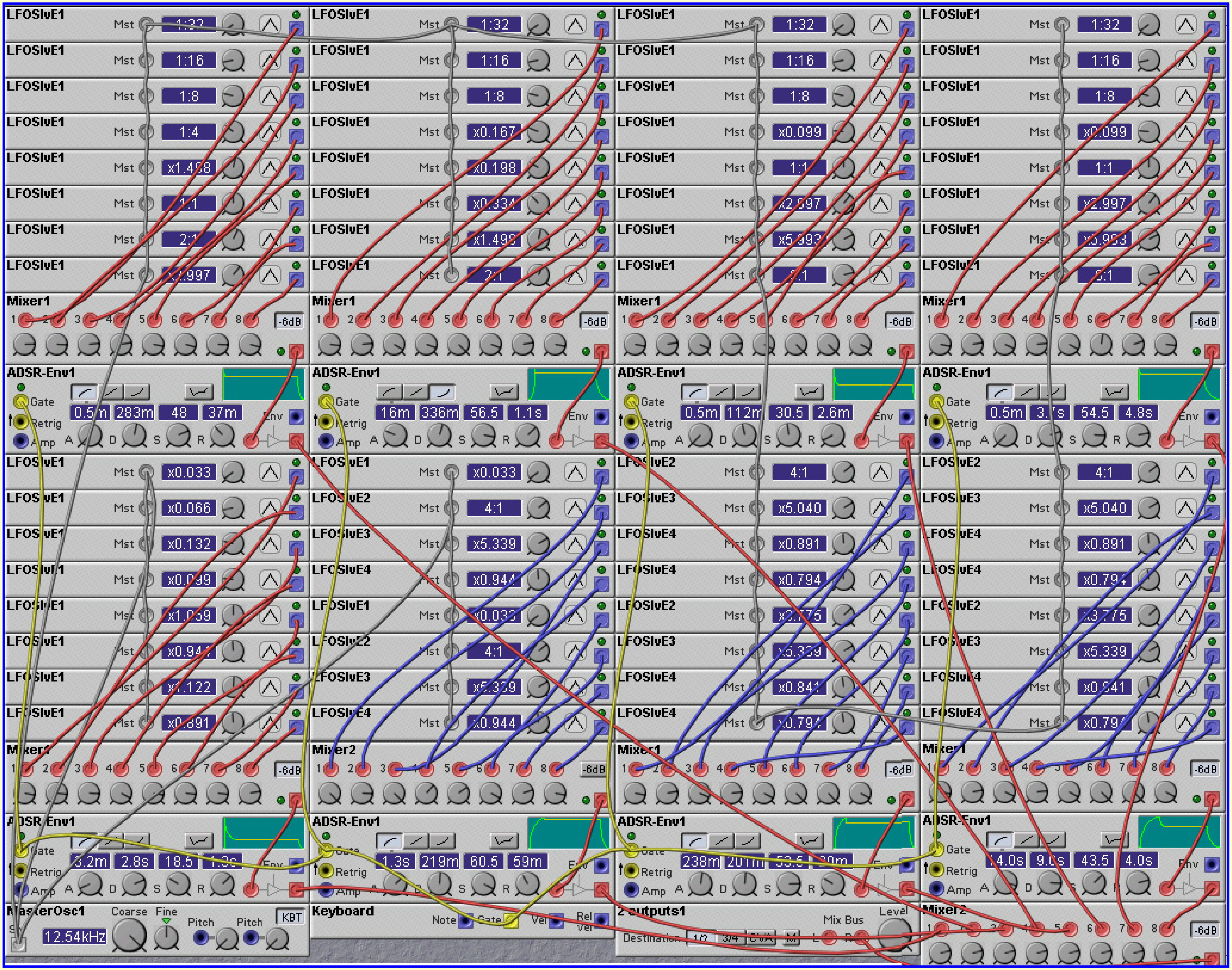

LFOSlvE1
Mixer1
ADSR-Env1
LFOSlvE2
LFOSlvE3
LFOSlvE4
Mixer2
MasterOsc1
Keyboard
2-outputs1

Figure 6.4. Additive synthesis patch using LFOs as partial sources (J. Clark).

[Prev (Percussion Synthesis)][Next (Physical Modeling)][Table of Contents]

Chapter 7. Physical Modeling

Physical modeling refers to a synthesis technique that aims to model, using simple mathematical approximations, the physical processes that give rise to the sound in a real acoustic instrument.

7.1 Introduction to Physical Modeling

The physical modeling approach can be applied, in principle, to any musical instrument, the most success has been obtained in modeling the physical processes involved in stringed instruments and in woodwinds.

The reason for the relative success of physical modeling in synthesizing strings and woodwinds is that the sound generation process in these instruments can be described as an excitation of a transmission line or waveguide. The mathematical equations describing such structures are rather simple (wave equations) and straightforward to implement computationally.

In a waveguide or transmission line model, the vibrating portions of an acoustic instrument are broken up into short segments. In each of these segments the incoming wave is partially transmitted to the next segment, and partially reflected back to the previous segment. A portion of the input is also lost (neither reflected or transmitted) as heat. In more complex models, a portion of the input can also be stored, typically in the elastic material making up the instrument. These complex models can be very nonlinear and difficult to control. They are also very challenging to implement computationally in a stable manner.

Another important aspect of the waveguide segments is the propagation delay time for the input to make its way to the next segment. This delay time can also vary with the frequency of the signal. Frequency dependent delays result in spectral dispersion, which can be useful in modeling complex sounds such as cymbals.

The relative amounts of input that are transmitted, reflected, lost, or stored in a waveguide section, along with the delay time of the section, are the parameters that determine the sound of the instrument. These parameters can also vary with the frequency of the input. For example, more high frequency energy can be lost than low frequency energy, resulting in a sound that progressively gets duller. By piecing together a string of such sections, with appropriate values of the parameters, we can get a sound which closely approximates that of a given acoustic instrument.

7.2 The Karplus-Strong Algorithm

Before plunging ahead into all the details of physically modeled instruments, let us look at a precursor to modern digital waveguide designs, the Karplus-Strong approach. The Karplus-Strong algorithm was developed as a simple model of the oscillation of a plucked string. The basic element of the Karplus-Strong model is a short delayline whose length is equal to that of one cycle of the waveform to be produced. The delayline is initially filled with noise, or with a triangular waveform, whose peak corresponds to the location of the plucking action being modeled. The average value of the initial delayline values should be zero. The contents of the delay line are then read out serially, providing the first `cycle' of the sound. The delay line is then re-filled, but this time with an averaged waveform obtained by adding two adjacent delayline values together, dividing the result by two and put the result back into the first of the two locations. The delayline is then read out, providing the second `cycle' of the sound. This averaging and delayline readout process is repeated continually. Eventually all of the delayline values will approach the average value of the initial delayline waveform. That is, they will approach zero, and the sound will be perceived to have died away. The rate of decay depends on the details of the averaging process and on the length of the delayline. A shorter delayline will create a faster decay than a longer one. The pitch of the sound will also depend on the length of the delayline, with longer delaylines giving lower frequencies. The timbre, or brightness, of the sound depends on the shape of the initialization waveform.

One can combine the averaging process with the delayline readout and re-writing process. To do this, the delayline can be continually read-out and re-written, one sample at a time. The output of the delayline can be passed into a lowpass filter (which does the averaging) and the output of the filter fed to the input of the delayline.

Implementing the Karplus-Strong algorithm on the Nord Modular is simple. Connect a crossfade module in the feedbackloop of the delay. Switch the crossfade with a short pulse from the logic pulse module filling the delay with some sound, which might be noise, or your voice. Then you hear this little 'soundsample' play a frequency that is controllable by the delaytime. To attenuate the input to the delay line in a frequency dependent manner a simple 6dB fixed lowpass filter module can be used. This filter's response is very similar to the 'summing and divide by two' method, but is more flexible, as by varying the frequency of the filter one can obtain a longer or shorter decay time.

Such a simple implementation of the Karplus-Strong algorithm is shown in the figure below:

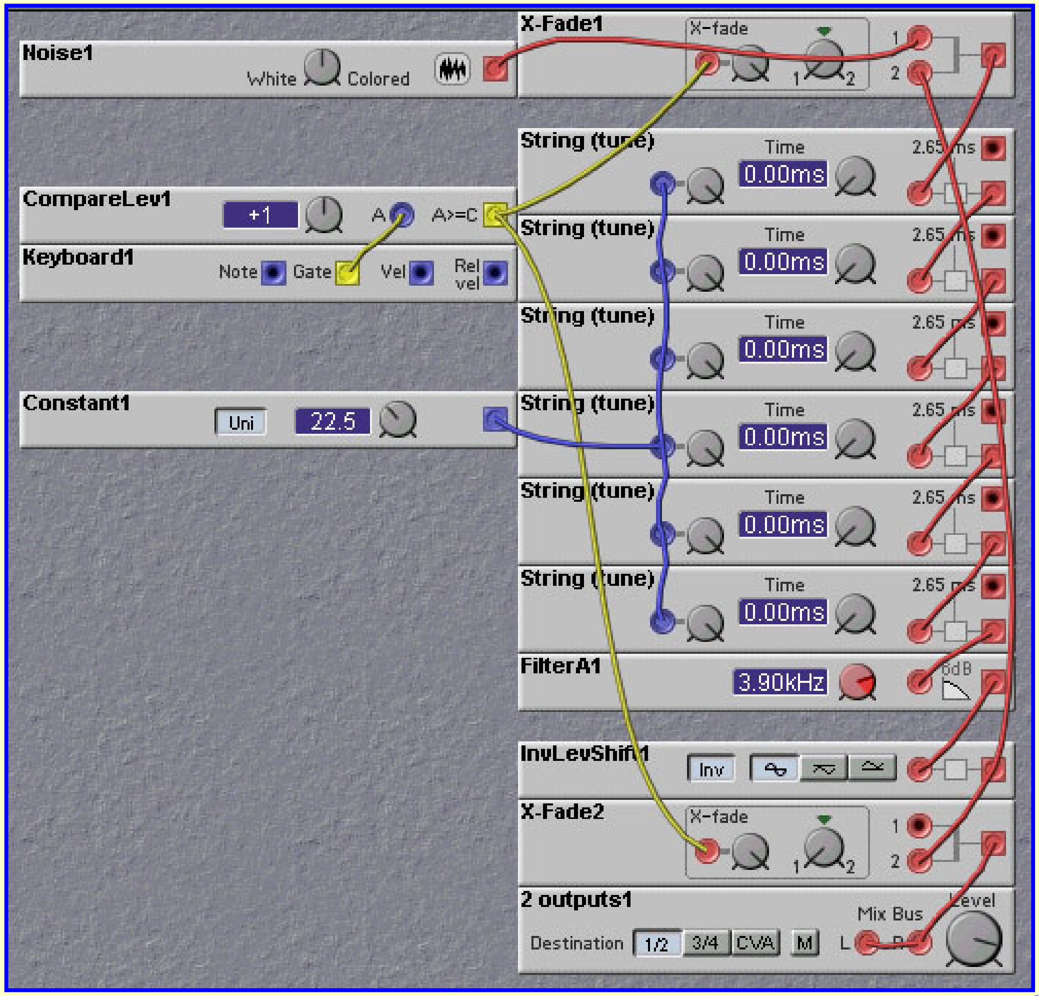


Figure 7.1. An implementation of the Karplus-Strong algorithm (J. Clark).

There are some improvements possible on the algorithm, but sadly not on the Nord Modular. For instance, by having a table (looped delayline) of fixed length and separating the averaging routine and the table readout routine reading out in variable interpolated steps the decay is no longer frequency dependant but can be varied by the speed that the table is averaged. Also there is better control over the frequency that is hard to control on the Nord Modular. By using two tables and 'pre'-filling one table and switch to the new table on the next gatepulse, the averaging can start immediately without having to wait to fill the table. That definitely improves the attack with longer tables and non-noise audio-input. It can sound much better that it does now. When the averaging goes with less speed a nice very lifelike and an `acoustic' soft phasing is introduced in the sound.

Sean Costello points out that one can simulate guitar feedback with a simple Karplus-Strong algorithm (this was described in a CMJ article in the early 90's). To do this, feed the output of the Karplus-Strong delay lines (one for each of the 6 strings) into a nonlinear shaping function that simulates an overdriven amplifier and fuzzbox. You can also compress the output to provide *sustain*. Feed a portion of the output into another delay line, to simulate the distance from the amplifier to the "strings". This delay line then feeds back into the Karplus-Strong delay lines. By controlling the amount of the output fed into the delay line, and the length of the delay line, you can control the intensity and pitch of the feedback note. A Nord Modular patch that implements this feedback guitar is shown below. In this patch knob 3 controls the "distance" between the simulated amp/speaker and the string. Changing the distance changes the feedback delay time as well as changing the amount of feedback.

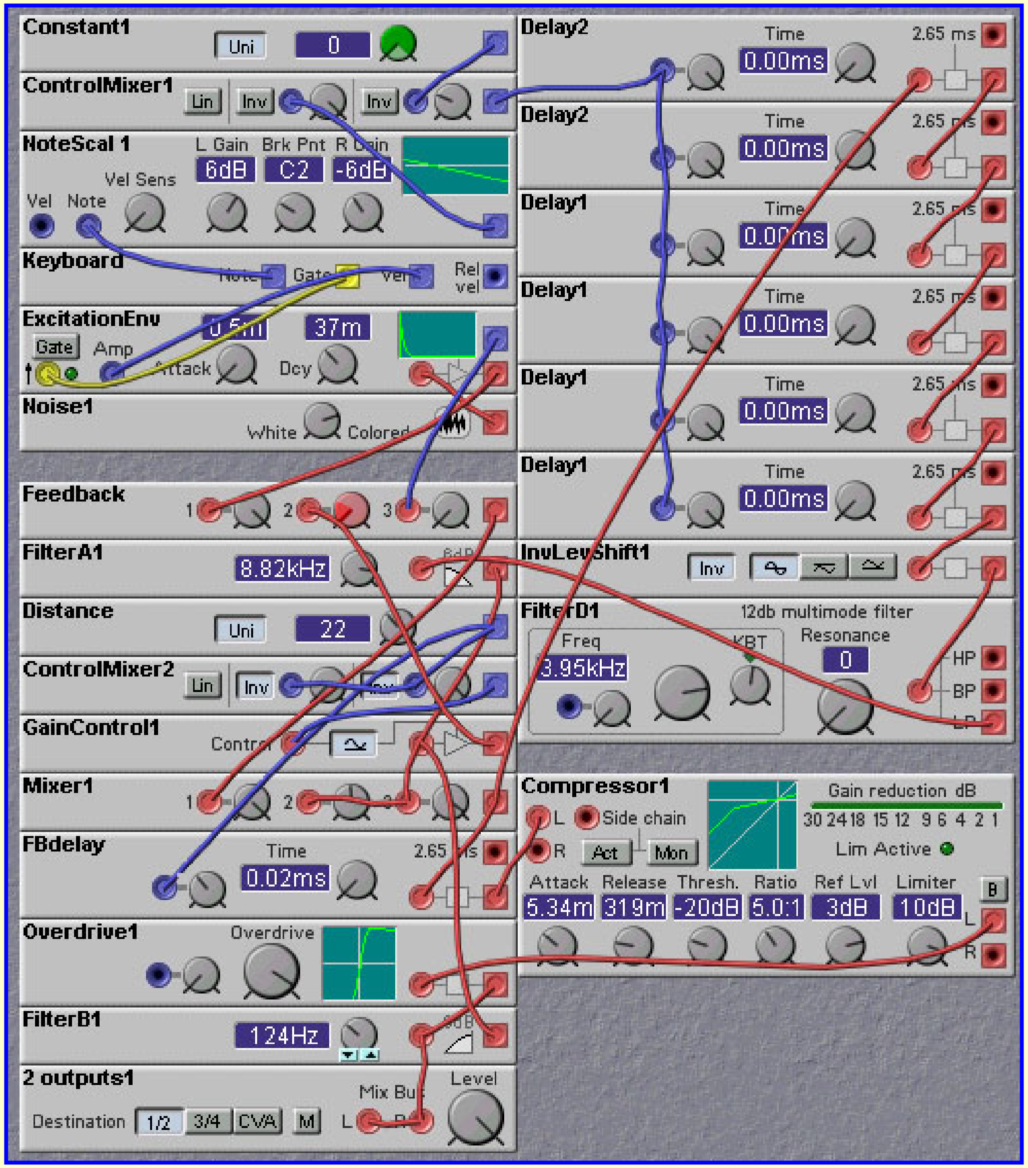


Figure 7.2. Guitar feedback emulation with a Karplus-Strong algorithm (J. Clark).

7.3 Tuning of Delay Lines

The pitch of acoustic sounds generated with digital waveguide models depends mainly on the delaytime of the waveguide segments. Thus, tuning the instrument involves setting the proper delaytimes. On the Nord Modular, this is a bit trickier than it may seem at first, due to a number of factors.

The first factor to consider is that the delaytime of the Nord Modular delaytimes is a linear function of the delay control input. But the pitch of the resulting sound is inversely proportional to the delaytime. Thus, we need some way of computing the reciprocal of the pitch control value (e.g. from the keyboard module). Clavia does not (yet) supply an inversion module. Fortunately, there is a module that can be coerced into doing what we want. *Richard Thibert*, in his "physicflute" patch, discovered that a NoteVelScaler module can be used to implement the required reciprocal computation. More precisely, it computes the reciprocal of the exponential of the note number, which is actually what we want, since the pitch is exponentially related to the note number.

This approach to tuning is illustrated in the following Nord Modular patch, made by *Rob Hordijk*:

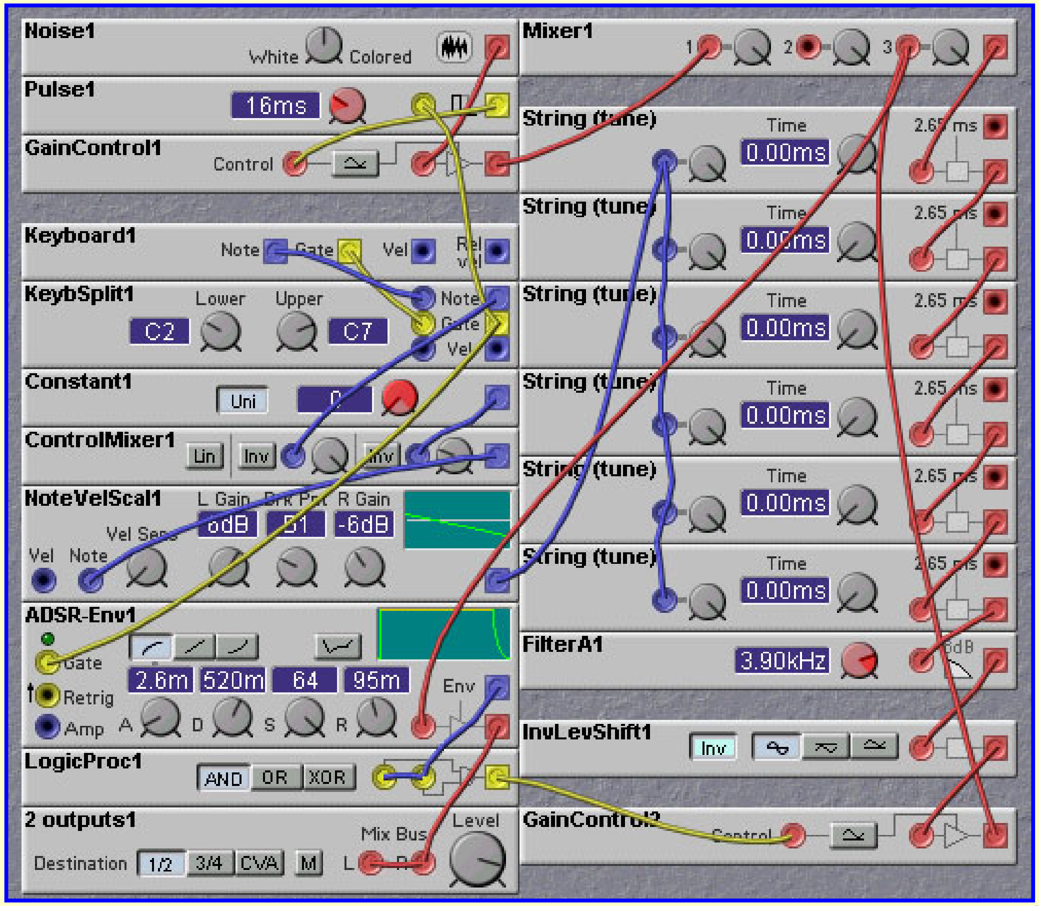


Figure 7.3. Tuning of a Karplus-Strong delay-line using the NoteVelScaler module (R. Hordijk).

Note that the gain of the NoteVelScaler has been set to -6dB/octave. This value gives the desired reciprocal operation. A short pulse of noise is inserted into the delay line whenever a key is pressed. This simulates the plucking of a string. The feedback level of the delayline back into itself can be adjusted. This sets the decay rate of the waveform. If the feedback is too high, the loop can become unstable. The filter in the loop controls the relative rate at which high and low frequencies become attenuated. Adjusting the cutoff frequency of this filter controls the `brightness' of the sound.

Another, more subtle, factor influencing the tuning of physical models are the computational delays inherent in the various operations going on. For example, the filters that are used to provide the frequency dependent attenuations generate some delay. Since the filter delay is not influenced by the delayline delay time (since they are independent modules) the result will be a fixed shift in delay, causing a detuning of the sound, especially at high frequencies, where this fixed filter delay is relatively larger in comparison to the delay line delay time. This high frequency detuning can be compensated for somewhat by adding in an extra pitch-dependent delay time (what in the older days was called 'high frequency-tracking') or by slightly reducing the control value being sent to the delaylines by small amounts at high pitch values. In the above patch this is accomplished by adding in a note-value dependant `constant' to the tuning signal coming from the NoteVelScaler module.

Another issue to consider when trying to tune delay lines is whether the delayline has a variable length, but constant time step, or a constant length with variable time step. Think of modeling a guitar. You can increase the pitch either by tightening the string (which increases the propagation speed) or by making the string shorter. When working with a variable length/fixed step delayline the decay of the plucked string gets shorter in proportion to the pitch of the sound. But in a guitar the high strings have more or less the same decaytime as the low strings. So for KarplusStrong patches it's much better to work with a fixed length / variable step delayline. Then you can have the whole frequency range from very, very low to over 20kHz within a 256 sample delayline! It also gives the possibility to 'eat' the samples up with a steprate independent of the frequency! Unfortunately, this is not an option with the Nord Modular, as the only delayline modules that are available are of the variable length fixed step size type.

Finally, one should consider the total range of pitches that are available to physically modeled instruments on the Nord Modular. The maximum frequency depends on the minimum delay of a string of delay-lines. This is not a problem, as one can readily make an instrument that can produce supersonic pitches. The major difficulty arises when trying to achieve low pitches. The lower the required pitch the longer the required delay line. Because of the amount of memory and DSP resources used by the delay line module, one can only string together at most 6 delay lines. One trick of waveguide synthesis is to make sure that the reflected portion of the waveguide segment has an inverted sign relative to the input. This will effectively drop the pitch by one octave. This trick also cancels 'DC' components that could build up while feeding back, if the average DC of the input signal would be different to zero level (which would eventually force the delayline to overflow).

7.4 Delay Line Details

The delay line modules provided by Clavia in the Nord Modular are the fundamental components of any physically modeled instrument, so it is useful to know a little about their characteristics.

The first thing that people notice about the delay line modules is that they use a relatively large proportion of the systems's DSP cycles and memory. The reason for the high memory usage is basically that the Nord Modular has very limited memory available resources, so it doesn't take much to use it all up.

The high DSP load for the delay lines may appear to be puzzling at first, especially when you consider that to compute a delay line should only require one memory read, one memory write, and one pointer increment. But the Nord Modular delay line module has the ability to have delay times that are non-integer (i.e. fractional) multiples of the sampling period. In order to achieve this high resolution, interpolation is performed, which involves filtering of the data stored in the delay line. This filtering is what consumes most of the DSP cycles. This DSP usage is a good price to pay for the ability to accurately tune the delay lines. If interpolation was not done, then it would not be possible to have a properly tuned physically modeled instrument.

The actual resolution of the delaytime in the Nord Modular is very high. Although Clavia has not released any figures on this, it is estimated that the resolution is 2^19 steps in 64 control units, or 5.05 nsec over the 0-2.65 msec range.

One of the drawbacks of using interpolation to obtain high delaytime resolution is that the interpolation always attenuates the signal in a frequency dependent manner. If you apply a Gaussian Blur (a form of weighted interpolation) to an image, the sharp edges and details -- the high frequencies -- get lost. The same takes place if you interpolate an audio signal.

After a single pass, the loss of high frequencies is not that noticable, but if you repeatedly pass the signal through the process of interpolation, the signal is attenuated, and eventually decays down to zero. In any delay line, there is always this trade-off between accurate delay time (giving acurate pitch in a physical modelling waveguide configuration) and accurate frequency response.

7.5 Physical Modeling with Digital Waveguides

The transmission lines or waveguides that make up a physically modeled acoustic instrument are easily implemented with digital filter structures. These have been termed *digital waveguides*. The typical form of a digital waveguide is shown in the figure below.

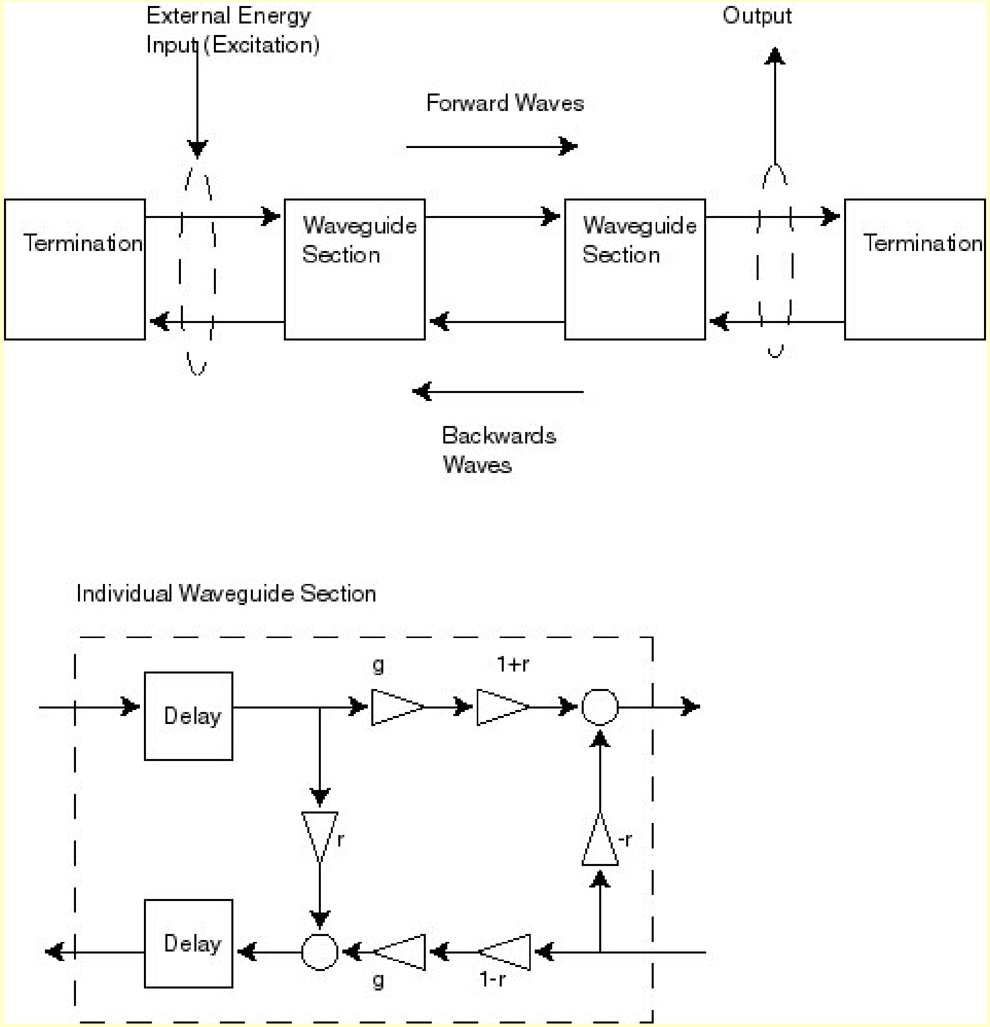


Figure 7.4. Digital waveguides consist of digital filter implementations of lumped waveguide sections.

Making a physically modeled instrument consists of connecting a series of these waveguide sections. Note that each waveguide section has two signals going into it, and two signals going out. These two sets of signals model the waves travelling in one direction, and the waves travelling in the opposite direction. Each section contains two summers, six attenuators, and two delay lines. The delay lines models the time it takes for the wave to propagate from one end of the section to the other, in either direction. The attenuators with values g and -g model the losses that may occur in the waveguide. For example, energy can be lost in a vibrating string by acoustic radiation, or by heating the surrounding air. These losses can be frequency dependent, and typically losses are greater for high frequencies than for low frequencies. In this case the attenuators would be implemented with lowpass filters. The other attenuators (with values of r, -r, 1+r and 1-r) model the scattering of the signal at the junction between two sections - part of the signal is transmitted across the junction, while the rest is reflected back in the opposite direction. These attenuations are also usually frequency dependant, that is, the relative proportion, r, of signal that is reflected at a junction depends on frequency. Thus these attenuators are often implemented with filters. Typically the reflection coefficient r has a lowpass characteristic (and therefore 1-r has a highpass characteristic), modeling the usual physical situation that high frequencies are transmitted preferentially, and low frequencies are reflected preferentially. Different sections can have different characteristics. This change in characteristic from section to section can model physical systems that have changing geometry, such as a woodwind with a non-constant bore diameter. A termination of the waveguide can also be modeled in this way, by setting the reflection coefficient to r=1, thereby causing all of the energy to be reflected. In some instruments, such as brass instruments, sound can be radiated from the termination point. In this case r is not set to one, but is set to a lowpass characteristic, with the low frequency response equal to one. The transmitted energy will have a high pass characteristic and will form the output signal.

7.6 String Modeling

Perhaps the most straightforward instrument to model with a digital waveguide is a string. A string is modeled as a set of waveguide sections connecting two special `termination' waveguide sections. The termination sections model the rigid connections at either ends of the string. The model for these terminations is quite simple - just an inverter. This inverter implements the inversion of the wave that occurs during reflection at the rigid termination. This inversion occurs because at, the end point, the forward and reflected waves must sum to zero in order to keep the end point from moving. This is equivalent to setting the reflection coefficient of the terminating waveguide sections to r=-1.

The reflection coefficients for the intermediate waveguide sections are set to zero. This models the fact that the strings are uniform along their length, and do not have any discontinuities that would cause reflections. The only reflections occur at the termination points of the string. The loss coefficients k are taken to have a lowpass characteristic, modeling the loss of high frequency energy as waves propagate along the string.

The string is set into motion by creating an initial displacement somewhere (anywhere) along the string. This models a percussive `striking' of the string.

The output can be taken from any point along the modeled string, and is obtained by summing together the forward and backward waveforms (as these together determine the string displacement). The overall model for the string is shown in part A) of the figure below.

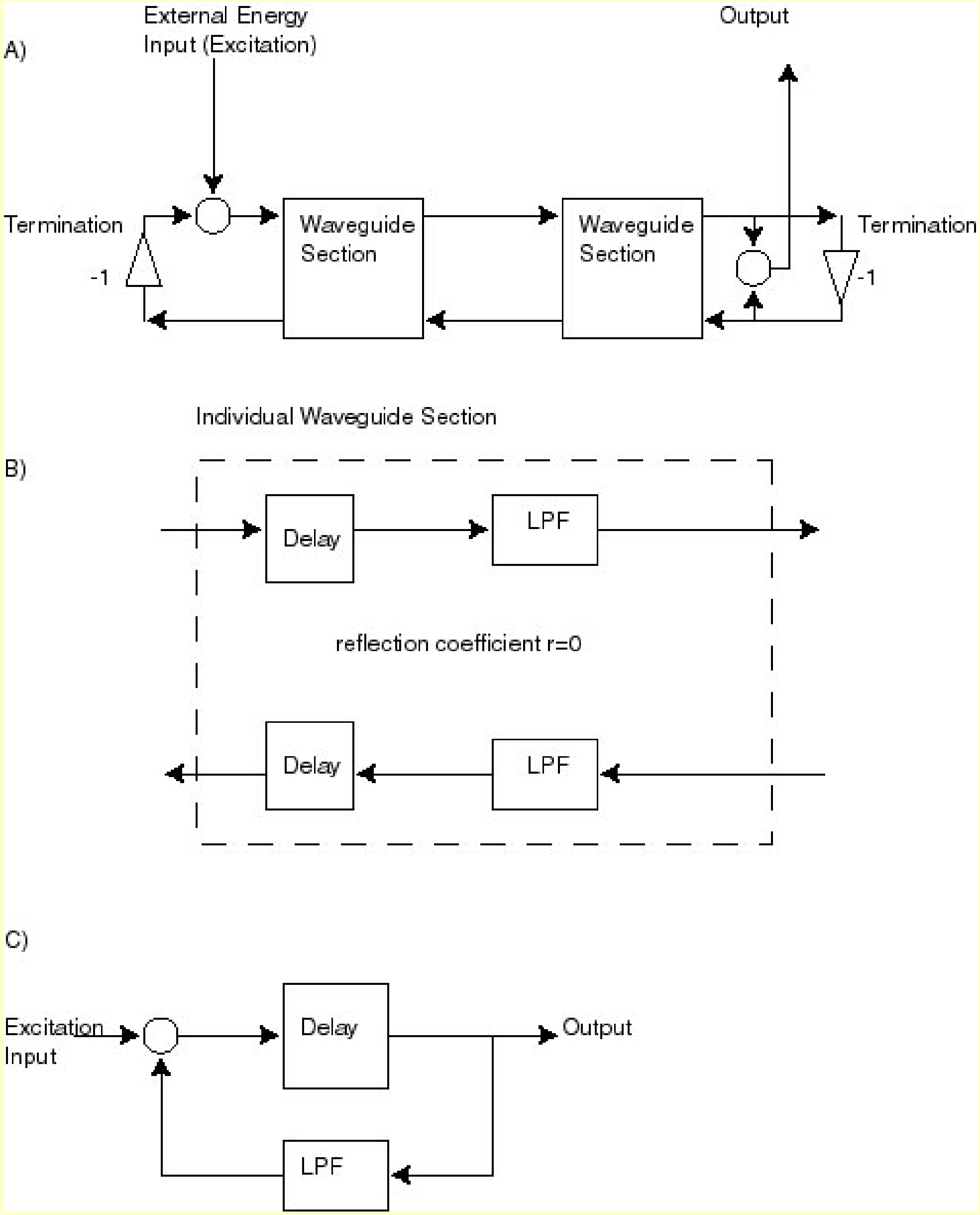


Figure 7.5. Digital waveguide model of a vibrating string. A) A multiple-section model. The inverters at either end model the rigid terminations of the string, which requires that the amplitudes of the forward and backward going waves at these terminations sum to zero. B) The individual waveguide sections consist of two delay lines and two lowpass filters. The reflection coefficient is set to zero, modeling the uniformity of the string. C) A simplified model, using the linearity of the network to combine elements.

This model is more complicated than it needs to be, however. The circuit is completely linear in the mathematical sense. This means that we can combine all of the delaylines into one delayline. Similarly, we can combine all of the lowpass filters into a single lowpass filter, which models the overall frequency dependent losses. We can even combine the inverters, which, since their effect cancels, means that we can remove them. Doing these modifications results in a much simpler model, shown in part C) of the figure above. This simplified model is implemented in the following patch:

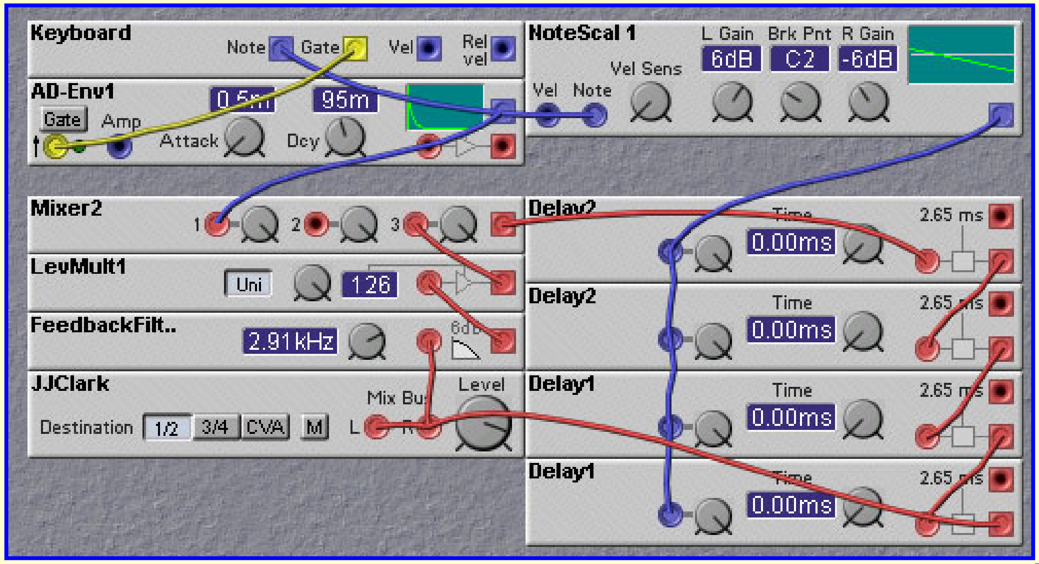


Figure 7.6. A Nord modular patch implementing a lumped digital waveguide model of a vibrating string (J. Clark).

Notice the similarity between this patch and the Karplus-Strong patches shown earlier. The only real difference is in the way in which the string is excited, or struck. In the Karplus-Strong patch, white noise was used to initialize the delayline (and hence the initial string displacement). In the digital waveguide patch, the initial displacement is an impulse at one end of the string. The impulse could be modeled as occuring anywhere along the strong. But to do this, one needs to break the waveguide into two parts - one before and one after the point of the impulse. Thus it will be a somewhat more complex model than the simple one shown above. Using an impulsive (in space, along the string) excitation is not very realistic. In a real guitar, for example, the string is excited by pulling it sideways at the plucking point. Thus the initial waveform is a smooth displacement along the length of the string. To accurately model such an excitation we would need a large number of waveguide sections, each with a small delay, and relatively little frequency dependance in the transmission and reflection amounts.

Coupled Strings

In most stringed instruments there is more than one string present. These strings do not oscillate independently of each other, but interact. The interact takes place through transmission of some of the energy of each vibrating string to the other strings. This transmission takes place through the body of the instrument that the strings are attached to. In a guitar, for example, most of this coupling takes place at the bridge, where the strings are fixed in close proximity. If the bridge was infinitely stiff there would be no coupling, but if it is somewhat non-rigid, part of the vibrational energy of a string will go into creating a small vibration of the bridge, which will then be transferred in part to the other strings. Coupling between strings can be implemented in a digital waveguide model as shown in the following diagram:

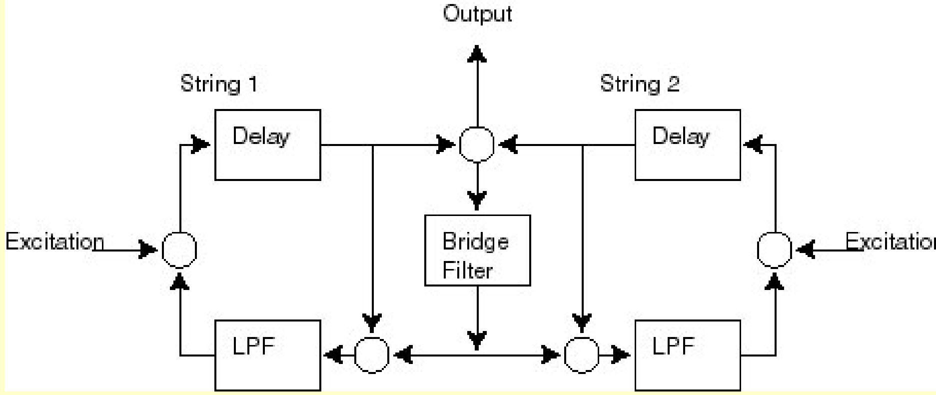
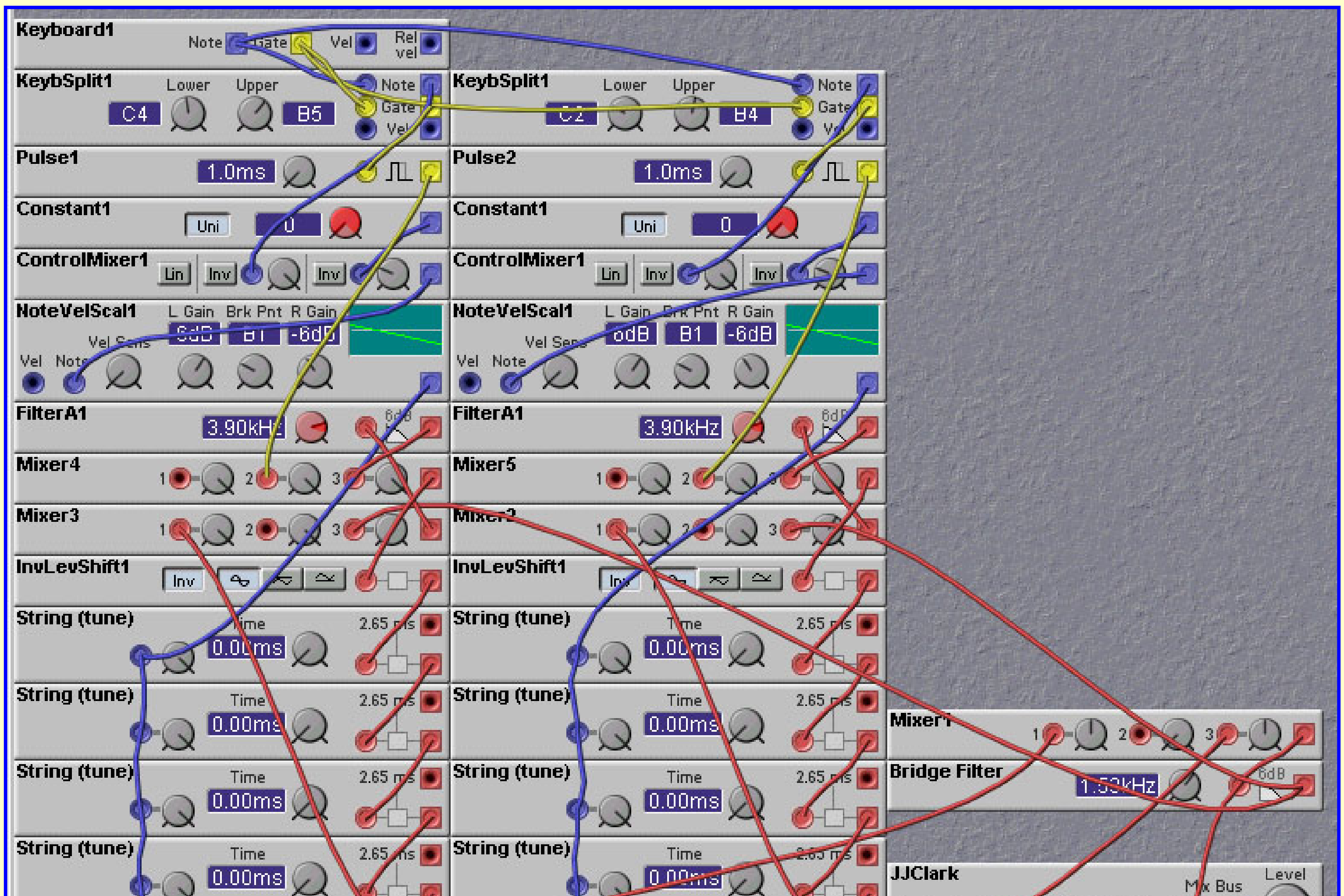


Figure 7.7. Physical model of two coupled vibrating strings.

A non-rigid bridge will introduce losses, and one can eliminate the loss filters in the individual string models, thereby simplifying the design. Only one filter is needed, that modeling the bridge. A Nord Modular patch implementing this simplified model is given in the following figure.



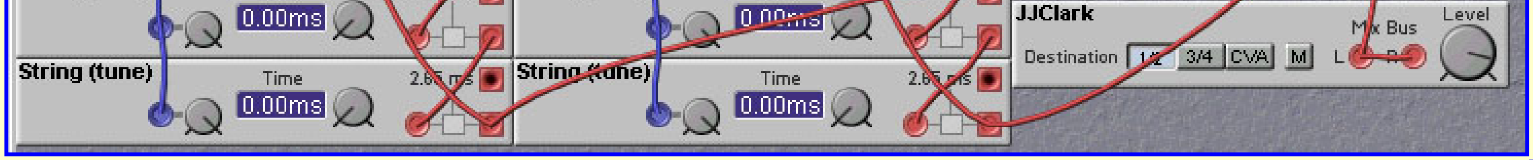


Figure 7.8. A Nord modular patch implementing coupled vibrating strings (J. Clark).

7.7 Woodwind Modeling

Woodwinds are also suitable candidates for physical modelling. As with strings, waveguide sections can be used to model the propagation of waves up and down the tube of the instrument. Woodwinds are somewhat more complicated to model than strings, because the bore of the woodwind can change along its length, whereas the string has a more or less constant thickness along its length. In addition, the bore can contain holes which also need to be modeled. That being said, a simple woodwind model looks very much like a string model. We can assume that the instrument bore has a constant diameter (as in a clarinet), and has no holes, except at the end. Clearly, the lack of holes is not a realistic assumption, as it doesn't seem to give us anyway of changing the pitch! We can cheat, however, and vary the pitch by changing the delay time of the waveguide section delays, much as is effectively done in instruments such as the kazoo or trombone.

The excitation process is more complicated in woodwinds than in strings. One does not `pluck' a clarinet, one blows into it. The key aspect of the excitation of a clarinet is the reed. The reed acts as a nonlinearity, affecting the flow of air into the bore, as well as the reflection of the pressure wave at the mouth end. In simple terms, as a pressure differential across the reed is built up by blowing into the mouthpiece, the amount of reflected energy increases. Further pressure increases, however, begin to close the reed, and the amount of reflected energy begins to drop, going to zero once the reed closes. This nonlinearity, driven by the pressure provided by the player, provides amplification or gain in the loop formed by the outgoing and reflected pressure waves. If the gain is high enough, oscillation can occur. This oscillation is what provides the basic tone of the clarinet. The frequency of oscillation depends on the total delay time of the loop. Unlike a string, the nonlinearity at the mouth end of the clarinet causes the creation of harmonics, giving the characteristic square wave tone of the clarinet. In a physical model on a computer or a Nord Modular, one can increase the gain of the nonlinearity, and create different regimes, including chaotic ones, where the frequency components created by the nonlinearity are inharmonic and noisy.

In the Nord Modular it is difficult to implement a precise model of the nonlinearity of the clarinet excitation, but one can easily implement an approximation. In the patch shown below we use the WaveWrapper module to provide an approximation to a cubic nonlinearity, which is known to work well in generating oscillations. We use the Shaper module to round off the corners of the WaveWrapper function a bit. The effect of the mouth pressure is faked somewhat as compared with what occurs in a real clarinet. In this patch the breath input (modeled as the output of the Envelope Generator module) is used to vary the gain of the nonlinearity. If the gain is too low, no oscillation will occur. In real clarinets the effect of the mouth pressure is to shift the nonlinearity along a 45 degree line. This is hard to implement with the Nord Modular, so we will stick with the simple approximation.

You should play around with this patch a bit, and especially explore the effect of changing the gain of the nonlinearity by adjusting the level of the third input to the Mixer module. It is quite easy to get chaotic sounds. A schematic diagram of this model is shown in the next figure, and the Nord Modular is shown in the figure following it.

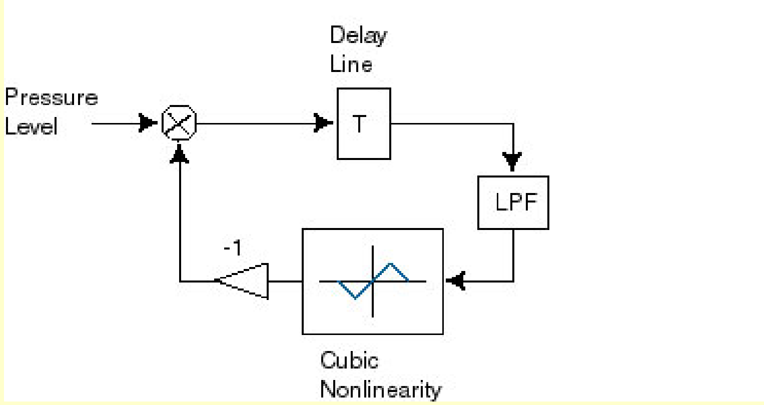


Figure 7.9. A simple digital waveguide model of a clarinet.

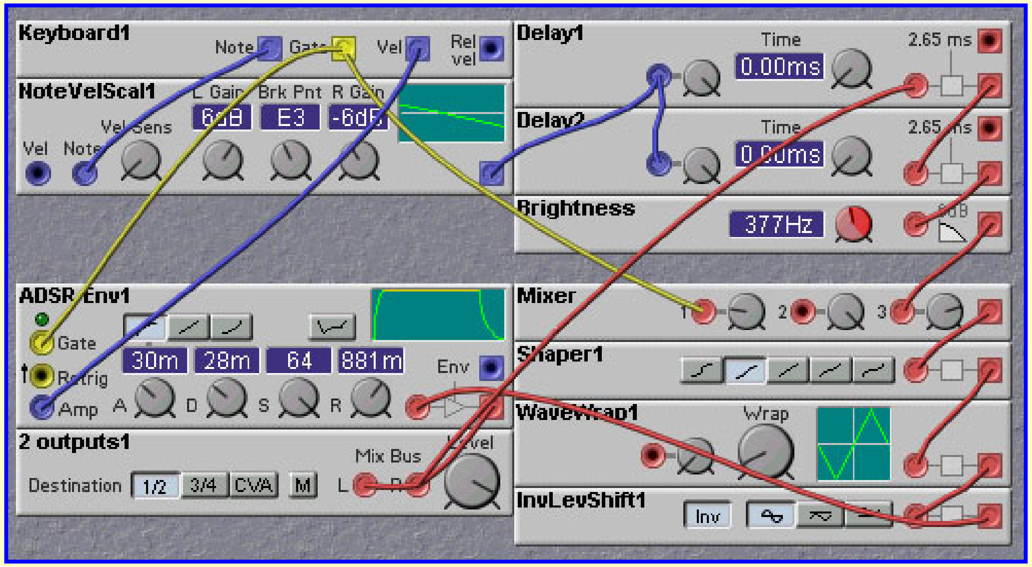


Figure 7.10. A Nord Modular implementation of the simple clarinet model (J. Clark).

Tone-hole Modelling

In the above patch, while it sounds like a clarinet, we cheated in modeling the tuning. The pitch of a real clarinet isn't changed by changing the length of the instrument bore! Rather, pitch changes are obtained by opening and closing `tone-holes' in the instrument body.

To a first approximation, one can consider the sound emitted by the instrument as coming entirely from the first open tonehole (nearest to the mouthpiece). The tonehole will transmit a portion of the energy and reflect the rest. Typically high frequency energy is transmitted and the low frequencies are reflected. Thus we can implement the tonehole as a pair of filters, a lowpass for the reflection and a highpass for the transmission. To reduce computational load, we can implement the highpass filter simply by subtracting the lowpass output from its input. To reduce computation even further we can model the tonehole just as a switch. When it is closed it transmits all of the energy, and reflects nothing, and when it is open it reflects all of the energy and transmits nothing. The losses can be lumped together in a single loop lowpass filter. The energy transmitted by each tonehole are summed together, although in the simple model being used only one tonehole at a time (the one closest to the excitation) will have any output. In a multiple tonehole instrument we can use switches to select which tonehole the sound will be reflected from/transmitted through. This simple approach is used in the patch shown in the figure below.

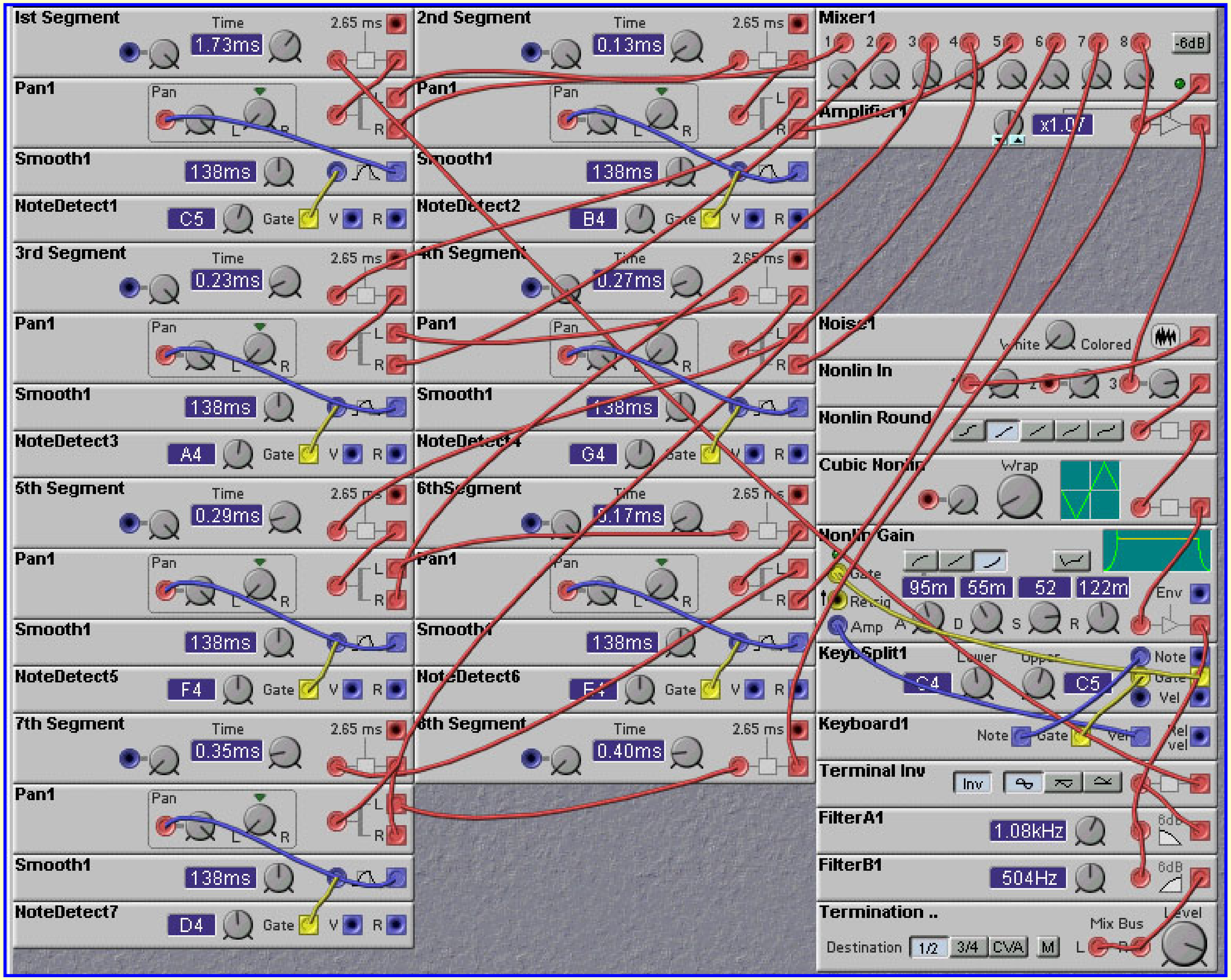


Figure 7.11. A digital waveguide model of a clarinet with toneholes (J. Clark).

In this patch NoteDetect modules are used to select which toneholes are *open*. This is opposite to what you would normally do with your fingers when playing an instrument with toneholes, i.e. press down your fingers to close the tonehole. The reason for doing it this way in the patch is to allow MIDI'ed wind-controllers to drive the patch. These controllers convert the closures on the instrument into corresponding MIDI note values. They do not transmit tone-hole closures. If you want to change the patch so that key-downs close the toneholes instead of opening them, you simply need to swap the outputs of each of the Pan modules in the patch.

Note that the outputs of the NoteDetect are passed through Smoothing modules before being fed into the Pan control inputs. This is to eliminate the objectionable elastic click which would otherwise occur when closing one tonehole and opening another. This click is caused by discontinuities in the audio DelayLine modules. Going from one note to another requires a re-configuration of the instrument's overall delay line structure. If one plays in a legato style, and moves from a low pitched note to a higher pitch, there will be little problem. Playing in a staccato fashion, or moving from a high pitch to a lower requires switching in extra delay line sections to the currently active ones. The currently active section has the ongoing wave passing through it, whereas the new sections will just have zeros in them. Thus there will be a significant discontinuity in the wave passing through the reconfigured delay line section. This discontinuity will be quickly smoothed out but it is enough to give a noticeable transient. Transients such as these DO occur in real instruments, but they are of a less objectionable nature than in our simple model. We will have to do a better job of modelling the tonehole-bore interfaces to obtain more natural transitions between notes.

Slide-Flute

Perry Cook has devised many physical models of musical instruments. One of them is the "Slide-Flute", which he described in the following conference paper:

Cook, P., "A Meta-Wind-Instrument Physical Model, and a Meta-Controller for Real Time Performance Control", Proceedings of the ICMC, 1992.

The slide flute model consists of two delay lines, one to model the flute's bore and the other to model the embouchure (the opening into which the player blows to sound the flute). The length of the flute bore delay is twice that of the embouchure delay line. A Nord Modular patch implementing this model is shown below. In this patch a wave-wrapper module is used to simulate the nonlinearity of the excitation and the interaction between energy bouncing back from the end of the flute bore with incoming breath pressure. The output from the bore delay line is fed back into the system in two places. The jet delay knob simulates changing the angle with which the player blows into the flute - thereby allowing overblowing techniques. In this patch the jet delay is also controlled by the mod wheel.

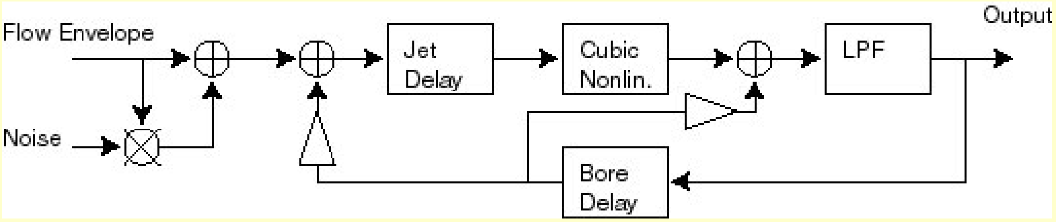


Figure 7.12. Cook's physical model of a slide-flute.

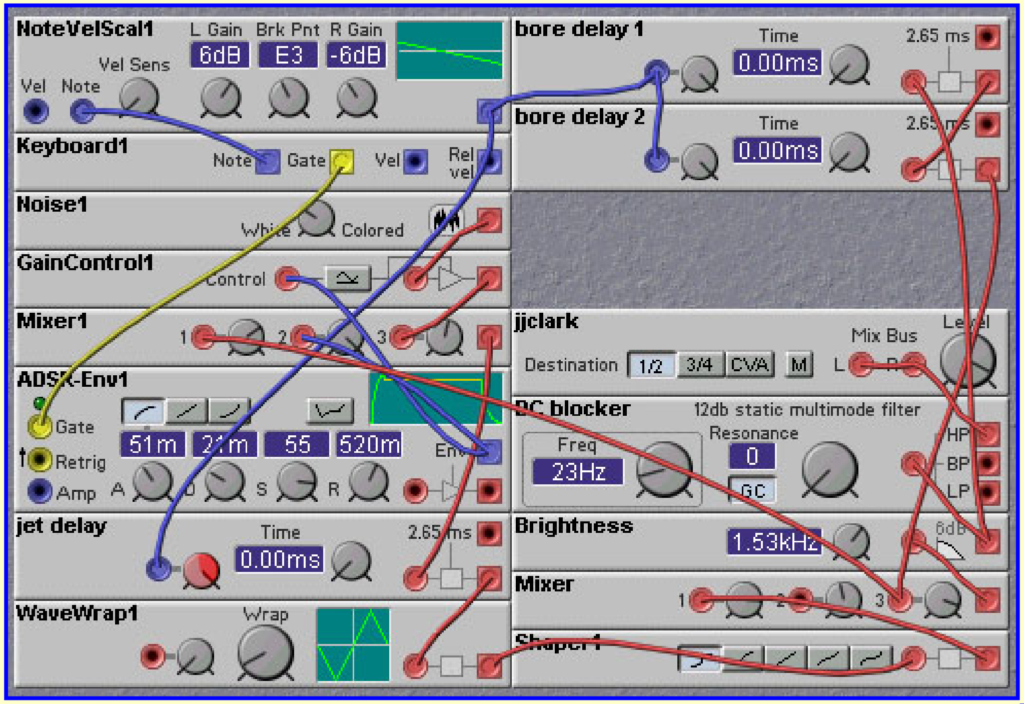


Figure 7.13. Nord Modular implementation of Cook's slide flute model (J. Clark).

All-Pass Filter Delays

There are other ways to obtain delays with the Nord Modular than the delay module. For example, delays in analog and digital signal processing systems are often obtained with *all-pass* filters. An all-pass filter is a filter that doesn't change the spectrum of the input, that is, the frequency response of the filter is flat. What is the use of this, you ask? Well, while the frequency response of the filter may be flat, its phase response is not. If the phase shift imposed by the filter changes linearly with frequency, the effect will be to cause a constant delay. We can implement an all-pass filter in the Nord Modular by combining a low-pass filter and a high-pass filter in parallel. In fact, we can use the multi-mode "D"-filter module and just sum its HP and LP outputs. The resulting phase is not linear, however. The delay is greatest near the cutoff frequency of the HP and LP filters. The frequency-dependent delay will cause 'dispersion', which may distort the sound if harmonics exist. For a pure sinusoidal waveform, there should be no distortion.

An example of this approach is given in the slide flute patch shown below. It is more difficult with this patch to ensure that the flute bore delay is twice that of the embouchure delay, so the sound is not as good as the previous slide flute patch, but it uses about 25% less DSP power.

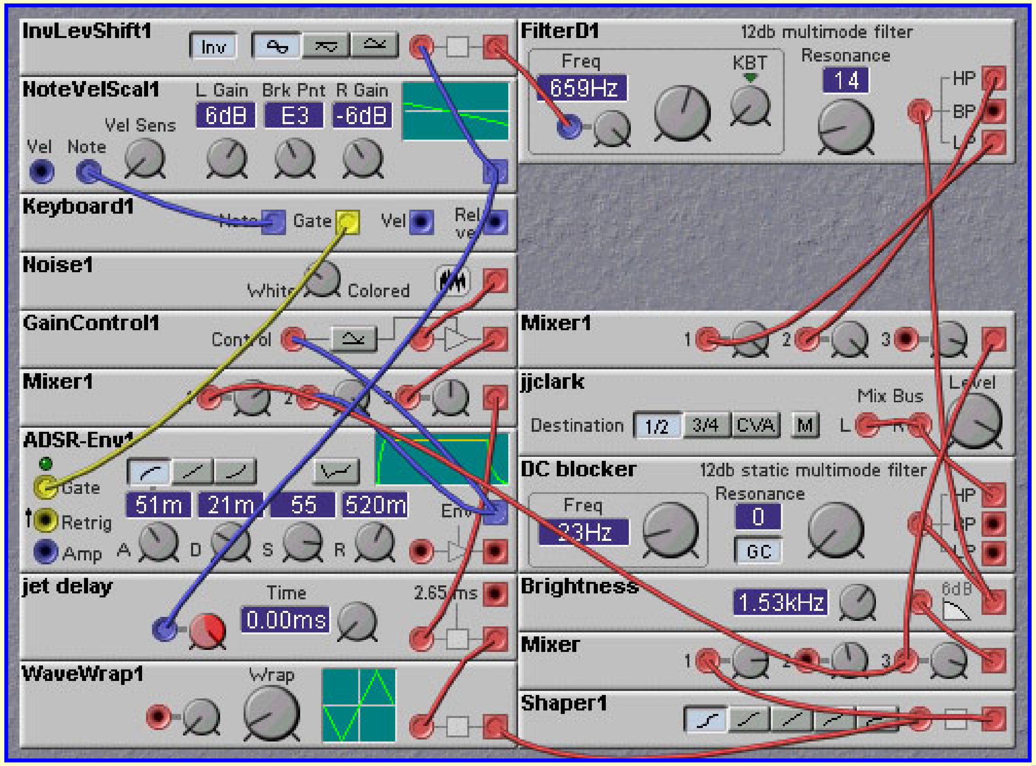


Figure 7.14. A slide flute model using all-pass delay lines (J. Clark).

7.8 Related Links

This chapter has just given the briefest of overviews of physical modelling of instruments. There are many, many, more details and issues to be considered, and many other instruments to be analyzed (such as brass instruments and percussion). If you are interested in implementing more complex physical models there is a wealth of literature on the topic. A good introduction to physical modeling can be found at harmony-central.com/Synth/Articles/Physical_Modeling.

For a thorough analysis of the subject of interpolation in delay lines, see the paper "Discrete-Time Modeling of Acoustic Tubes Using Fractional Delay Lines", by Vesa Välimäki, available at www.acoustics.hut.fi/~vpv/publications/vesa_phd.html.

Rob Hordijk has a good overview of the Nord Modular delay lines at www.clavia.se/nordmodular/Modularzone/DelayModule.html

Chet Singer (creator of many wonderful physical model patches for the Nord Modular) provides the following links:

1. http://www-ccrma.stanford.edu/software/clm/compmus/clm-tutorials/pm.html
This has examples of the Karplus-Strong algorithm and a flute. The flute is difficult to tune on the NM, because there are two different waveguides in it, and they must be precisely tuned one octave apart.

2. http://ccrma-www.stanford.edu/~jos/
This is Julius Smith's home page. He didn't invent physical modeling, but he's probably advanced it as far, or further, than anyone else.

3. http://www-ccrma.stanford.edu/~jos/waveguide/waveguide.html
This is a particularly good document within Julius Smith's home page. It's huge. Some of it gets kind of deep, but it also includes block diagrams of clarinets and violins.

4. http://www-ccrma.stanford.edu/software/stk/
If you're familiar with computer programming, this is Perry Cook's Synthesis ToolKit software package. It contains some examples of physical models written in C++.

5. http://windsynth.org/iwsa_labs/patch_programming/prog_techniques/VL1_Guide/
This is a VL programming guide. There's some interesting information in here, especially after reading some of Julius Smith's stuff, and some of the Yamaha patents.

6. http://www.delphion.com
This is a patent search site. Some relevant US patents are:

5,117,729: basic woodwind, brass, and a synthetic wind instrument model.
5,157,216: using pulsed noise for simulating bow scraping.
5,272,275: really complicated brass model.
5,286,914: multiple-model instrument. I think it's the VL1.
5,438,156: modeling a conical tube.
5,508,473: simulating period-synchronous noise in a wind instrument.
5,748,513: using coupled waveguides to create inharmonic sounds.
6,175,073: an improved bowed model.

These are all either Yamaha or Stanford patents. A list of the Stanford patents can be found at http://www.sondiusxg.com/patent.html.

7. M. E. McIntyre, R. T. Schumacher, and J. Woodhouse. "On the Oscillations of Musical Instruments". J. Acoust. Soc. Amer., Vol. 74, No. 5, pp. 1325-1345, 1983.
introduced the driver-and-waveguide idea. It describes a clarinet, a bowed string, and a flute.

8. http://www.sospubs.co.uk/sos/1997_articles/jul97/ronberry.html
This describes some work done by Ron Berry on implementing physical modeling on his modular analog synth.

Chapter 8. Speech Synthesis and Processing

Perhaps the most complex musical instrument of all is the human voice. Because of this, synthesists have been long trying to recreate the sound of the human voice. Of course, there are many non-musical applications of synthetic voice, and these have driven the development of much of the speech synthesis technology. In this chapter we examine some of the techniques that have been developed for doing speech synthesis and processing and see how these techniques can be implemented on the Nord Modular.

8.1 Vocoder Techniques

A channel vocoder is a device for compressing, or encoding, the data needed to represent a speech waveform, while still retaining the intelligibilty of the original waveform. The first channel vocoder was developed by Homer Dudley in 1936. It passed the speech signal through a bank of band-pass filters. These filters each covered a portion of the audio spectrum. The energy of each filter's output was then measured, or sampled, at regular time intervals and stored. This collection of filter energy samples then comprised the "coding" of the speech signal. This code could then be transmitted over a communication channel of lower bandwidth than would be neccessary for the raw speech signal. At the receiving end, the speech signal is reconstructed from this code by using the time sequence of filter energy samples to modulate the amplitude of a pulse signal being fed into a bank of filters similar to the ones used for the encoding. The result, while clearly not the original speech signal, is nonetheless intelligible, and one can, in most cases, understand what is being said.

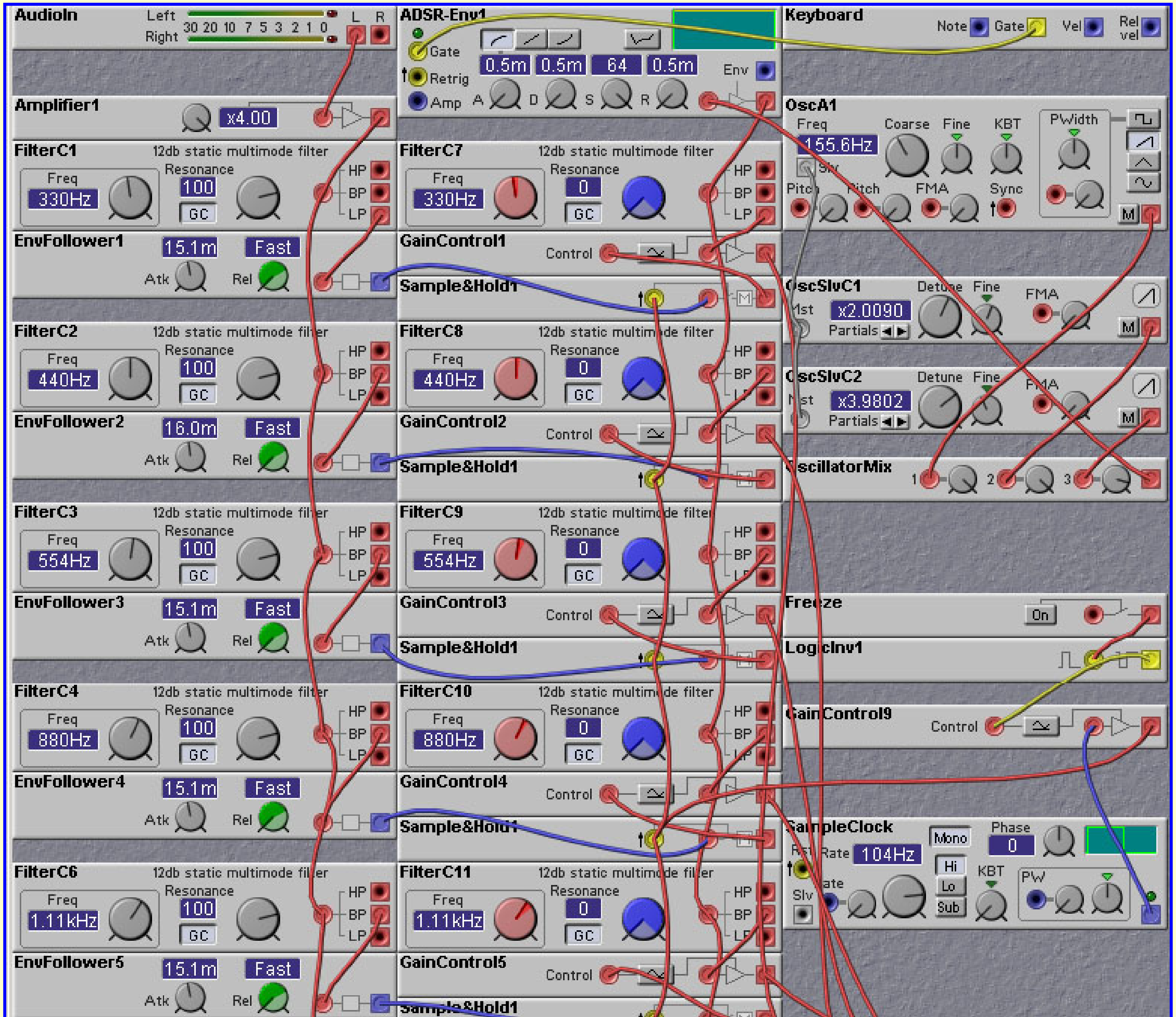
The idea for the channel vocoder technique arises from the manner in which speech is generated in the human vocal tract. Simply put, the vocal chords produce a periodic, pulse-like, stream of air, which is then acoustically filtered by the elements of the vocal tract: the esophagus, tongue, lips, teeth, and the oral and nasal cavities. As one speaks different sounds, the shape and elastic properties of these elements are being constantly changed in response to neural signals arising from speech centres in the brain. This causes a time-varying filtering, or spectral variation, of the excitation arising from the vocal chords.

The channel vocoder, then, first analyzes the speech signal to estimate this time-varying spectral variation. To do this it uses the filters in the filter banks to determine how a particular frequency component of the speech signal is changing with time. On the output end, this analysis of the spectral variations are used to synthesize the speech signal by using another filter bank to apply these same spectral variations to an artificial periodic pulse like signal. The output filter bank acts as an artificial vocal track and the pulse signal acts as a set of artificial vocal chords.

Some sounds produced during speech do not arise from the vocal chords, but are produced by turbulent air flow near constrictions in the vocal tract such as may occur between the tongue and the teeth. For example, such sounds as "SSS", "K", "SSHH", "P", and so forth arise in this manner. These sounds would be poorly reconstructed using a pulse excitation source, and so most channel vocoders also have a noise signal that can be used as an excitation source as well. A "Voiced/Unvoiced" detector circuit is used to detect whether the speech signal is arising from vocal chord excitation (Voiced speech) or is arising from noise excitation (Unvoiced speech), and the appropriate excitation source is then selected at the output end.

Channel vocoders were originally developed for signal coding purposes, with an eye (ear?) towards reducing the amount of data that would be needed to be transmitted over communication channels. In fact, speech coding system development continues to this day to be a vigorous area of research and development. These systems have far outstripped the basic channel vocoder idea in complexity, coding efficiency, and intelligibilty, however. So why do we still care about channel vocoders? The reason is that channel vocoders (and the functionally equivalent, but computationally quite different, phase vocoder) have found application to music production. In the 1960's Siemens in Germany produced a vocoder which was used in some recordings. The BBC Radiophonic Workshop in England likewise pioneered the use of vocoders in recording and in radio and television. The vocoders used in these early musical efforts were very large and unsuited to general use. In the mid-70's a breakthrough of sorts came about when a number of companies, notably EMS (Electronic Music Studios) in England, produced relatively small and easy to use vocoders designed for use in musical applications. After that, the vocoder sound became a staple of the music and entertainment industry. Many extremely popular records (Kraftwerk!), TV shows (the Cylons of Battlestar-Galactica), and movies (Darth Vader in Star Wars) are identfied with vocoders. Although the introduction of these relatively small vocoding systems made it possible for the wide application of vocoders to music and film, they were still quite expensive for your average musician in the street. The EMS vocoder cost upwards of 6500 UK pounds! There are now, however, a number of inexpensive hardware vocoders available, as well as a few software emulators (including the world-famous Cylonix vocoder, developed by the yours truly).

The Nord Modular is also capable of implementing very nice vocoders. The following patch, developed by Kees van de Maarel, shows how to make an 8-band channel vocoder from the Nord Modular static filter modules. The quality of this vocoder implementation is nowhere near that of systems like the EMS vocoder, due to the relatively small number of bands and the slow cutoff rate (12dB) of the filters used. But the sound is still quite nice, and the patch illustrates the idea. It has a number of features not found on many cheaper vocoders - sampling and holding of the spectral features, and control of the speed of the envelope detectors. By slowing down the envelope detectors one can obtain a "slewing" effect where phonemes appear to get blurred together. This can produce a smoother sound.



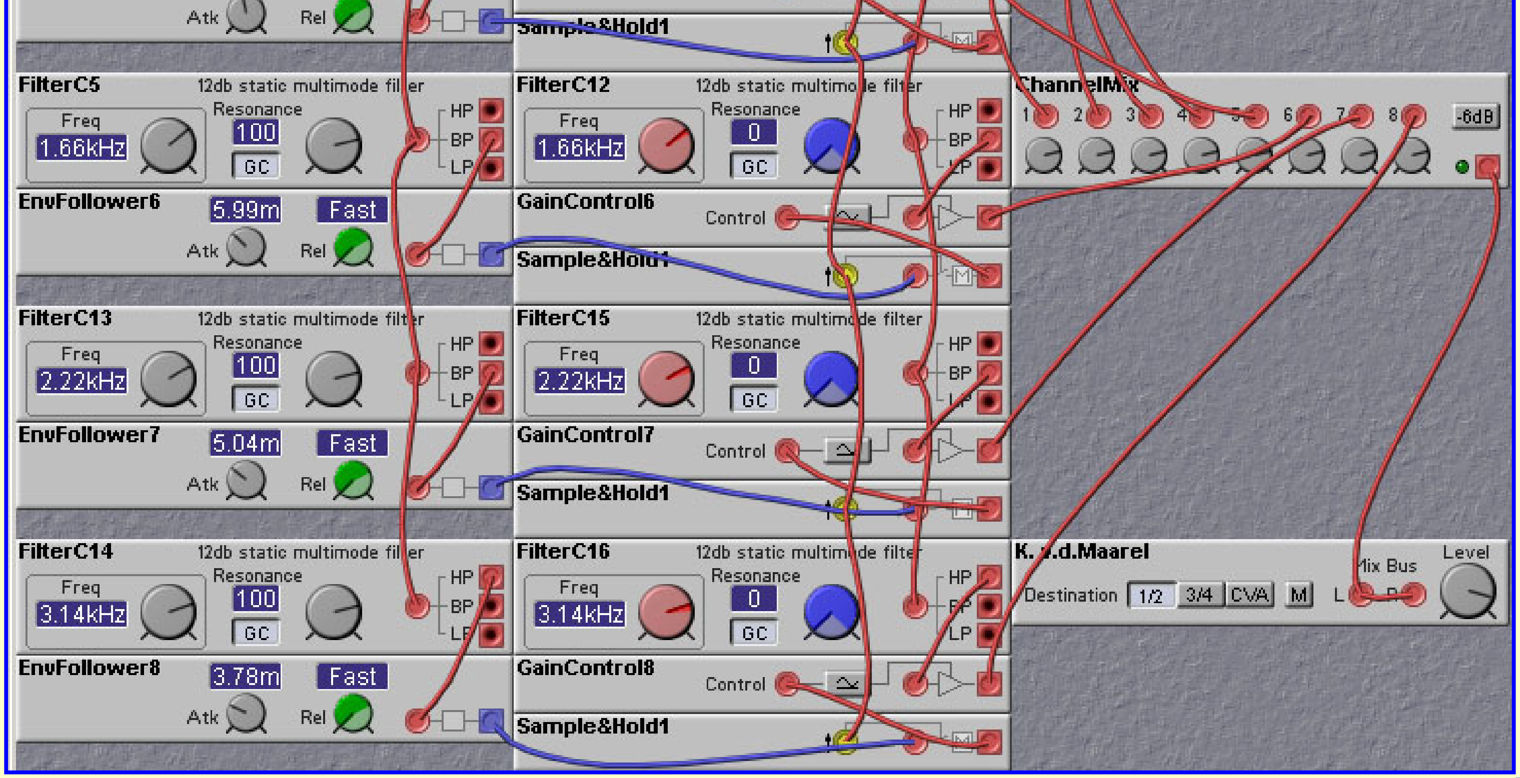


Figure 8.1. A vocoder made from bandpass filter modules. Includes features such as spectrum sample/hold and freeze (K. v.d. Maarel).

Of course, the Nord Modular also provides a self-contained vocoder module, so one does not have to go through all the trouble of constructing a vocoder patch from scratch, as in the patch shown earlier. The Vocoder module has a limited number of features, however, so there may still be some need to construct a custom vocoder patch. Some features can be added to the Vocoder module, however. For example, we show in the patch below how to add a Voiced/Unvoiced detection feature. In this patch the input signal is passed through two filters - a lowpass and a highpass, to estimate the frequency content of the input signal. If the lowpass filter output has an amplitude greater than the highpass filter output, then the input signal is mostly likely a pitched signal, such as a vowel sound. If, on the other hand, the highpass filter output has a higher amplitude then the input signal is most likely an *unvoiced* sound, such as a consonant. To do the decision as to the type of input signal we pass the filter outputs through envelope detectors (to estimate the amplitude of the filter outputs) and then compare the envelope detector output values. We use the output of the comparator module to switch, with a cross-fade module, between two signal sources - a mixed set of three square waves and a noise waveform.

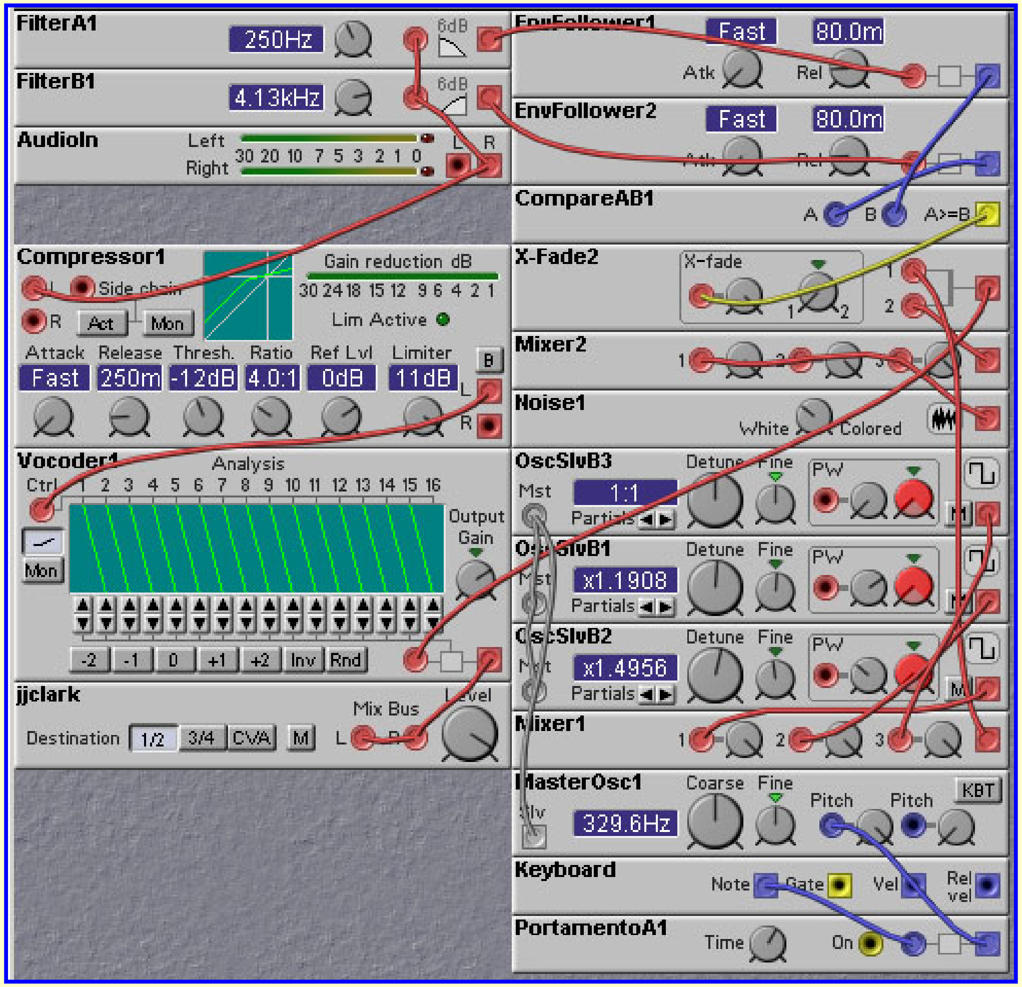


Figure 8.2. A vocoder patch made from the Nord Modular vocoder module. Includes a voiced/unvoiced decision process to distinguish between vowels and plosives (J. Clark).

Note the use of a compressor in the above patch. This is a common technique used with vocoders, as it prevents the sound from becoming too "choppy" due to the relatively low amplitude of vocal sounds at the beginning and end of spoken words.

The Nord Modular vocoder module allows the remapping of analysis channels to synthesis channels. This warping of the frequency content of a sound can be used in many ways; one of the most interesting being its effect on percussive sounds (try it!). It can also be used to make a male voice sound more feminine as crossing over some of the higher frequency bands can simulate the differences in the resonant structures of the male and female vocal tract. This is demonstrated in the following patch by Tommi Lindell.

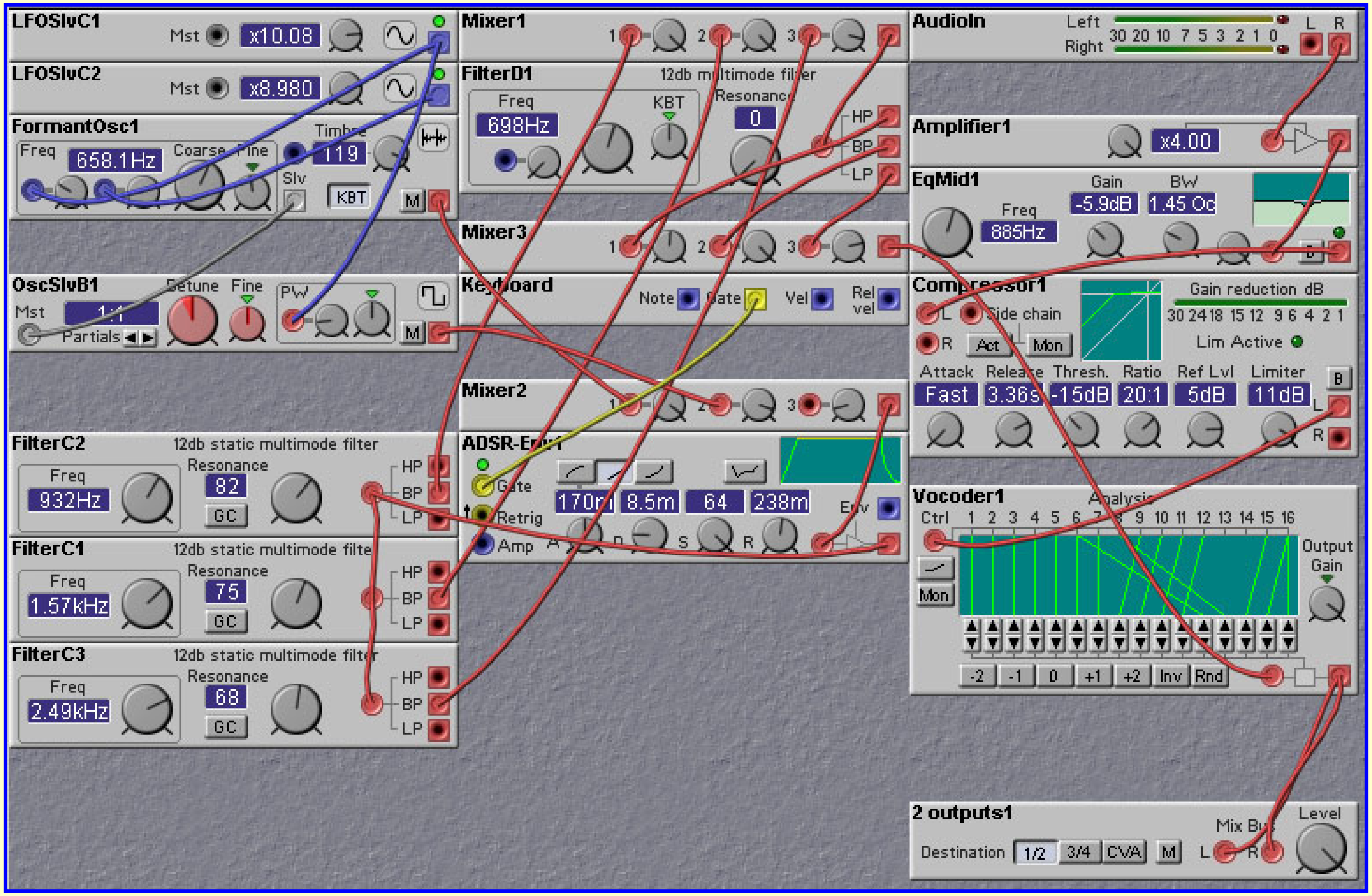


Figure 8.3. A patch that emulates female voices, with vibrato (T. Lindell).

8.2 Speech Synthesis

One of the most popular uses of the vocal filter is to make the Nord Modular 'talk'. Speech for the most part consists of sequences of phonemes. Phonemes are the sonic building blocks of speech and are short segments of sound.

There are four main forms of phonemes - vowels, diphthongs, semivowels, and consonants. We will examine the synthesis of each of these types of phonemes.

Vowel Synthesis

The vowel sounds are pitched and have a definite spectral structure. The vocal filter module is designed to produce the spectrum of vowel sounds. Vowel sounds are created by passing a periodic pulse waveform (which sets the basic pitch of the voice) through a complex filter with multiple resonances. The pulse waveform models the vibration of the vocal cords. In the patches below we use a narrow pulse wave, but half-wave rectified sine-waves and sawtooth waves could also be used. The complex filter models the effect of the vocal cavity, formed by the mouth, throat, and nasal passages. Because of the complicated shape of these passages, some frequencies are enhanced while others are diminished in strength.

This results in resonances and anti-resonances. The resonances are known as *formants*. There are usually only three formants with a significant presence in vowel sounds. The first two formants play the biggest role in distinguishing one vowel sound from another. The center frequencies and bandwidths of these formants depend on the shape of the vocal tract. People can change this shape quite dramatically, mainly through changing the position of the tongue, resulting in large shifts of the formant frequencies. It is this shape-dependent shift of the formants that gives rise to the different vowel sounds. For some sounds, such as the vowel /a/ (as in "father"), the vocal tract is open at the front and constricted at the back by the tongue. In this case the frequency of the first formant is rather high and the frequency of the second formant is quite low. On the other hand, for sounds such as the vowel /i/ (as in "eve"), the tongue is raised towards the palate, constricting the front of the vocal tract while opening up the back. In these vowels, the first formant frequency is rather low, while the second formant frequency is high. Sounds like /uh/ (as in "but") are somewhere in the middle, with a relatively open vocal tract throughout.

The following patch is a demonstration of vowel sound creation. It uses three bandpass filters to form the formants. Control sequencers are used to adjust the formant frequencies (the sequencers are stepped on each keypress). Five different vowels are implemented -
ee - as in "beet"
eh - as in "bet"
ah - as in "bought"
oo - as in "boot"
uh - as in "but"

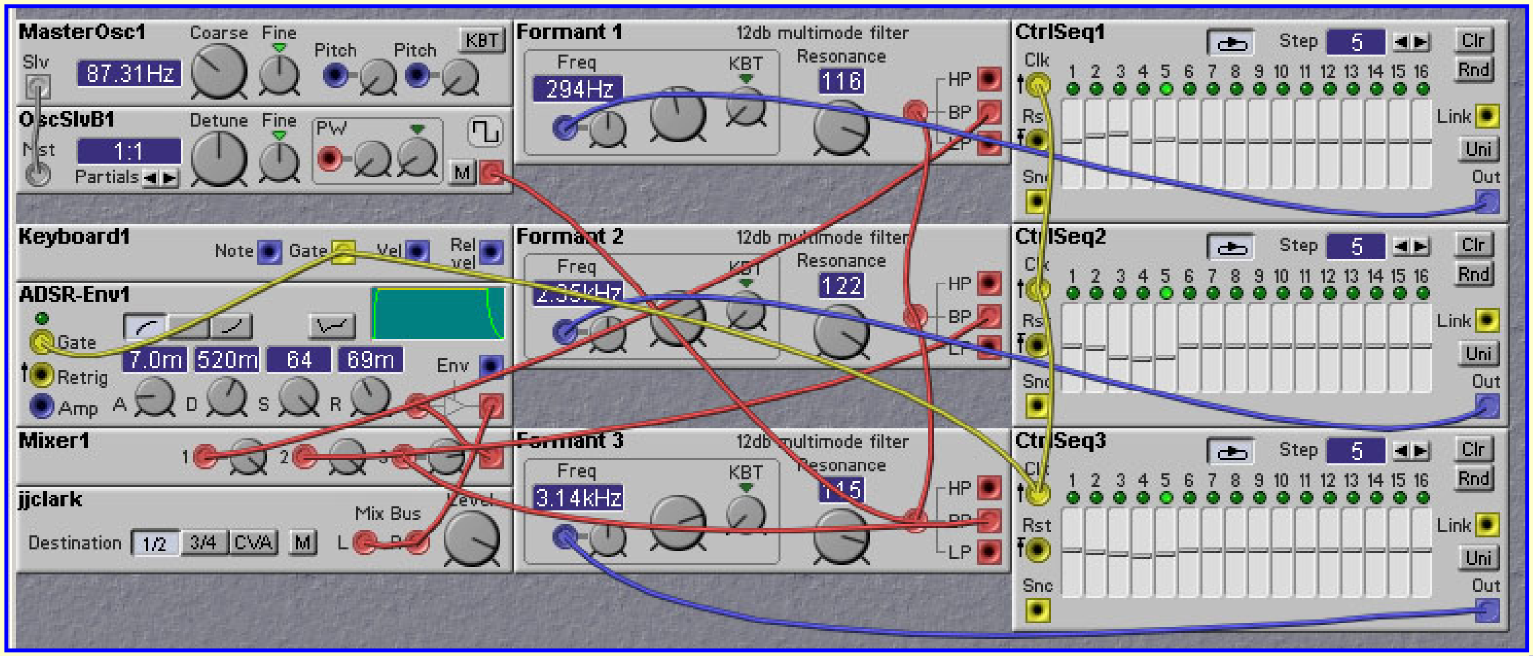


Figure 8.4. A patch illustrating the generation of five different vowel sounds (J. Clark).

In the above patch the bandwidths of the bandpass filters were held constant. In reality, the bandwidths of the formants will change somewhat from vowel to vowel. If we wanted to go to that level of detail in our model, we could use a filter with a voltage controllable resonance level.

Semi-Vowel and Diphthong Synthesis

Some sounds are made by a rapid transition between two different vowels. Such sounds are called *diphthongs*. An example is shown in the following patch which generates a "aye" sound by sliding from the /a/ (as in "hot") vowel to the /i/ vowel. This is mainly implemented as a rise in the frequency of the second formant, and a slight drop in the first formant frequency.

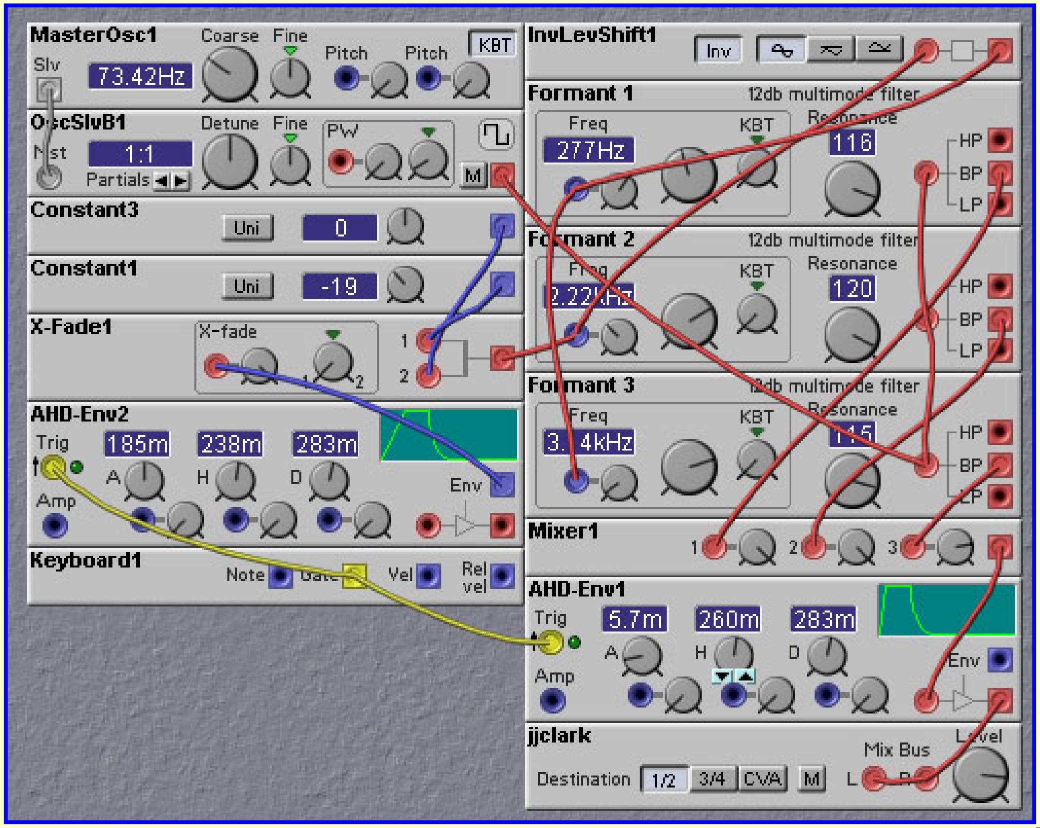


Figure 8.5. A patch that generates the diphthong "aye", by sliding between two vowel sounds (J. Clark).

Note the use of the AHD envelope type in the patch above. These are convenient for speech synthesis for two reasons - they allow holding of the vowel for a certain length of time, and they have a trigger input rather than a gate input, which allows for repeatable timing when using a keyboard key press to begin the sound.

Related to diphthongs are so-called semivowels, such as /w/, /l/, /r/ and /y/. These are also created by a slide between two different phonemes as with diphthongs. The rate of the glide is somewhat slower than with the diphthongs. They can be difficult to synthesize, as they are often not even explicitly generated, but instead are inferred to exist between succesive two phonemes.

The following patch demonstrates the synthesis of a /w/ sound. This sound is not synthesized on its own, but instead arises when a glide between phonemes is created, starting from a /u/ sound. In the patch we follow the /w/ sound with a /ai/ diphthong to form a 3-way glide from /u/ to /a/ to /i/. The first glide creates the /w/ sound and the second glide creates the final vowel sound. The overall result is the sound "why". In general, any slide from /u/ to some other vowel will create a /w/ sound.

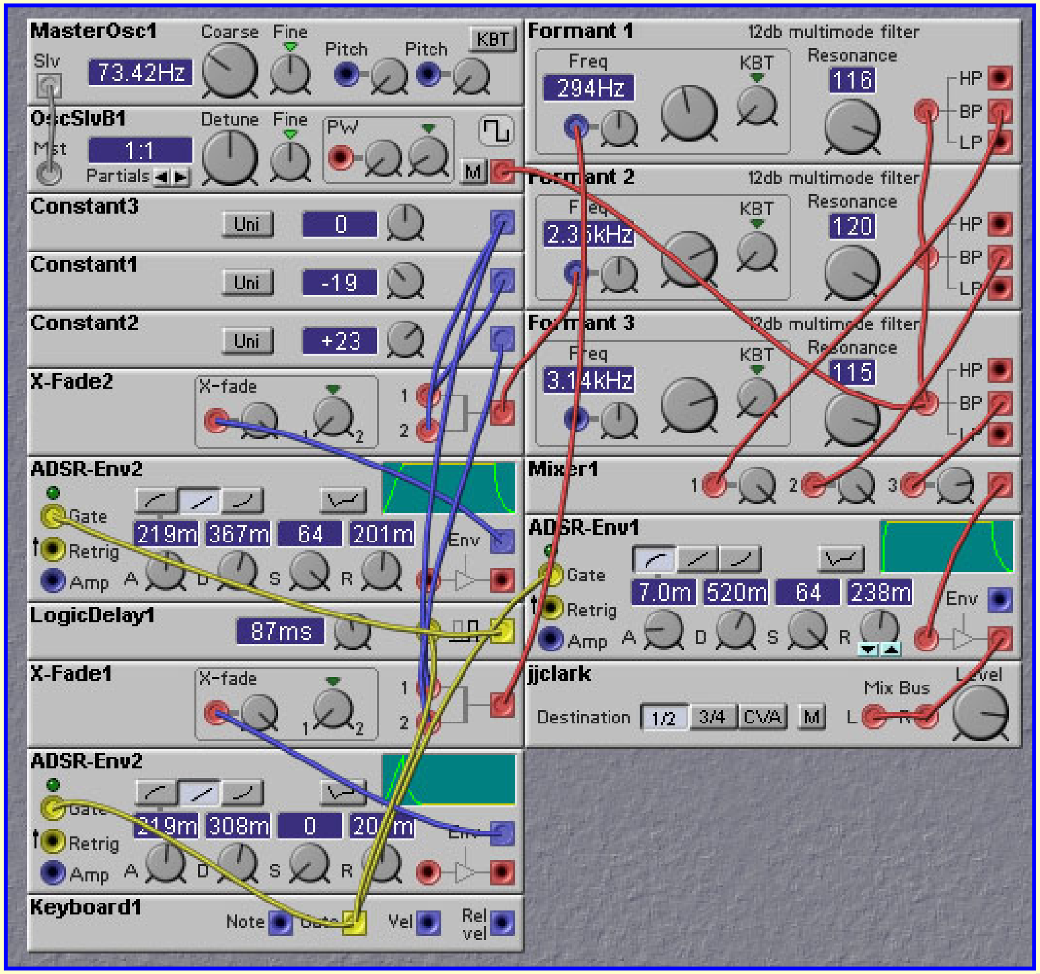


Figure 8.6. A patch that asks 'why' (J. Clark).

The next patch shows how to synthesize a "y" sound. Again, it is created by doing a slide from one vowel to another, in this case from the /i/ vowel to the /ae/ (as in "bat") vowel. In general, as slide from an /i/ vowel to any other will be heard as beginning with a /y/. This is almost the opposite of the "Aye" patch seen above.

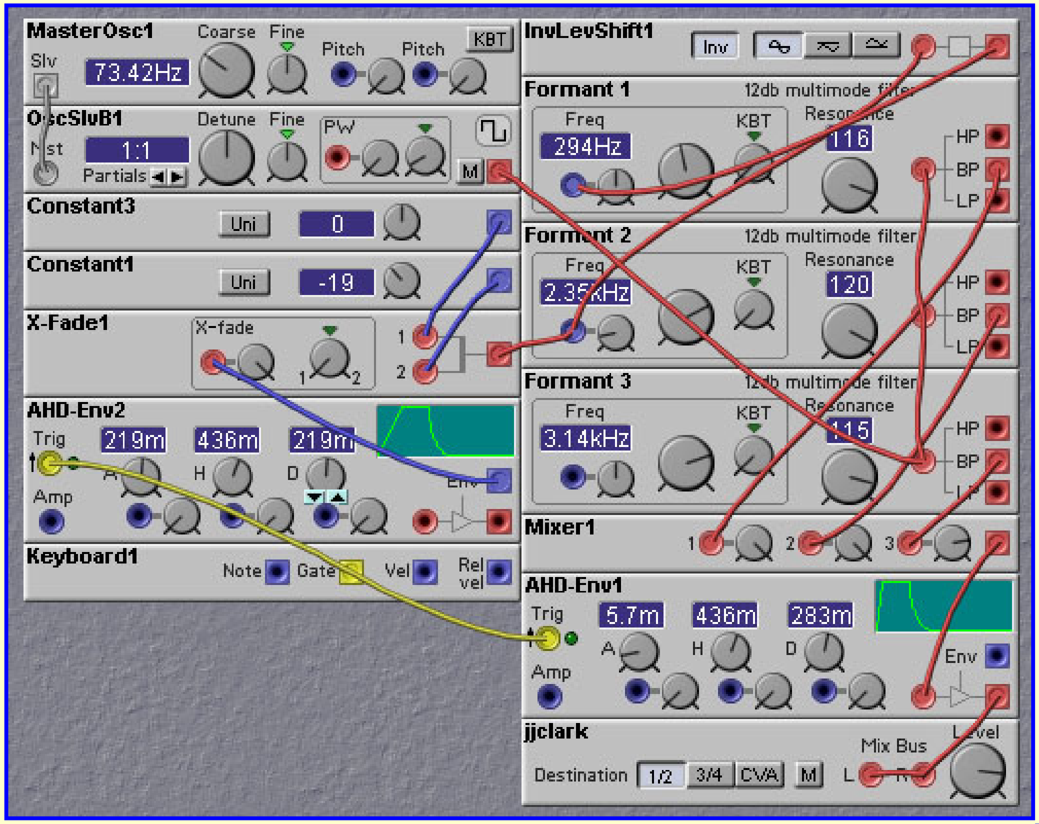


Figure 8.7. A patch that says 'yeah' (J. Clark).

The semivowel "r" can be made by sliding relatively slowly from the vowel /ir/ (as in "bird") to some other vowel. The /ir/ sound has a lowering of the 3rd formant frequency. This is due to the retroflexion (rounding) of the tongue. This is demonstrated in the following patch.

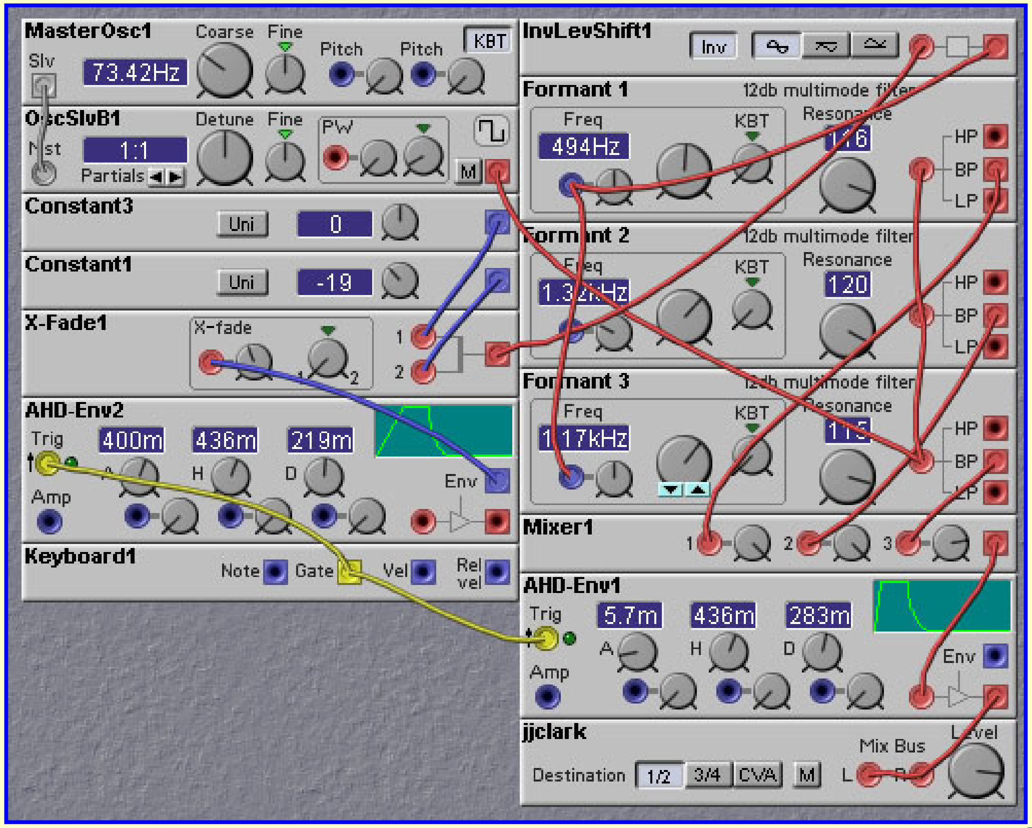


Figure 8.8. A patch that goes 'rah' (J. Clark).

Consonant Synthesis

Consonants are created when the vocal tract is constricted in some way, usually with the tongue, teeth or lips. There are five types of consonants employed in European languages - nasals, stops (voiced and unvoiced), fricatives (voiced and unvoiced), whisper, and affricates.

Synthesis of Stops

Stops are transient sounds created by the buildup and release of pressure behind a complete constriction somewhere in the vocal tract. There are three main types of stops, defined according to the location of the constriction of the vocal tract. In the alveolar stops /t/ and /d/, the constriction is formed by the tongue resting against the alveolar ridge behind the teeth. In the labial stops /p/ and /b/, the constriction is formed by the lips coming together. In velar stops /k/ and /g/ the constriction is created by pushing the back of the tongue against the soft palate (velum).

There are two forms of stops - voiced and unvoiced. These are distinguished by the presence (in unvoiced stops) or absence of an unvoiced excitation phase. In the unvoiced excitation phase, the vocal chords do not vibrate. Pressure builds up behind the closure site, and when the pressure is released there is a brief period of friction followed by a period of aspiration (steady stream of air passing through the vocal tract, exciting its resonances). In the voiced stops, the unvoiced excitation phase does not take place, and there is only a brief silent period.

Similar to the semivowels (/r/, /w/, /y/) the different voiced stops (/d/, /g/, /b/) are distinguished mainly by different shifts in the first two formant frequencies from a fixed locus to the frequencies associated with the following vowel. This shift in formant frequencies is caused by the changing configuration of the vocal tract from the initial closed form to the final vowel form. For /b/ sounds the second formant is initially in the range 600-800 Hz. For /g/ sounds F2 is initially around 1300 Hz, and is around 2KHz for /d/ sounds. In all stops the first formant frequency starts out very low and rises rapidly to that of the following vowel sound. There is also a brief percussive transient which has a broad spectrum falling off slightly with frequency. A brief silent period before the onset also emphasizes the consonant. The formant transition begins during the silent period.

The following patch illustrates the synthesis of the voiced stops /d/ and /b/. The patch says "daddy" and "babby", selected by knob 1.

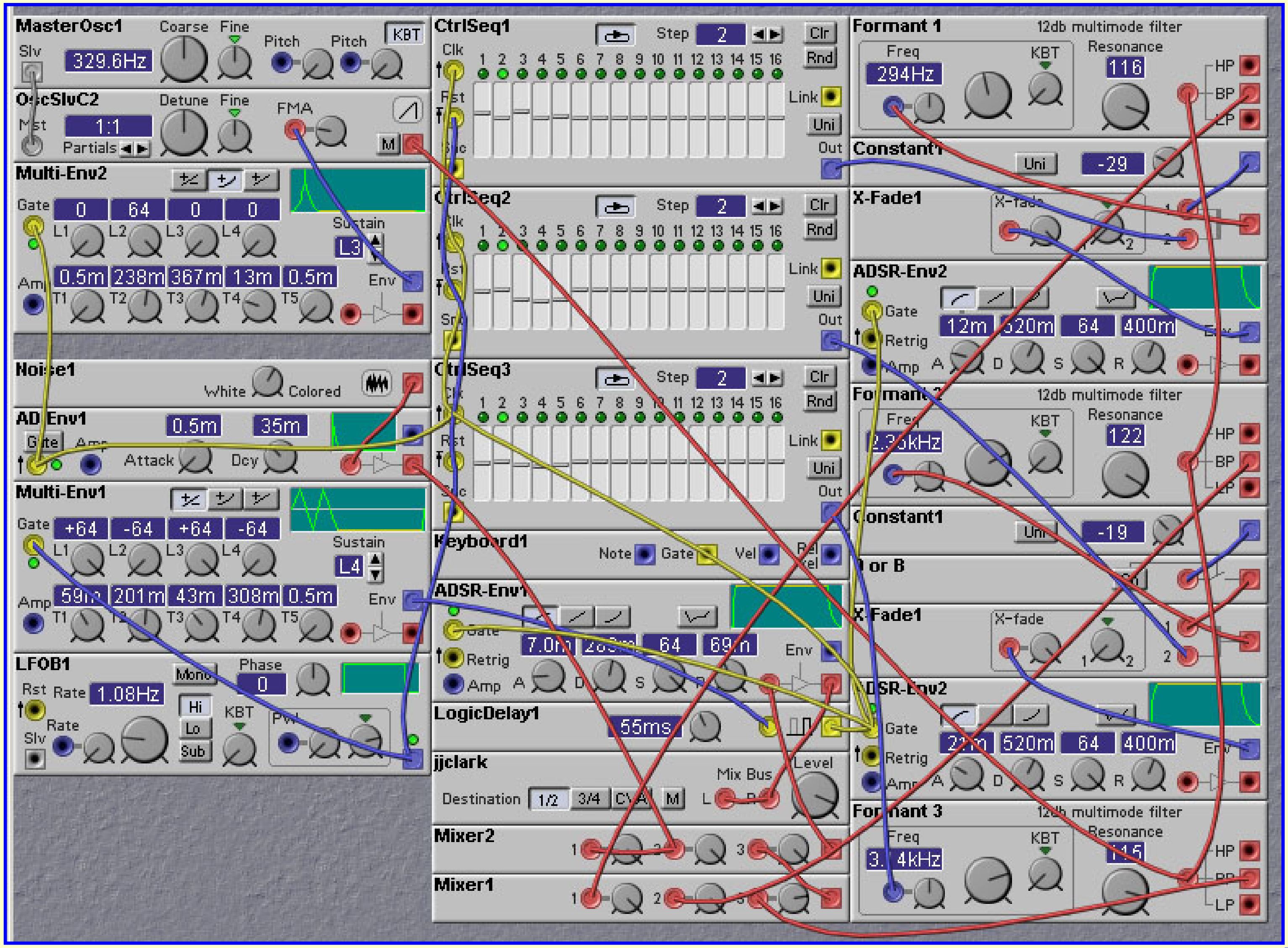


Figure 8.9. A patch demonstrating the synthesis of the voiced stops 'd' and 'b' (J. Clark).

As mentioned above, unvoiced stops differ from voiced stops in that they have an unvoiced excitation phase consisting of a noise burst (due to the turbulent release of air past the constriction) and a brief period of aspiration(a sort of whispering sound). The nature of the different sounds obtained by having the constriction at different places is mainly manifest in the spectrum of the noise burst and aspiration. For /p/ sounds, made by a closure at the lips, the spectrum is concentrated at low frequencies. For /t/ sounds the spectrum is concentrated at high frequencies. For /k/ sounds, the spectrum has a shape somewhere between the /b/ and /t/ cases. The aspiration sound is created by passing a noise waveform with the same spectrum of the noise burst through the vocal tract filter. Thus, there will be formants in the aspiration spectrum, just as in the voiced excitation case. There is also the shift in formant frequencies to the following vowel as in the case of voiced excitation.

The following patch shows how unvoiced stops (as well as voiced stops) can be synthesized. Knob 1 selects the particular stop to be synthesized, and selects the appropriate noise spectrum and formant transition. Knob 2 switches between voiced and unvoiced synthesis, by gating in and out the unvoiced excitation.

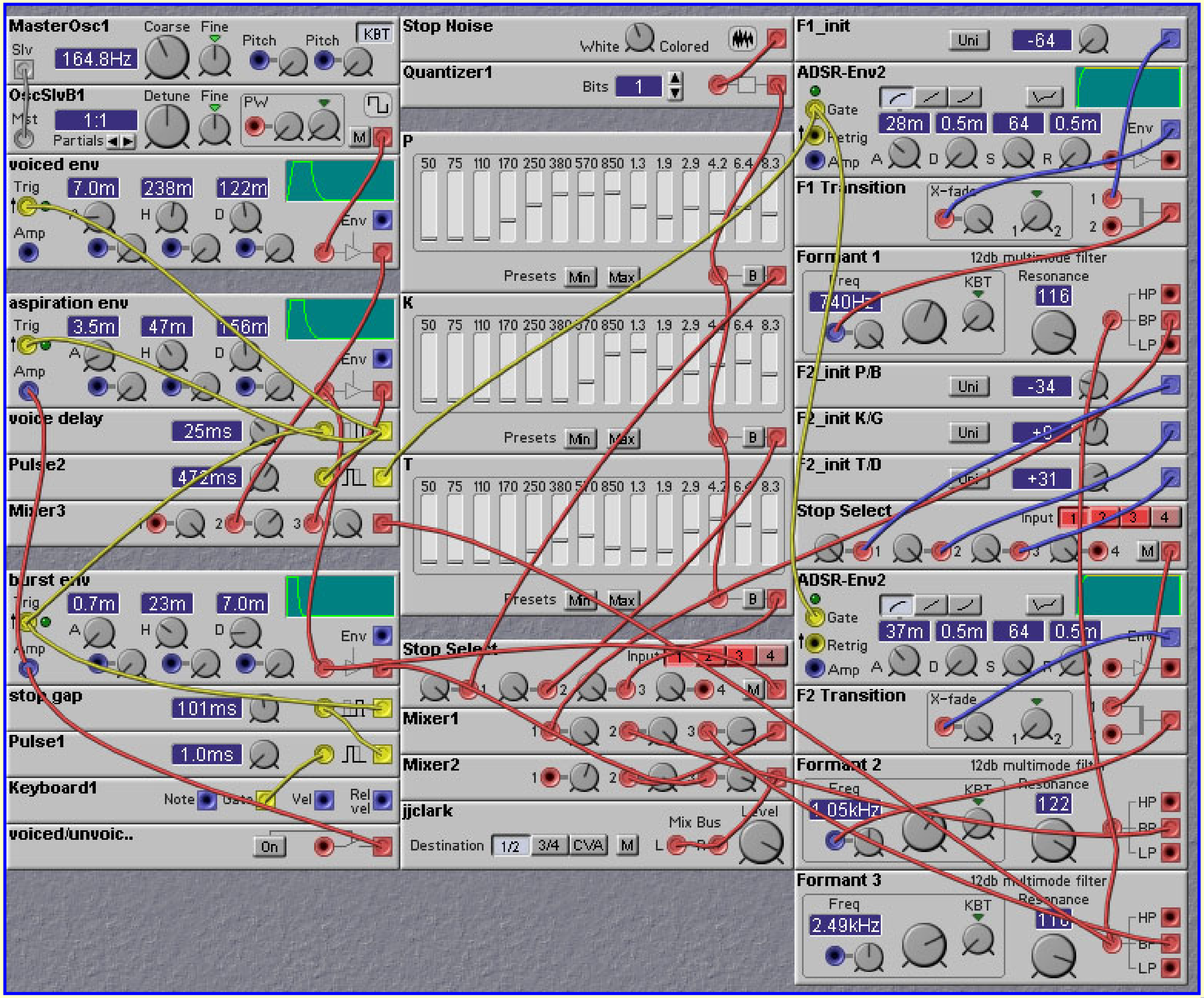


Figure 8.10. A patch demonstrating the synthesis of voiced and unvoiced stops (J. Clark).

Fricative Synthesis

There are two types of fricatives, voiced and unvoiced. In unvoiced fricatives, the vocal tract is excited by a steady flow of air which becomes turbulent (noisy) in a constricted region of the vocal tract. The location of the constriction determines the nature of the resulting sound. For the fricative /f/ the constriction is at the lips, for /th/ sounds the constriction is at the teeth, for /s/ the constriction is near the middle of the oral cavity, and for /sh/ sounds the constriction is at the back of the oral cavity. The constriction divides the vocal tract into two sections. The sound radiates from the front part, and the back part traps energy and creates "anti-resonances" or dips in the overall spectrum of the sound. Fricatives that are made with the constriction near the front of the vocal tract have a broad spectrum, while those with the constriction near the rear have a more narrowly distributed spectrum.

The patch shown in the following figure shows how unvoiced fricatives can be implemented on the Nord Modular. Filter banks are used to sculpture the required spectral characteristics which distinguish the various types of fricatives.

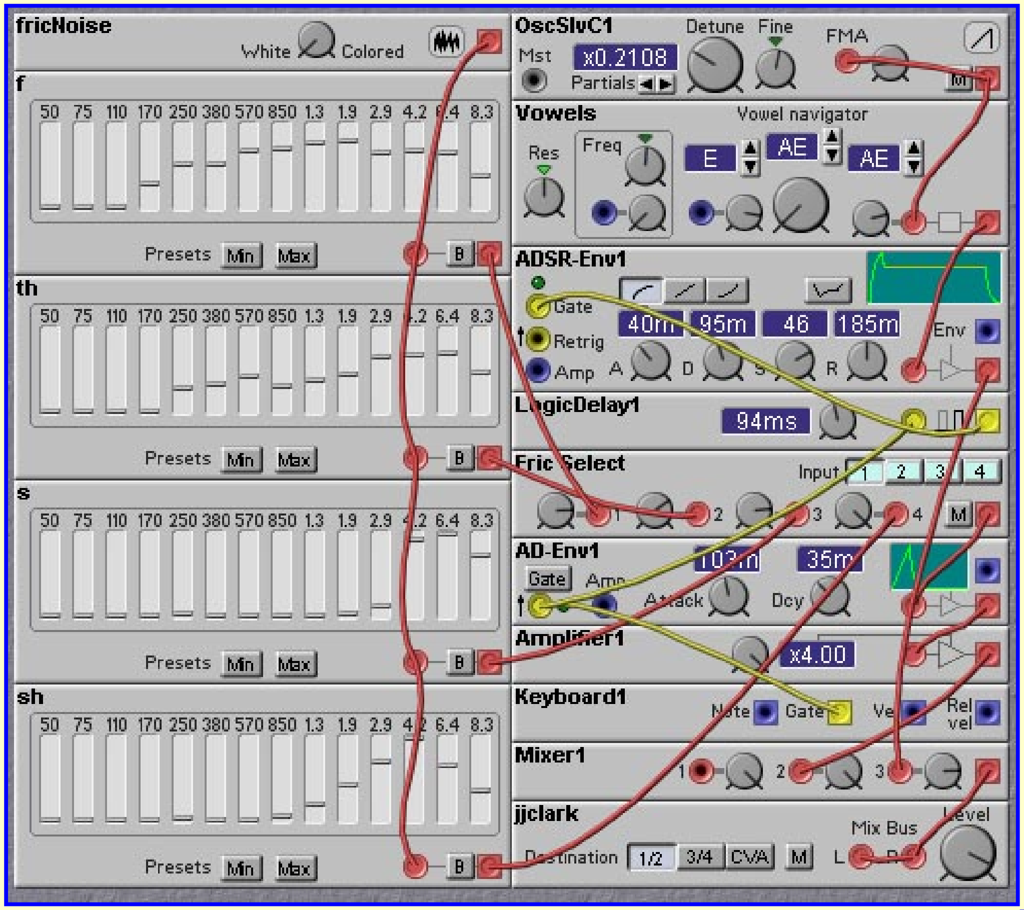


Figure 8.11. Implementation of unvoiced fricatives. The patch generates the sounds "Fee", "Thee" (unvoiced, like a lisped version of "See"), "See", and "Shee". The spectrum for each fricative is set by the filter banks.

Voiced fricatives are similar to the unvoiced ones, except that the vocal chords vibrate, causing a period pulsing of the air passing through the turbulence inducing constriction. The vocal chord vibration reduces the airflow, which lowers the amount of turbulence relative to the unvoiced fricatives. The patch shown in the next figure illustrates how we can implement voiced fricatives on the Nord Modular. This patch is essentially the same as the patch for generation of unvoiced fricatives. The major change is that the output of noise source modelling the turbulence is amplitude modulated by the oscillator modelling the vocal chord vibrations. This amplitude modulation simulates the pulsing of air through the turbulent constriction of the vocal tract.

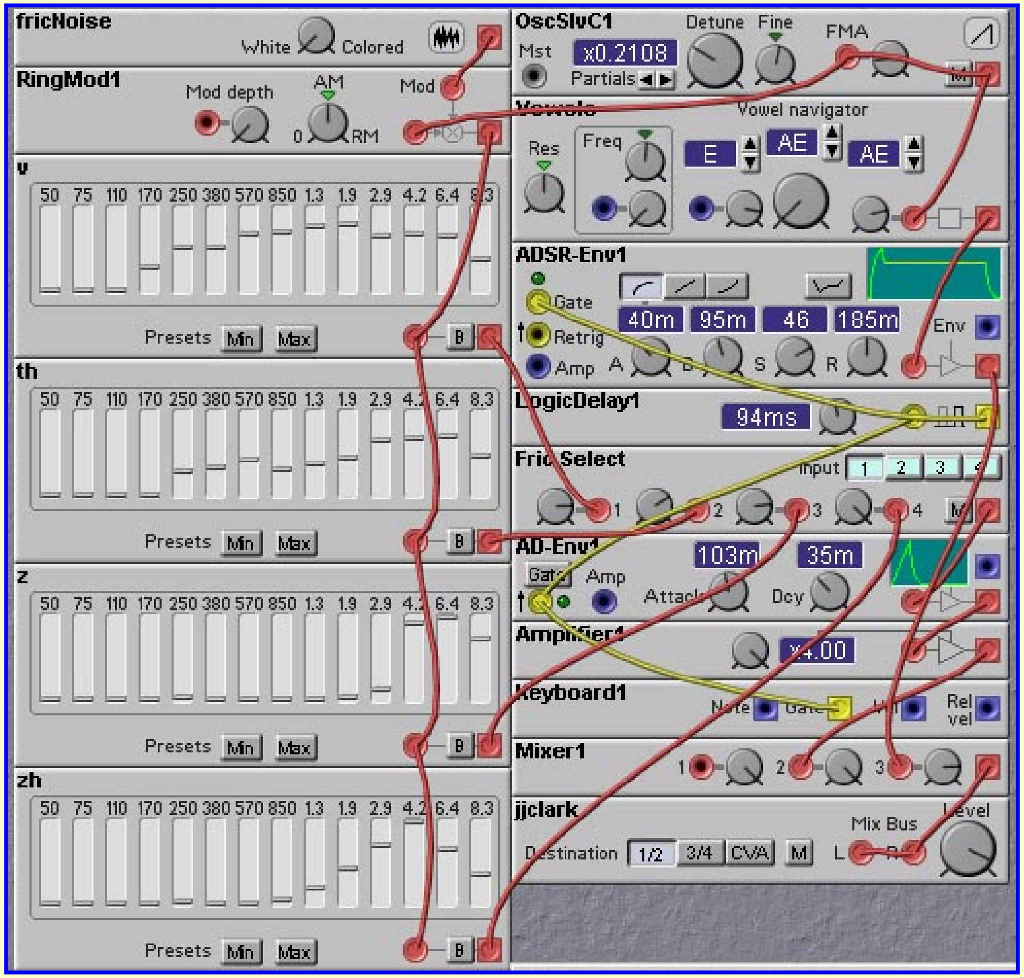


Figure 8.12. Implementation of voiced fricatives. The patch generates the sounds "Vee", "Thee" (voiced), "Zee", and "Zhee". The spectrum for each fricative is set by the filter banks.

Putting it All Together

Now you have most of what you need to make the Nord Modular say what you want (you might also want to find some tables of formant frequencies for the various vowels, these can be found on the net or in the library). You should tweak the formant frequencies, bandwidths, and envelope time constants to obtain just the sound you are looking for.

The following patches (by Kees van der Maarel) give examples of implementing various long utterances with the Nord Modular, combining vowels, semivowels, and consonants.

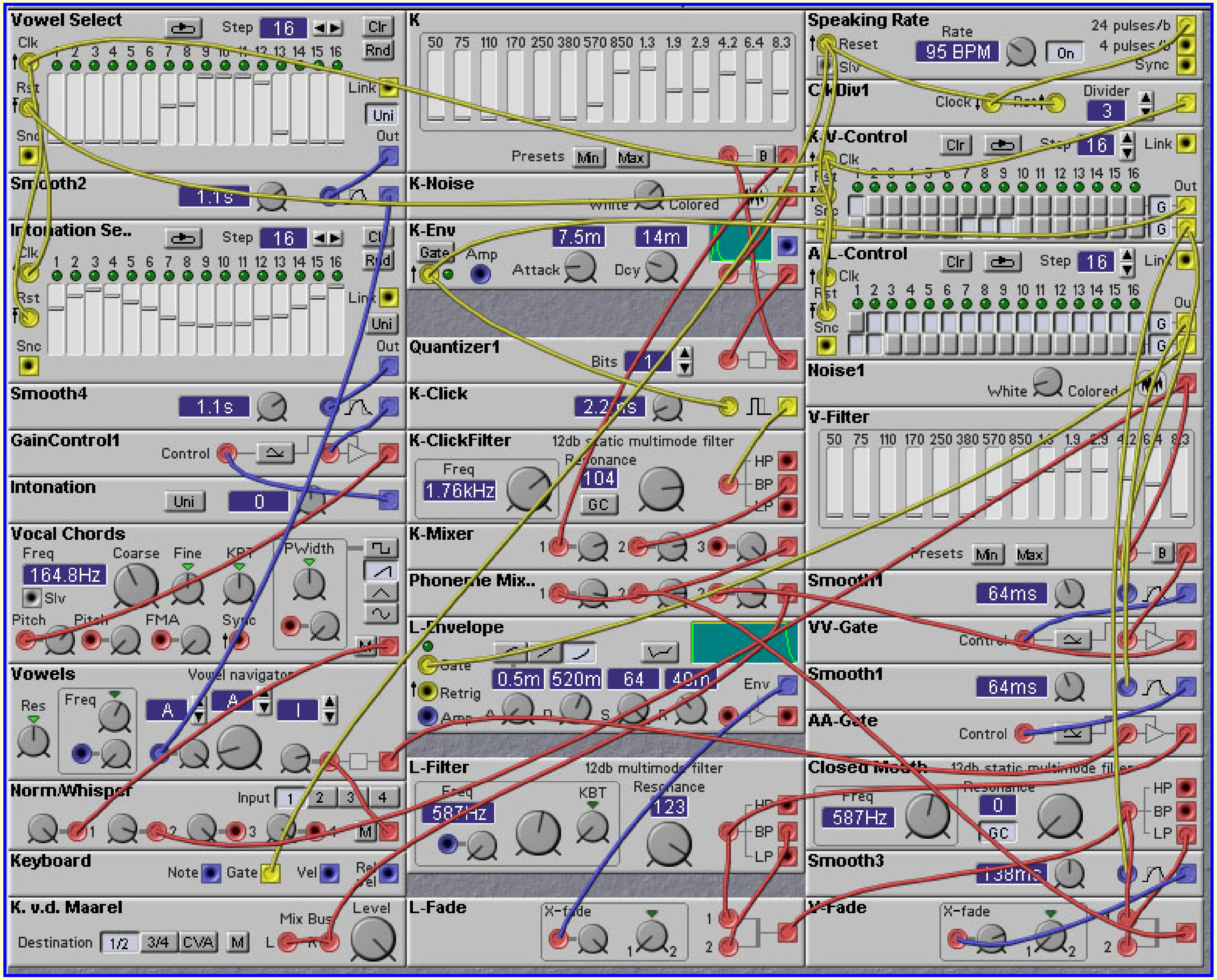


Figure 8.13. A talking patch. This one says 'Clavia' (K. v.d. Maarel).

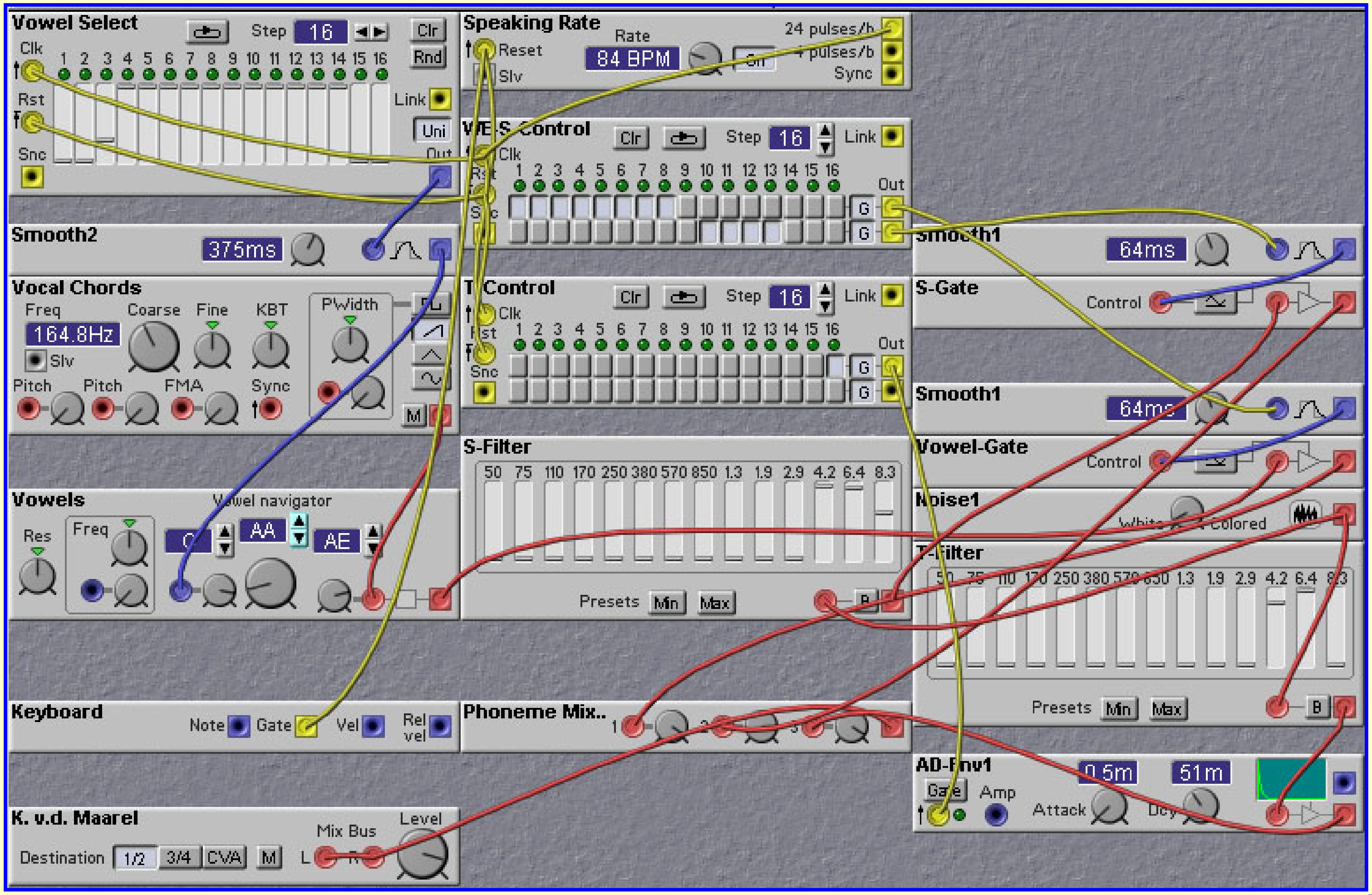


Figure 8.14 Another talking patch. This one says 'West' (K. v.d. Maarel).

8.3 Pitch Tracking

Musicians often want to control a synthesizer with another instrument - one that they have more skill in playing. For example, guitarists might want to control a synthesizer through playing their guitar, or a vocalist through means of their voice. The most important pieces of information that a performer would like to convey to the synthesizer are the pitch and amplitude of the sounds that are to be generated. The amplitude of the sound is usually measured through the use of an *Envelope Generator*. This is a circuit that estimates the RMS (root-mean-square) level of a signal over some time interval. The Nord Modular provides such a module. There is no module provided, however, for measuring the pitch of an input signal. Part of the reason for this is probably that it is very difficult to determine the pitch of a signal in an accurate and noise free manner. But there are some simple teachniques that can be applied with some success. We will look at a few of these techniques here.

The first approach we will look at involves synchonization of an oscillator to the input signal. This uses oscillators whose waveform cycle can be reset by an external signal. Review the section on oscillator sync for more details. The following patch demonstrates this approach. In the patch an oscillator is synced to the zero crossings of an input signal. If the input signal is a relatively pure tone, the rate of its zero crossings will be more or less the same as its fundamental frequency. Hence the synced oscillator will track the pitch of the input.

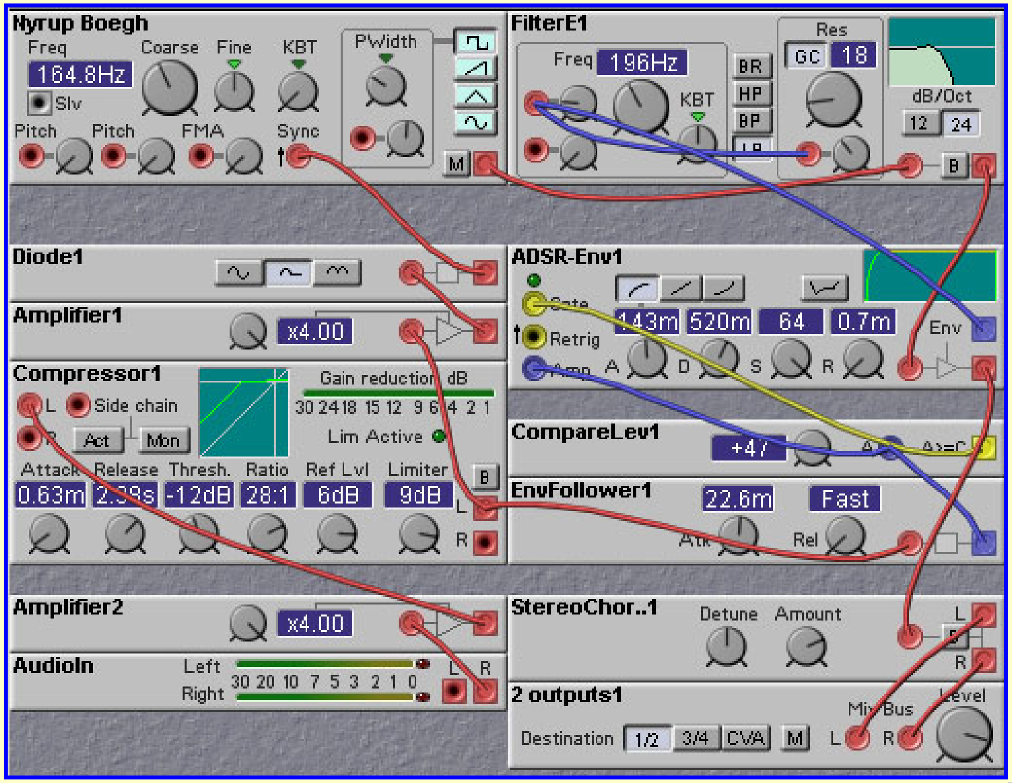


Figure 8.15. Another pitch follower patch. This uses an oscillator synced to the zero crossings of the input signal (N. Beogh).

Rather than directly synchronizing an oscillator to the zero crossings of an input signal, we can instead measure the zero crossing rate of the input signal and use this rate measure to adjust the pitch of a slave oscillator. The following patch demonstrates this approach. A lowpass filter removes most of the energy of the input signal above 262 Hz. This is done to emphasize the fundamental frequency component, so that the zero crossings occur at the same rate as the fundamental. For most male speakers (and many female), the pitch of the voiced speech signals is less than 250 Hz. At each zero crossing a pulse of fixed width (1.3msec) is generated. These pulses are then smoothed out, or averaged, with a smoothing filter. This produces a control value which is proportional to the zero crossing rate. The control value is then used to vary the pitch of an oscillator, through the grey MST oscillator inputs (which provide linear frequency control).

One of the most common uses for a pitch tracker is to tune the carrier signal for a vocoder. In this way the vocoder can follow the pitch of the incoming signal. This avoids a 'robotic' sound, and is especially useful when you want to shift the apparent pitch of a sound (e.g. to make a male speaker sound female). Of course, sometimes you WANT the robotic sound, so in that case you should leave out the pitch tracking! The following patch illustrates how to use a pitch tracker in conjunction with a vocoder.

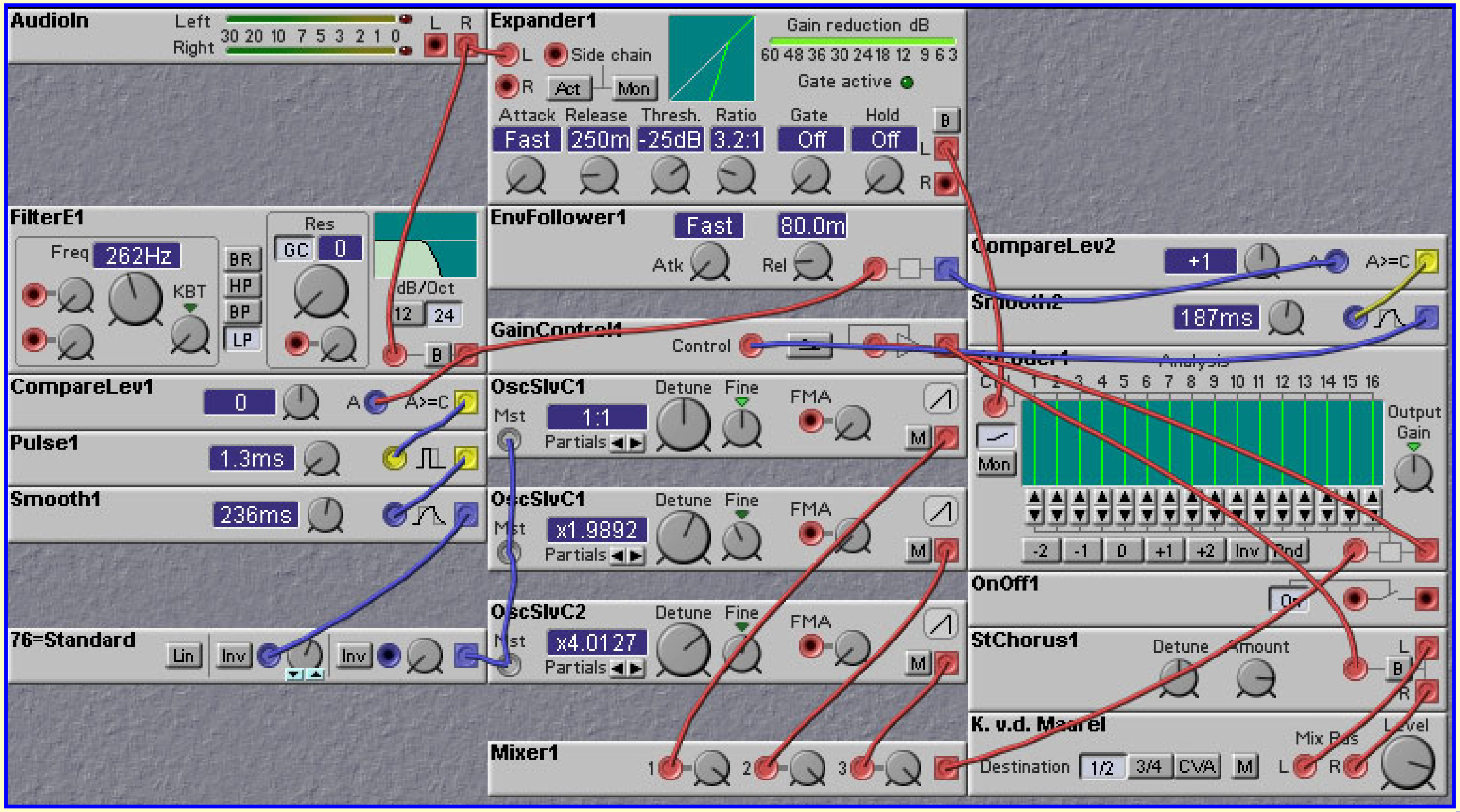


Figure 8.16. A pitch follower feeding into a vocoder (K. v.d. Maarel).

The final pitch tracking patch that we will look at uses a *phase-locked-loop* to track the pitch of the input. The phase-locked-loop is a circuit that is used extensively in communication systems, and in high-speed computer chips. Phase-locked-loops work by measuring the difference in phase between two signals - an input signal, and the output of an oscillator. The phase difference signal is smoothed with a lowpass filter and then used to change the frequency of the oscillator. If the phase difference is negative (phase of the oscillator lower than that of the input) the oscillator frequency is raised slightly, increasing the phase of the oscillator. This will tend to make the phase difference less negative. Similarly if the phase difference is positive (phase of the oscillator waveform is higher than that of the input signal) then the frequency of the oscillator is lowered slightly. When the phase difference is exactly zero, the oscillator frequency is held constant. In the following patch the phase comparator is implemented with a multiplier. The input signal is heavily amplified and clipped with an overdrive module to make it binary-valued (like a square wave). This binarised input is multiplied by the output of the tracking oscillator and the product is smoothed by the loop filter. In this design the phase-locked-loop tries to make the phase difference between the binarised input and the oscillator waveform 90 degrees. When the two signals are 90 degrees out of phase their product waveform (the output of the multiplier) will be half the time positive and half the time negative. Therefore the output of the loop filter, which averages out the multiplier output, will be zero (save for some ripple). If the input frequency rises, the phase difference will rise, and the output of the multiplier will have a duty cycle of more than 1/2, and the loop filter output will therefore also rise, causing the tracking oscillator frequency to rise. Similarly, when the input frequency drops, the phase difference will drop slightly causing the loop filter output to drop, resulting in a decrease in the frequency of the tracking oscillator.

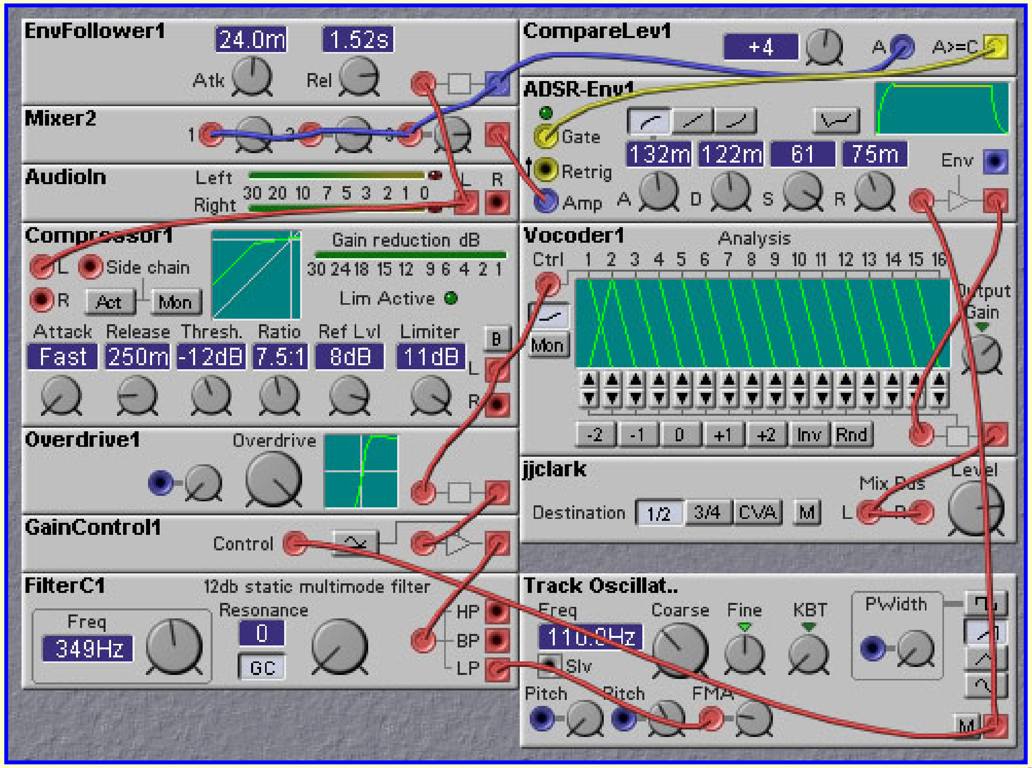


Figure 8.17. A phase-locked-loop pitch follower (J. Clark).

Pitch tracking can be used to control more than just the pitch of an oscillator. It can also be used to control filter cutoff, for example. Or even something non-frequency related, like the slope of an envelope generator attack (using a modulated enveloped generator module). The following two patches demonstrate such uses of a pitch track signal. The first uses the pitch track signal to control the cutoff frequency of a filter, the second to control the center frequency of a phaser module.

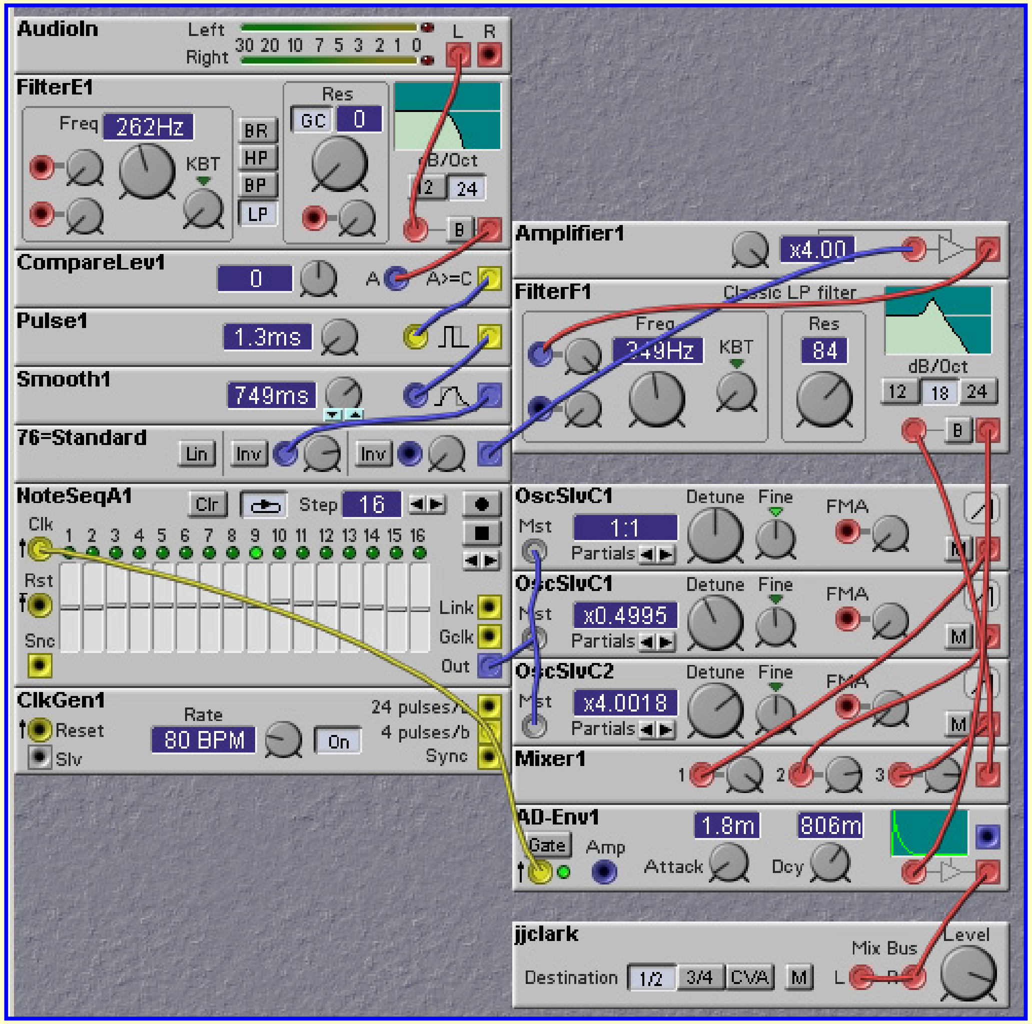


Figure 8.18. Using a pitch track signal to control filter cutoff frequency (J. Clark).

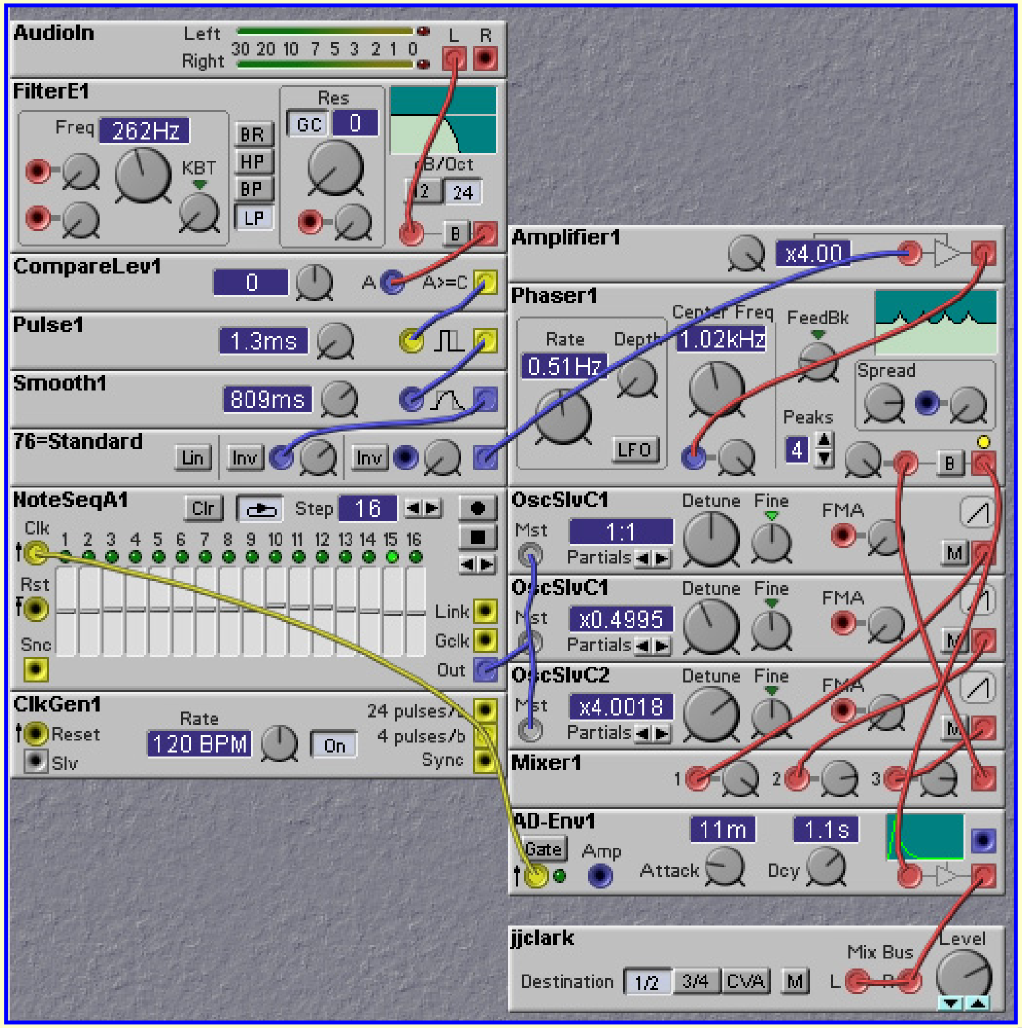


Figure 8.19. Using a pitch track signal to control the center frequency of a phaser (J. Clark).

[Prev (Physical Modeling)][Next (Using the Logic Modules)][Table of Contents]

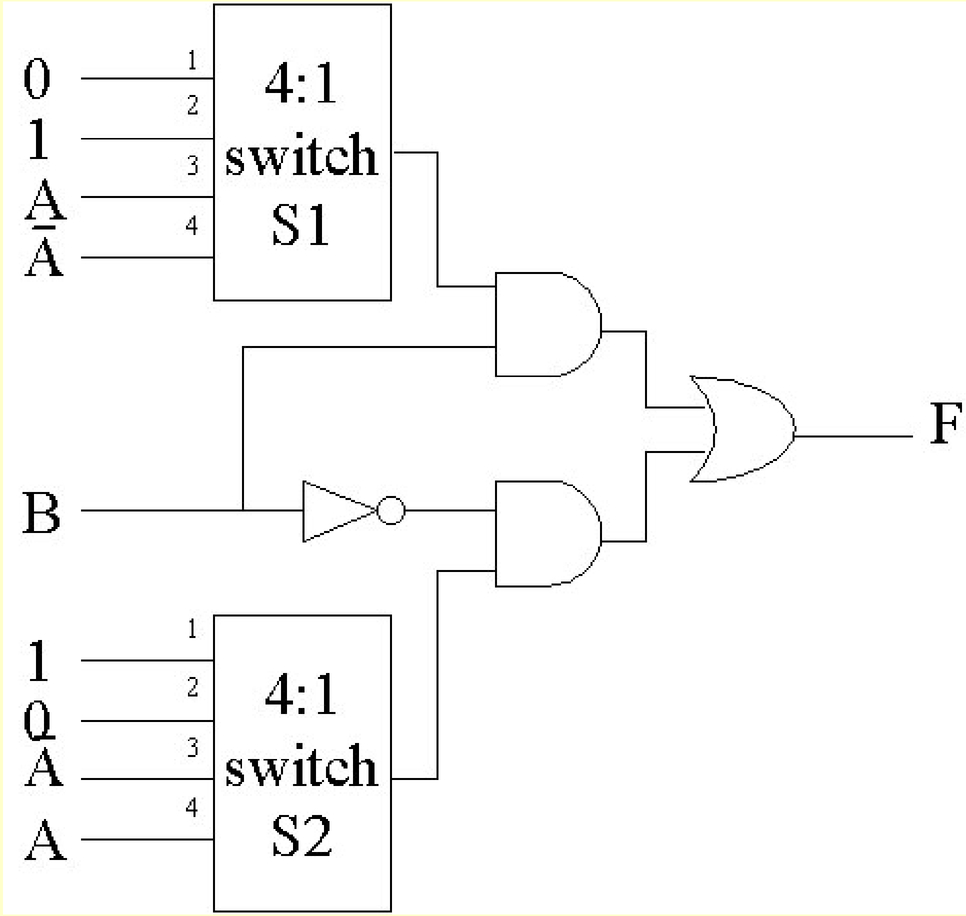
Chapter 9. Using the Logic Modules

Music synthesizers contain more than just the circuits involved in the generation of sound. In addition to the oscillators, filters, envelope generators and the like, are other circuits whose job it is to re-configure the audio chain and to control when and how various events occur. These circuits are usually logical in nature, and are implemented most often in software, with microprocessors or micro-controller chips. In some older and simpler synthesizers, microprocessors are not used. Instead logical operations are implemented with collections of small and medium complexity logic gates. And in Nord Modular patches there are no "microprocessor modules" to be had, so one is forced to construct logic processing structures out of the basic logic modules that ARE available. In this chapter we look at a few types of useful logic circuits and how they can be implemented on the Nord Modular.

9.1 Complex Logic Functions

Universal Logic Modules

The Nord Modular provides only the most basic logic functions - AND, OR, XOR, and NOT (or Inversion). Fortunately, one can make *any* Boolean function (logic function) out of AND gates and NOT gates alone. In the following figure we show a "Universal Logic Module", which can implement any of the 16 different possible Boolean (logical) functions of 2 inputs. The different logical functions are selected with the two 1-of-4 switches, according to the table given in the figure.



S1	S2	Function
1	1	$\bar{B}$
1	2	0
1	3	$\bar{A}\bar{B}$
1	4	$A\bar{B}$
2	1	1
2	2	B
2	3	$\bar{A}+B$
2	4	$A+B$
3	1	$A+\bar{B}$
3	2	AB
3	3	$AB+\bar{A}\bar{B}$
3	4	A
4	1	$\bar{A}+\bar{B}$
4	2	$\bar{A}B$
4	3	$\bar{A}$
4	4	$\bar{A}B+A\bar{B}$

Figure 9.1. Schematic for a universal logic module. The different logic functions are selected by the two 1-of-4 switches, S1 and S2.

The universal logic module is easily implemented in the Nord Modular, as shown in the following patch. The envelope generator is used just for testing purposes; its gate LED can be used as a logic level indicator.

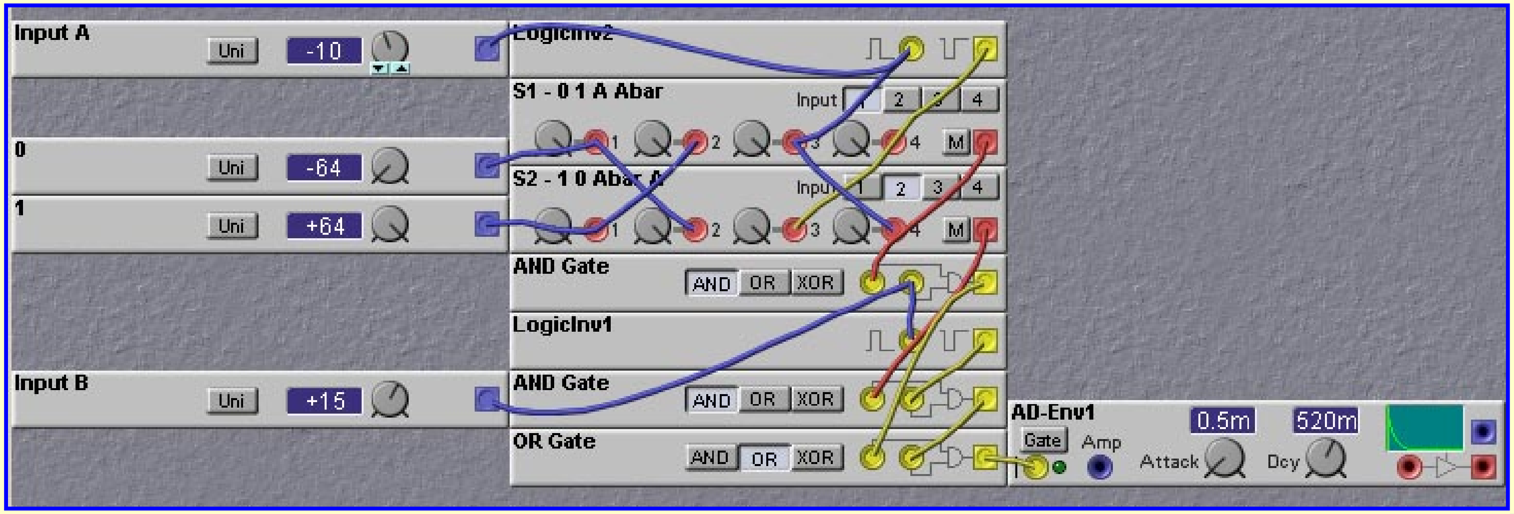


Figure 9.2. A Nord Modular implementation of the Universal Logic Module (J. Clark).

Multiplexers

One of the more useful logic functions is the multiplexer. You can think of a multiplexer as a "voltage-controlled switch". A binary-encoded select input is used to choose one of a number of inputs and pass the value of the chosen input to the output. Although they could be formed from many logic gates, multiplexers are most easily implemented on the Nord Modular using X-Fade (Cross-Fader) modules. The following figure shows how to construct a 1-of-4 multiplexer on the Nord Modular. You could use these circuits in place of the 1-of-4 switches in the universal logic module patch, to obtain a voltage-controlled universal logic module. The multiplexer circuit works in a hierarchical fashion. The first select bit is fed into the fade control input of two X-Fade modules. This selects one of the two inputs to the X-Fade modules. The outputs of the 2 X-Fade modules are then fed into another X-Fade module. The fade input of this module is controlled by the second select bit. This circuit can be extended to larger numbers of inputs, by adding more levels. The number of levels that is needed is given by the logarithm base 2 of the number of inputs (rounded up to the next integral value). For example, if you had 12 inputs (which we do in the arpeggiator circuit described later in this chapter) you would need ceil(log_2(12))=4 levels. The last level always has one X-Fade module, the preceding level has two, and so on, increasing the number by a multiple of 2 for every level. Thus, for a 4-level circuit there would be a total of 1+2+4+8 = 15 X-Fade modules. Since, in our example, we only have 12 inputs, some (2 in this case) of the X-Fade modules in the first level can be deleted, as they will have no inputs.

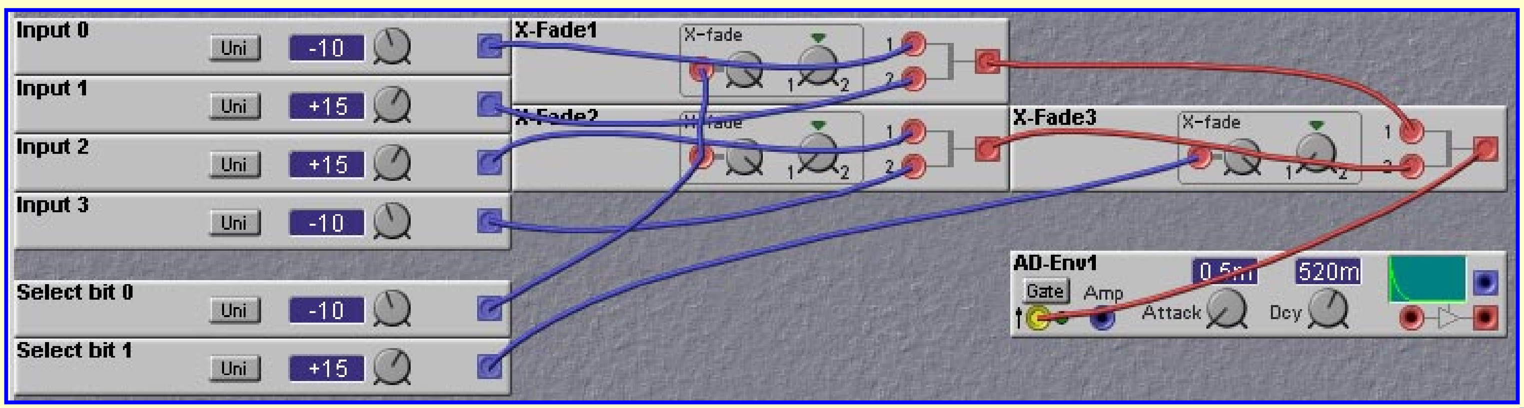


Figure 9.3. A Nord Modular implementation of a 4 to 1 multiplexer circuit using X-Fade modules (J. Clark).

Decoders

Another useful complex logic circuit is the Decoder circuit. This circuit takes in a binary input which is used to select which of a number of outputs will be set to a high level. All the other outputs are set to a low level. This can be used, for example, to select one of a number of oscillators or envelope generators to activate, or to choose a particular sample/hold module to sample. The patch shown in figure 9.4 shows how to make a 1-to-4 decoder on the Nord Modular.

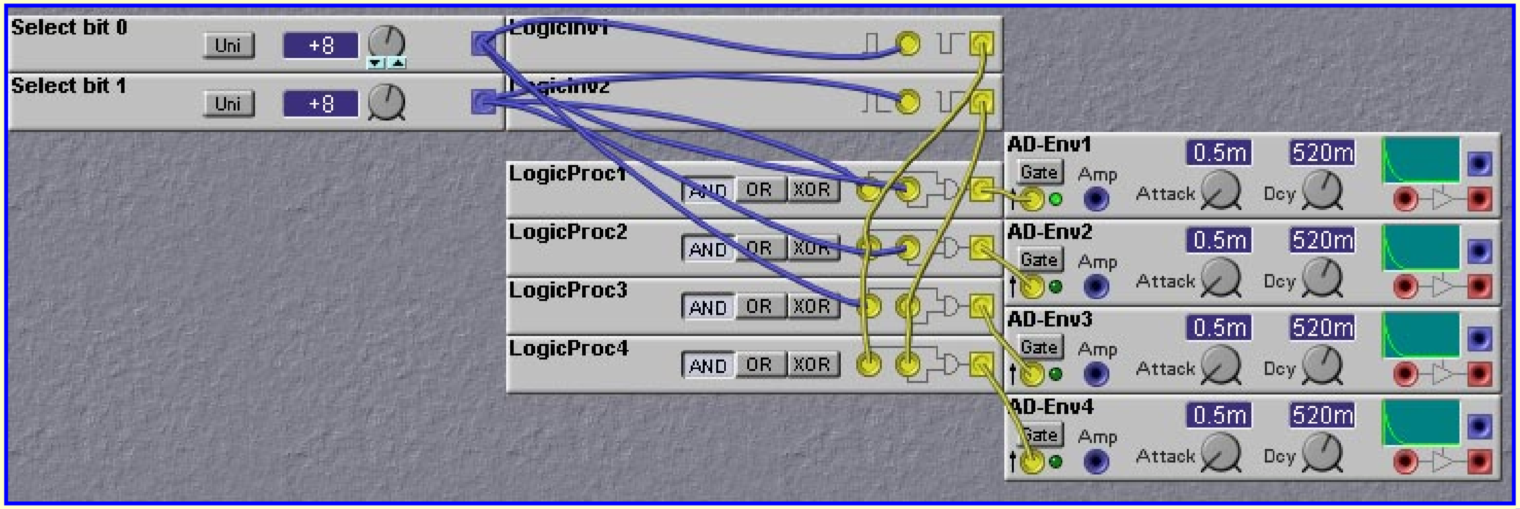


Figure 9.4. A Nord Modular implementation of a 1-to-4 decoder circuit (J. Clark).

Analog to Digital Converters

In this chapter we are concentrating mainly on binary signals, which take on either a low or high value. This is the case for the yellow logic signals in the Nord Modular, but the blue and red (control and audio) signals have a wide range of possible values. It is sometimes useful to be able to convert from a binary coded version of a quantity (using a set of binary signals to encode the number) to an integer valued signal such as the control or audio signals. And vice-versa. We will refer to the process of converting the integer valued signal to its binary coded version as "Analog to Digital Conversion". Of course, all of the signals in the Nord Modular (apart from the signals at the input and output jacks) are digital, so this terminology is inaccurate at best. But it captures the spirit of what we are trying to do.

There are a number of ways to implement A-to-D conversion, borrowed from techniques used to implement real A-to-D circuits. One of these techniques is shown in the following figure. The figure shows a fully combinational logic (i.e. no memory elements) approach. The basic idea is as follows. The input value, to be converted, is compared with a value of 2^N, where N is the number of bits to use in the binary code of the output. If the input is greater than or equal to 2^N, we set the Most-Significant-Bit (MSB) of the output code to 1 and subtract 2^N from the input value. Otherwise we set the MSB to 0 and do not alter the input. We then take the (possibly altered) input and repeat the process, this time comparing it to the value of 2^(N-1). Again, if the (altered) input is greater than or equal to 2^(N-1) we set the value of the next MSB output to 1 and subtract 2^(N-1) from the input, otherwise set the next MSB to 0 and pass the input on unchanged. We continue in this fashion, level by level, until we reach a comparison with 2^(N-N)=1. The output of this comparison represents the Least-Significant-Bit. A 4-bit implementation of this approach on the Nord Modular is shown in the following figure.

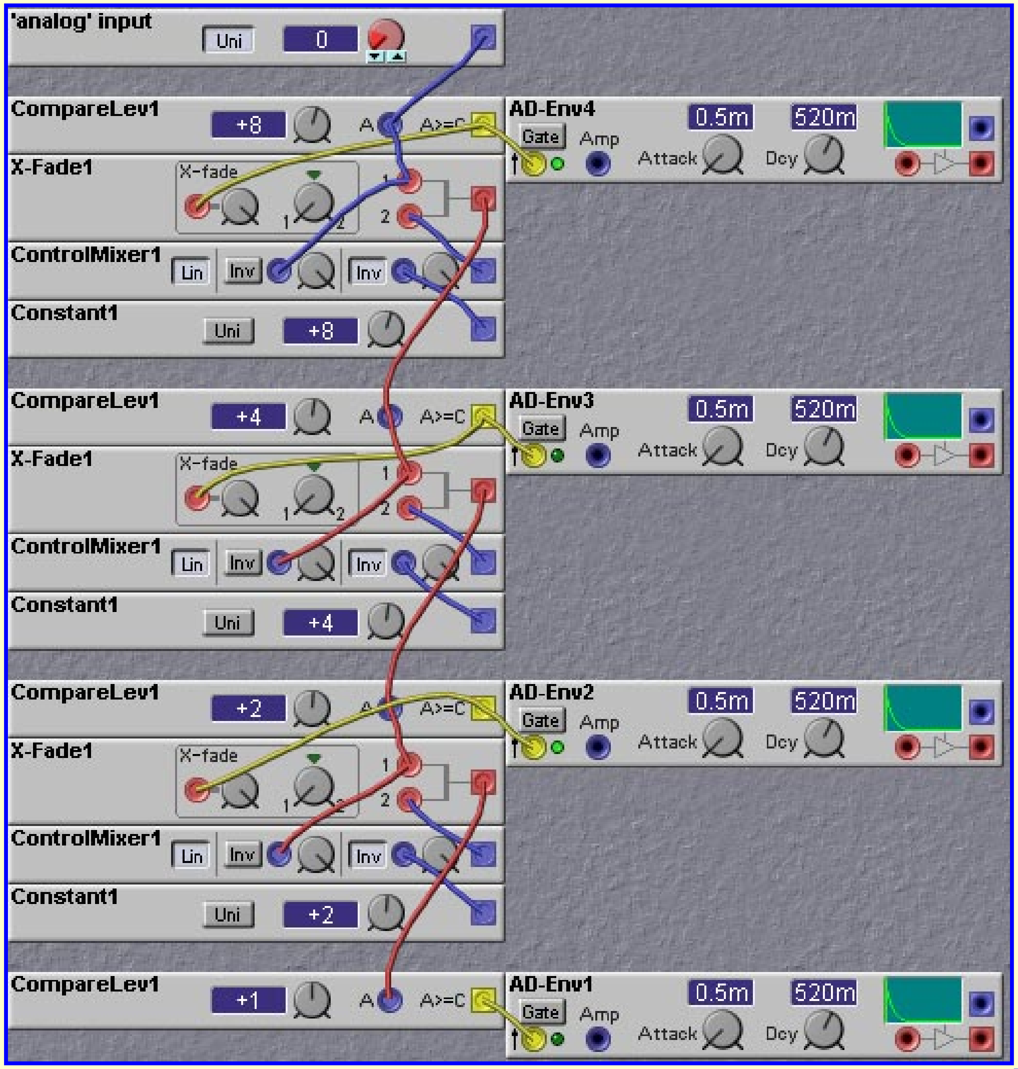


Figure 9.5. A Nord Modular implementation of a 4-bit Analog to Digital converter circuit (J. Clark).

9.2 Flipflops, Counters and Other Sequential Elements

The capabilities of logic circuits are greatly expanded if they contain some form of storage or memory. Circuitry containg storage elements is often refered to as "sequential logic" referring to the time dependent behavior that such circuits posess. The memory can store intermediate results or keep track of various counts, which can then be used to modify the circuit's functioning in complicated ways. In this section we will look at a few ways in which sequential logic blocks can be implemented on the Nord Modular.

Flipflops

The Nord Modular does not include a flipflop module, but we can make one using the sample-hold module. This is shown in the following patch, which implements a "toggle" flipflop. On each rising clock transition the output of the flipflop alternates, or "toggles", between high and low values. In the patch pressing the C4 key alternately turns the sound on, and then off. This toggling action is very useful in a wide variety of control tasks, and is also very useful in making so-called "noodles" or auto-play patches.

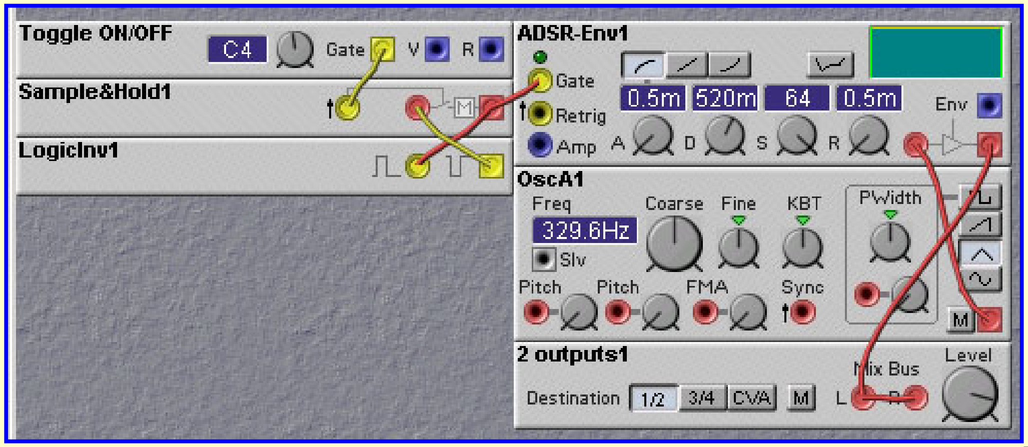


Figure 9.6. A toggle flipflop made using sample-hold modules (R. Hordijk).

One can use the sample/hold feedback approach to make a so-called "D" flipflop (D is for "data"). A D-flipflop merely holds the value of its input when the clock goes high. Now this seems to be the exact operation performed by the sample/hold modules. So why do we need to do anything else? Well there is a small problem, one that is actually quite prevalent whenever you try to emulate parallel or concurrent systems (such as analog synthesizers), in which every thing happens at the same time, with digital systems where only a few things (or only one thing) happen at a time. In the Nord Modular, on a single DSP slot, operations are executed one at a time. This means that the outputs of some modules are actually computed before the outputs of others. Normally the order in which computations are done does not matter. But sometimes it does, and in obtaining the proper functioning of a D-flipflop is one of these times. To see this, consider a string of sample/hold modules, intended to emulate a string of D-flipflops. Such strings are referred to as *shift registers* and are very useful. So encountering such a situation is not unlikely. Now consider two adjacent sample/hold modules, where the output of one feeds into the input of the other. Suppose that the operation is synchronous, which means that every unit should change state at the same time, usually in response to a clock signal transition. So let both the of sample inputs of these two sample/hold modules be connected to the same clock signal. Let the output of both sample/hold modules be initially zero, and let the input to the first be connected to a high signal. Now, what will happen when a rising transition of the clock signal is received? Well, the answer depends on the order of the computations! If the second sample/hold module is computed first, everything will be OK. After all the computations are done, the output of the first sample/hold will be high, and the output of the second will remain low. But if the order of the computation is reversed, the answer will be "wrong". Look carefully. If we compute the output of the first sample/hold first, it will sample its input, which is high, and therefore set its value to be high. Now we compute the output of the second sample hold, but its input is now high, since we have already computed the first module's output! Thus the output of the second sample/hold module will also be high. This behaviour is bad for two reasons. First is that sometimes we get the wrong answer. But worse still, we can't say for sure when designing the patch whether we will get the right answer or not, since we have no control over the order of computation.

But we *can* design circuits that are insensitive to the order of computation. We can use pairs of sample-hold modules that are clocked with different signals, out of phase relative to each other. This *dual-phase clocking* allows us to control when things happen. The resulting circuit is called a "Master-Slave" D-flipflop. The master is the first of the two sample/holds in the pair, and the slave is the second, which merely follows the lead of the master. A master-slave D-flipflop is shown in the following patch.

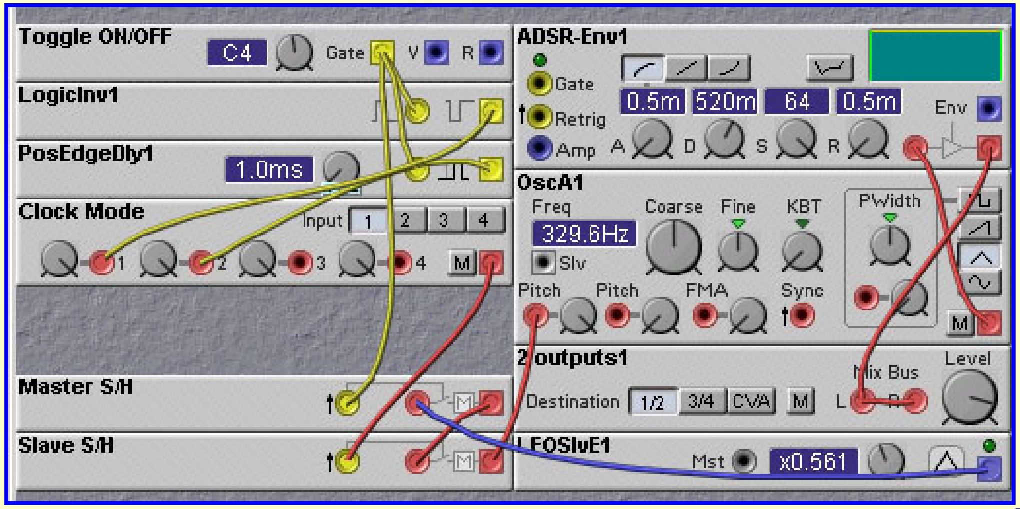


Figure 9.7. A master-slave flipflop made using sample-hold modules (J. Clark).

The patch shown above exhibits two different ways of clocking the flipflop from a single clock signal. The first way is to clock the first sample/hold module with the clock signal, and clock the second with the inverse of the clock signal. This has the drawback that the clock signal has to go high and then low again before the input value appears at the output of the second sample/hold module. In many situations, especially those in which the clock signal is a regular periodic signal, this will cause no problem. In some situations, it may be a long indeterminate time before the clocking signal goes low. For example, in the above patch, to get a new value latched into the D-flipflop one has to first press the C4 key and then release it. It might be a long time before you release the key. It would be preferable if the value could be stored in the flipflop whenever you press on the key. In this case the second clocking technique may be better. In this technique the clock signal is used to clock the first sample/hold as before, but is also fed into a positive edge delay module to generate a slightly delayed version of the clock signal. This delayed pulse is used to clock the second sample/hold module. This will guarantee that the first sample/hold module is clocked before the second. The delay time is set to the minimum non-zero value of 1 millisecond. If the clock rate is faster than 1KHz, this delay value will be too long and the second sample/hold will never be clocked. In this case an audio-rate delay line would have to be used. This is an expensive solution, however, so in such high speed situations it is better to use the first clocking technique and merely use an audio inverter to invert the clock signal.

Counters

Counters are useful for many musical tasks. There are many different ways to construct counters in the Nord Modular. Which way is best depends on the application. Many sequencing tasks require counters, and for these the ready-made sequencer modules fit the bill. The control and note sequencer modules are very good for general purpose counter design as well. They are easy to use and involve merely setting the levels of the sequence step values to the desired count values. For example, if you wanted to count repeatedly in the sequence 1,2,4,5,1,2,... you could take a control sequencer and set the loop period to 4 and set the first four levels to 1,2,4 and 5. Voila! Instant counter.

Using the sequencer modules as counters is an easy solution, but sometimes one needs more flexibility. Often one requires a "binary" output from the counter, where the count value is not in a "voltage" but rather represented as a multiple-bit digital word. One could use a "digital to analog converter" to convert a voltage level from a sequencer module into a digital word, but this would be inefficient. There are more straightforward techniques. One could implement a binary counter directly using toggle flipflops, or even D-flipflops, but unless you need to process audio-rate signals it is more efficient to use ClkDiv modules. If you need a binary counter that counts on its own at a certain constant rate (rather than using an external clock signal) you can use synced square-wave LFOs, whose frequencies are powers-of-2 multiples. The lowest frequency square wave output is the Most-Significant-Bit (MSB), while the highest frequency square wave output is the Least-Significant Bit (LSB).

Counters can be made with sample/hold modules feeding back to themselves through summing (or mixing) modules. The output of the sample/hold is added to a set value (say 1 or -1, for example) and the sum fed back to the sample/hold input. When the sample/hold is triggered, the new count value is loaded. This technique is used in the patch shown below, an absolutely delightful classic created by *Keith Crosley*:

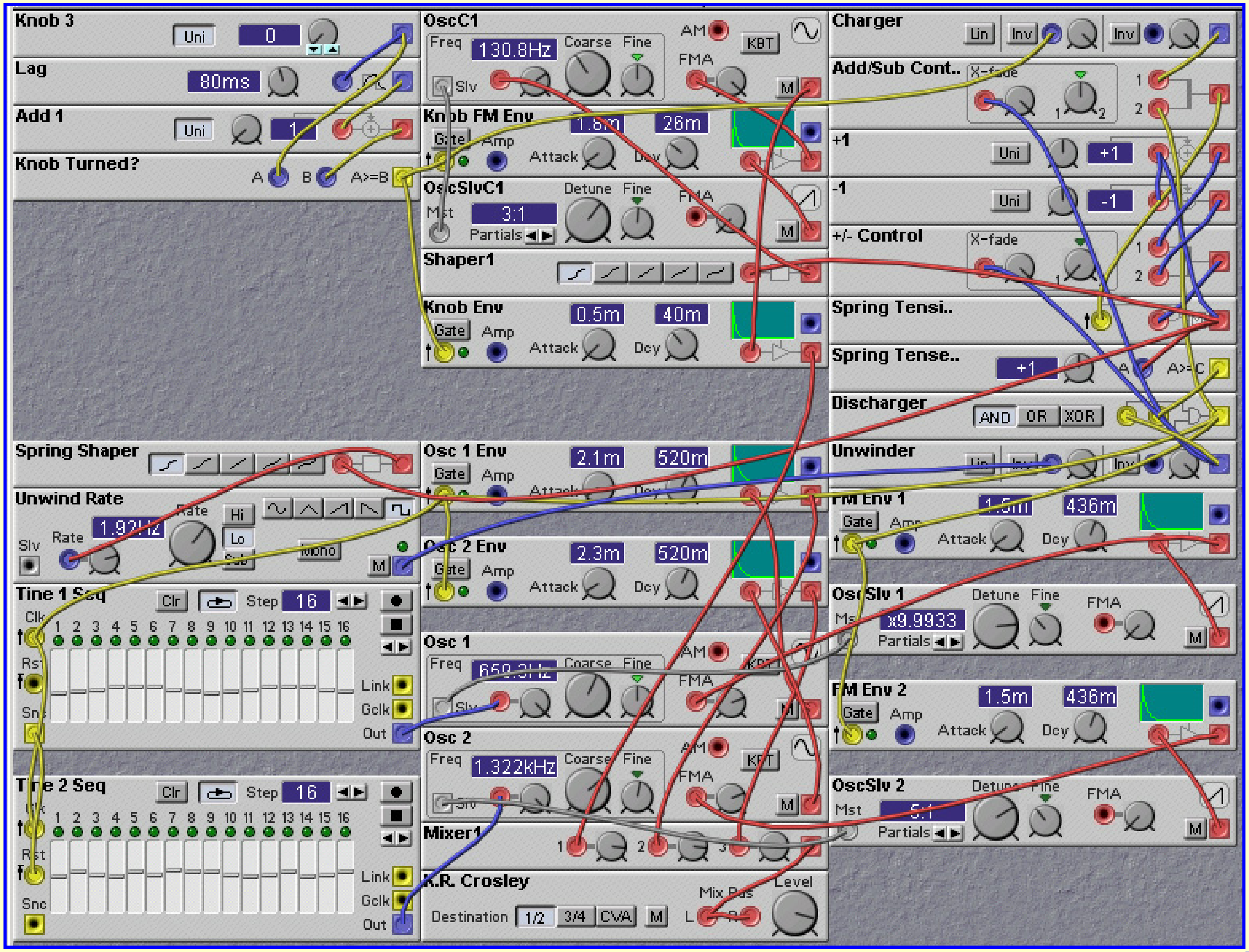


Figure 9.8. A patch simulating a wind-up toy. It contains many different uses of the logic modules (K. Crosley).

In this patch, an "Up/Down" counter (one which can either count up or count down) is formed from the sample/hold module labelled "Spring Tensi.." and the summing modules labelled "+1" and "-1". Which of these summing modules gets fed back to the sample/hold input determines whether the counter counts up or counts down. The selection of this feedback is done by the crossfade module labelled "+/- Control", under the direction of the "Unwinder" signal derived from the Squarewave LFO module labelled "Unwind Rate". When the phase of the squarewave output of this LFO is high, the input to the sample/hold comes from the "-1" summing module, turning the counter into a down counter. When the LFO waveform is in its low phase the counter is an up counter. This part is simple enough. The counter alternates between being an up counter and a down counter following the squarewave output of the "Unwind Rate" LFO. The actual *clocking* of the counter, on the other hand, is somewhat more complicated.

9.3 Asynchronous Elements

Although most synth programmers are comfortable with sequential logic elements such as counters, much of music cannot be tied to a rigid clock. In fact, it is better to think of an electronic musical instrument that is involved in a performance as an *Event-Driven* system. In such a system, what drives transitions from one state to the next is not the transitions of a clock signal, but the occurence of an Event, usually generated externally to the system. For example, events can be created by the pressing of a key on the keyboard of a synthesizer, or the striking of a drumstick on an electronic drum pad. As you are no doubt aware, even when playing music such as Bach, where a very precise periodic rhythm is desired, the events produced by human performers are by no means periodic. Some players are much better than others, of course, but then there is the drummer from your neighborhood garage band. Some music is arrhythmic. Some music has the timing of its events derived from the environment. In all of these cases, musical events essentially occur *asynchronously*, that is to say, they occur independently of a periodic clock.

Asynchronous systems have been studied extensively by computer engineers and many useful basic building blocks have been developed. One of these building blocks is the "C-element". A C-element can be thought of as an "AND-gate" for events. An event is generated on the output of the C-element only after an event has been received on both of the inputs. Thus, the C-element acts as a synchronizer - if an event occurs on one of the inputs, the C-element will wait until an event occurs on the other input before changing the output. The symbol for an inverting C-element is shown in the following figure, along with its truth table, and its implementation in the Nord Modular using an XOR gate and Sample/Hold module.

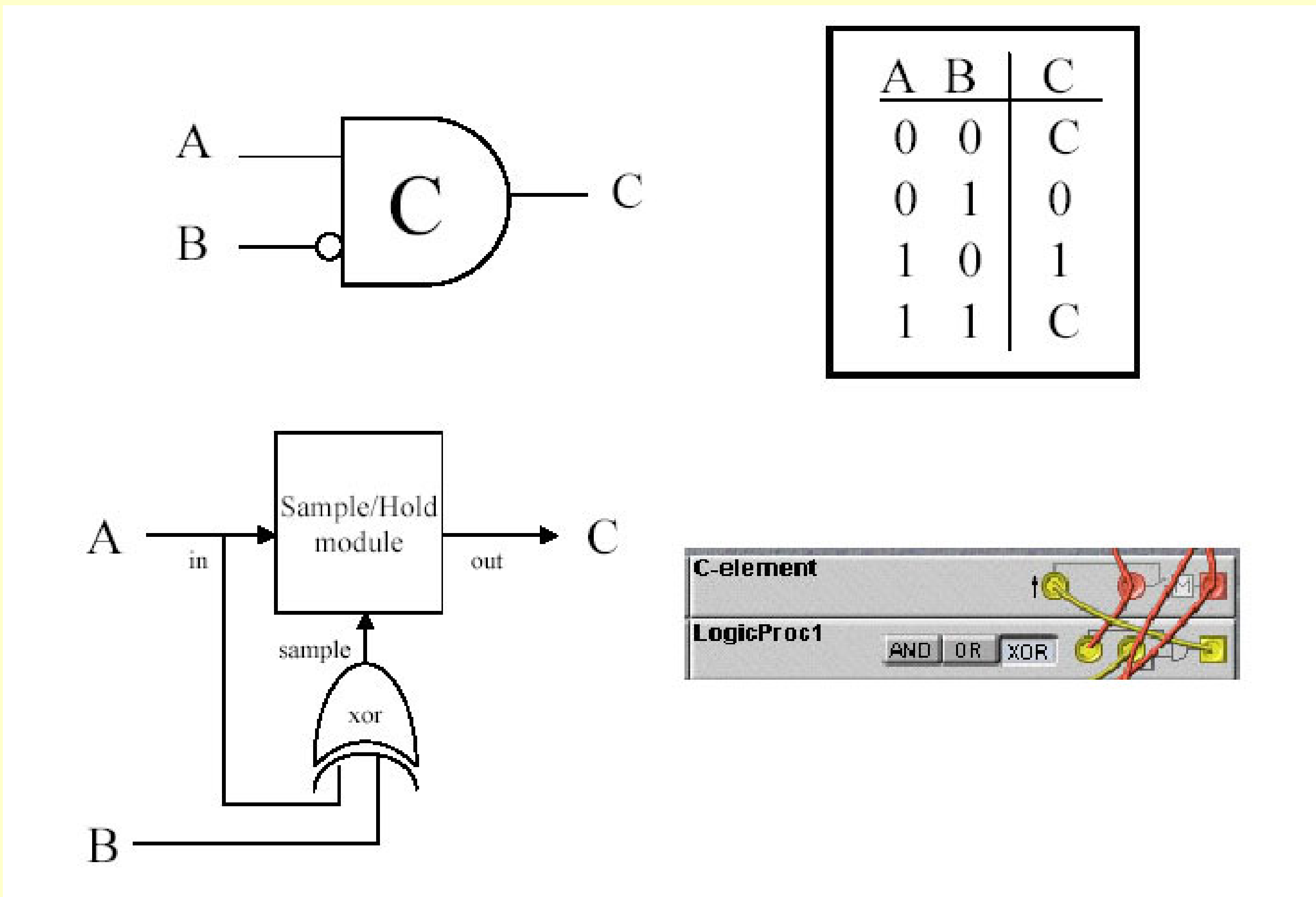
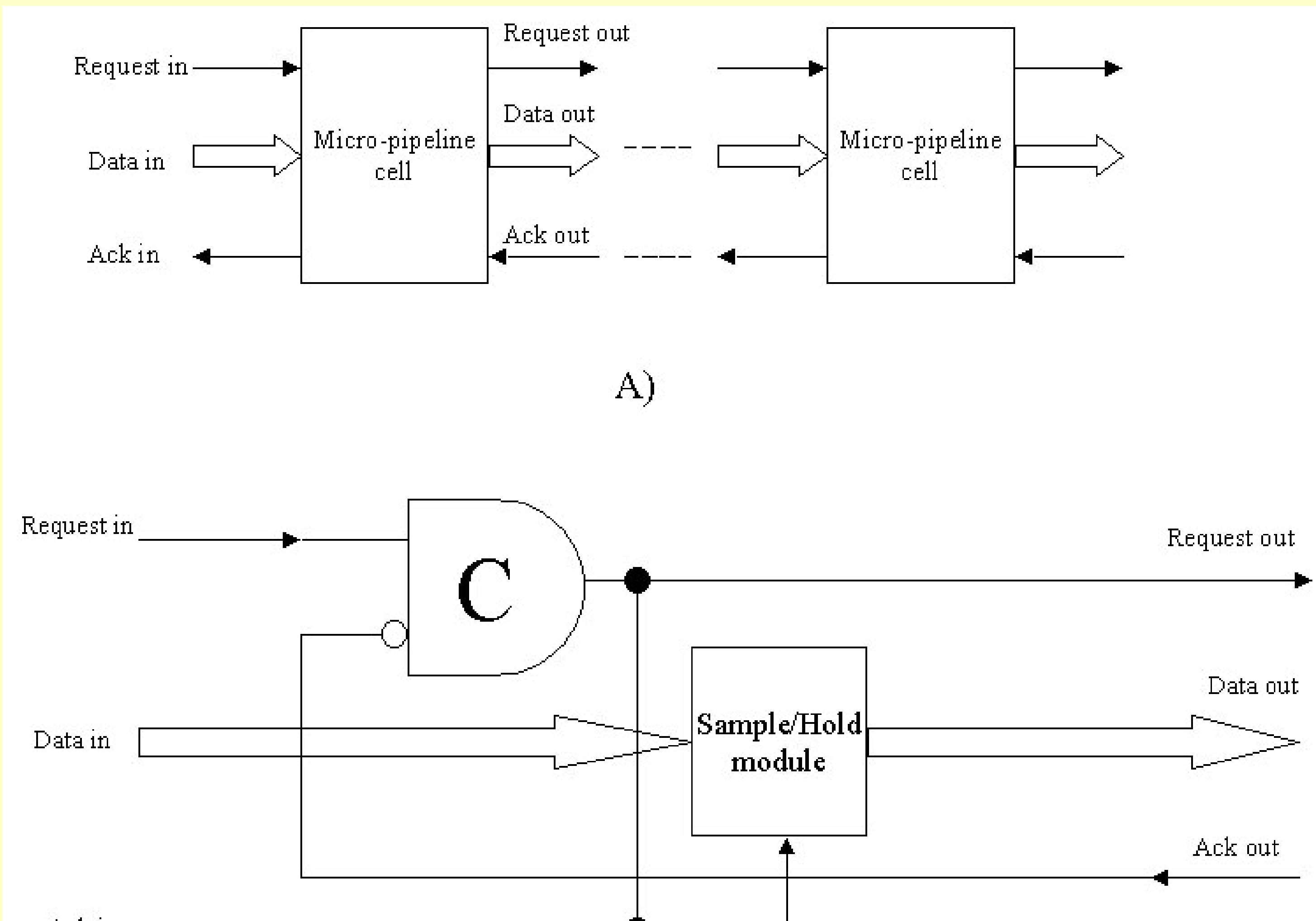


Figure 9.9. The (Inverting) Muller C-element. Upper-left: graphical symbol. Upper-right: truth table. Lower-left: an implementation using an XOR gate and a sample-hold module. Lower-right: implementation using Nord Modular modules.

The output of an inverting C-element is held whenever the inputs both have the same value. If the values are different, the output is set to the value of the A input (the non-inverting one).

C-elements are often used in control mechanisms for various asynchronous operations. One example that is useful in music applications is the asynchronous FIFO (first-in-first-out) buffer. In an asynchronous FIFO events arrive (asynchronously) and stored in the buffer, and then read out asynchronously. Think of the FIFO as a tube in which one can insert balls in one end and can remove them out the other end. Clearly the first ball in will be the first one out. There is no need to put balls in at the same time you take them out, nor do you have to do the insertion or removal at regular time intervals. Of course, if the rate of insertion events is greater than the rate of removal events, events will tend to pile up in the FIFO, which might then overflow at some point.

The following diagram shows how such an asynchronous buffer can be made out of inverting C-elements. This circuit was invented by Ivan Sutherland, who termed it a micro-pipeline. The key aspect of this buffer is the "hand-shaking" that is going on. Data is not just crammed into the buffer indiscriminitely. Instead, the source of data makes a "request" for data to be read into the buffer. This is done by raising the Request_In control line to a high value. If the Ack_Out line is low, then the request will be passed on, and the storage element will be loaded (in this case the storage element is a sample/hold module). If, on the other hand, the Ack_Out line is high, then the request will be blocked, until such time as the Ack_Out line does go low. The Request_Out line is connected to the Ack_In line, so that when a request is honored, any requests upstream will be blocked until the Request_In goes low again. In this way, a block at one point in the buffer (usually at the last stage, where data is waiting to be read out) will eventually propagate all the way to the first stage if the last stage block lasts long enough for all of the buffer cells to be loaded. If the last stage is being read out at a rapid enough rate, there should be no blocking anywhere, and data loaded into the first stage should quickly find its way to the last stage, where it is then read out.



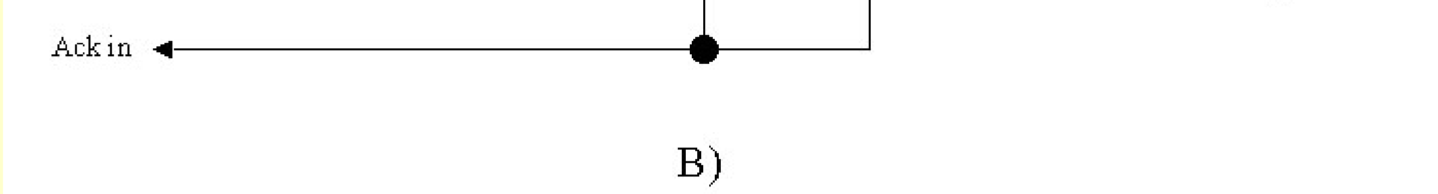
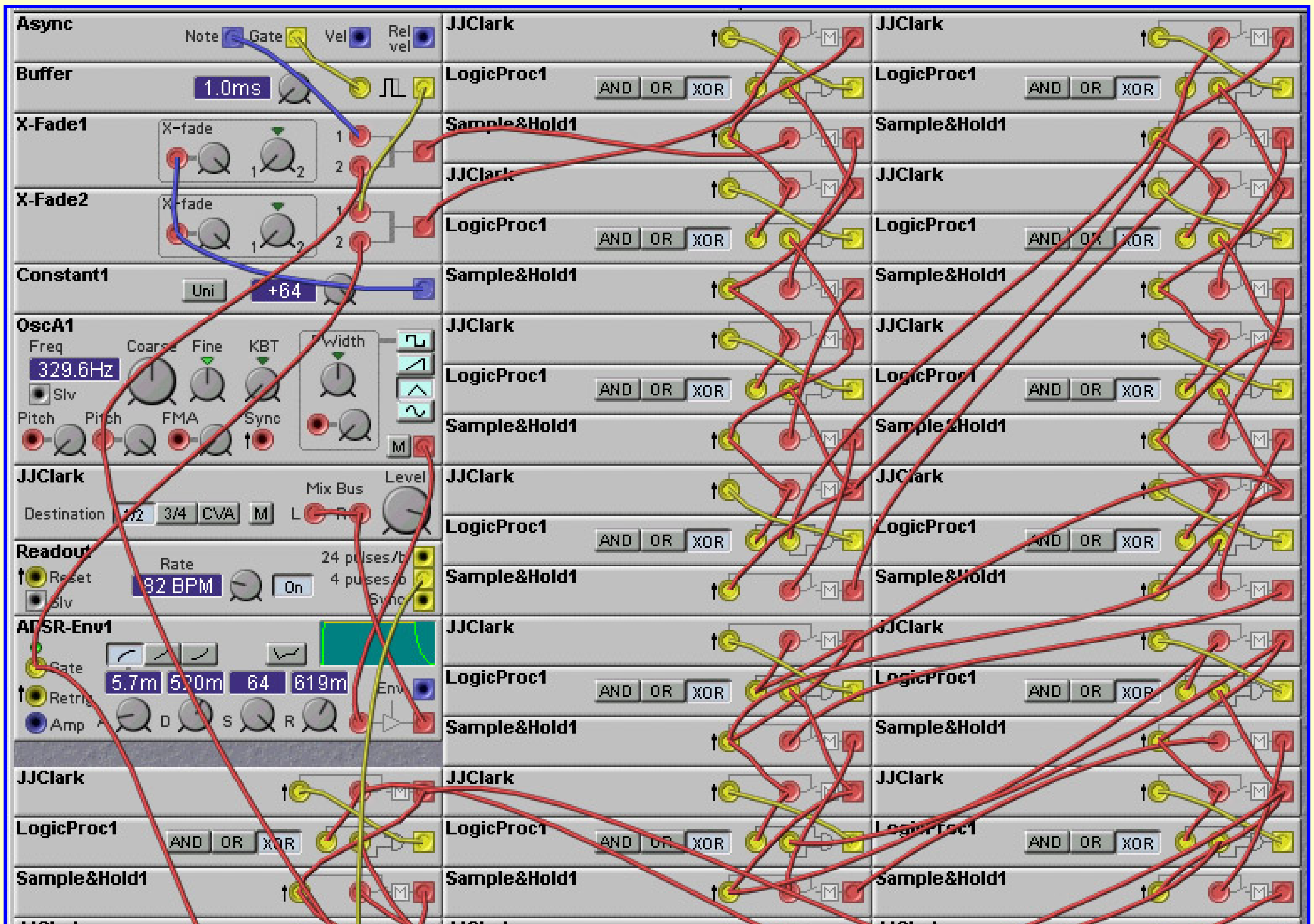


Figure 9.10. An asynchronous FIFO buffer made using C-elements. a) A block diagram of the async FIFO consisting of chained storage element blocks with asynchronous hand-shaking to control transfer of data from one block to the next. b) Circuit for one of the micro-pipeline blocks.

The following patch shows an implementation of a 16-stage asynchronous FIFO buffer on the Nord Modular. The input events come from the keyboard module, and hence are generated by the external world, perhaps by a garage band drummer, and therefore not guaranteed to be periodic. The output events are generated by a clock. Therefore this patch has the effect of making the possibly irregular input events into a nice smooth sequence (as long as the input events come in as fast, or faster, than the output events, otherwise the outputs will stop). Of course, one can just as easily do the opposite, where a regular sequence of events is input to the buffer, while the readout events are irregular. The last cell blocks until the readout clock occurs, at which time the buffer is shifted by one place, and then resumes blocking. As long as the read-outs occur more often than the input requests, the buffer will never become full, and the input (first) stage will never become blocked.



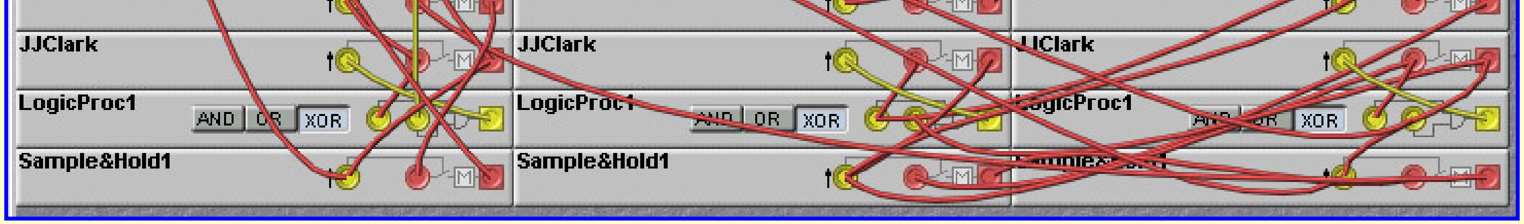


Figure 9.11. A 16-element asynchronous buffer Nord Modular patch (J. Clark).

9.4 Arpeggiation

Constructing an arpeggiator in the Nord Modular (or in any modular, for that matter) is deceptively difficult. It might seem that one just needs to throw together some sample/holds and be done with it, but things are not that simple. The main problem is that the order in which notes are played in an arpeggiaton is usually in order of pitch, rather than in order of being played. This means one can not just store the sequence of notes (e.g in sample/hold modules) as they are played. Instead one must scan the notes that are being played, in pitch order. This scanning must be done continually, as notes can be played or released at any time.

I will describe one particular implementation bit by bit. The patch is loosely based on the Korg Mono/Poly arpeggiator. It works by quickly scanning one octave of key detectors once during each arpeggiation cycle.

Let us start by looking at the part of the patch that scans the octave of keys. Twelve note-detect modules are used to detect the pressing of each of the twelve keys in the octave. Each of these modules outputs a high logic level when the corresponding key is pressed (based on NOTE-ON commands in the MIDI stream input to the Nord Modular). We use a counter to scan through each of the note-detect modules one by one. The scan counter is implemented with a set of ClkDiv modules. The outputs of the ClkDiv modules are passed through XOR gates which permits the scanning to be either upwards or downwards. The XOR gates either invert the counter outputs or leave them unchanged. The effect of inverting the counter outputs is to have the counter count downwards instead of upwards. The counter bits are used to control a 12-input multiplexer. A multiplexer is a circuit which selects one of a set of inputs and passes that signal to the output. So here, the scan counter selects which of the twelve NoteDetect module outputs to examine. The multiplexer is implemented on the Nord Modular with a set of CrossFader modules acting as binary analog switches. An expensive solution perhaps, but in the absence of voltage-controlled switches on the Nord Modular, the best we can do.

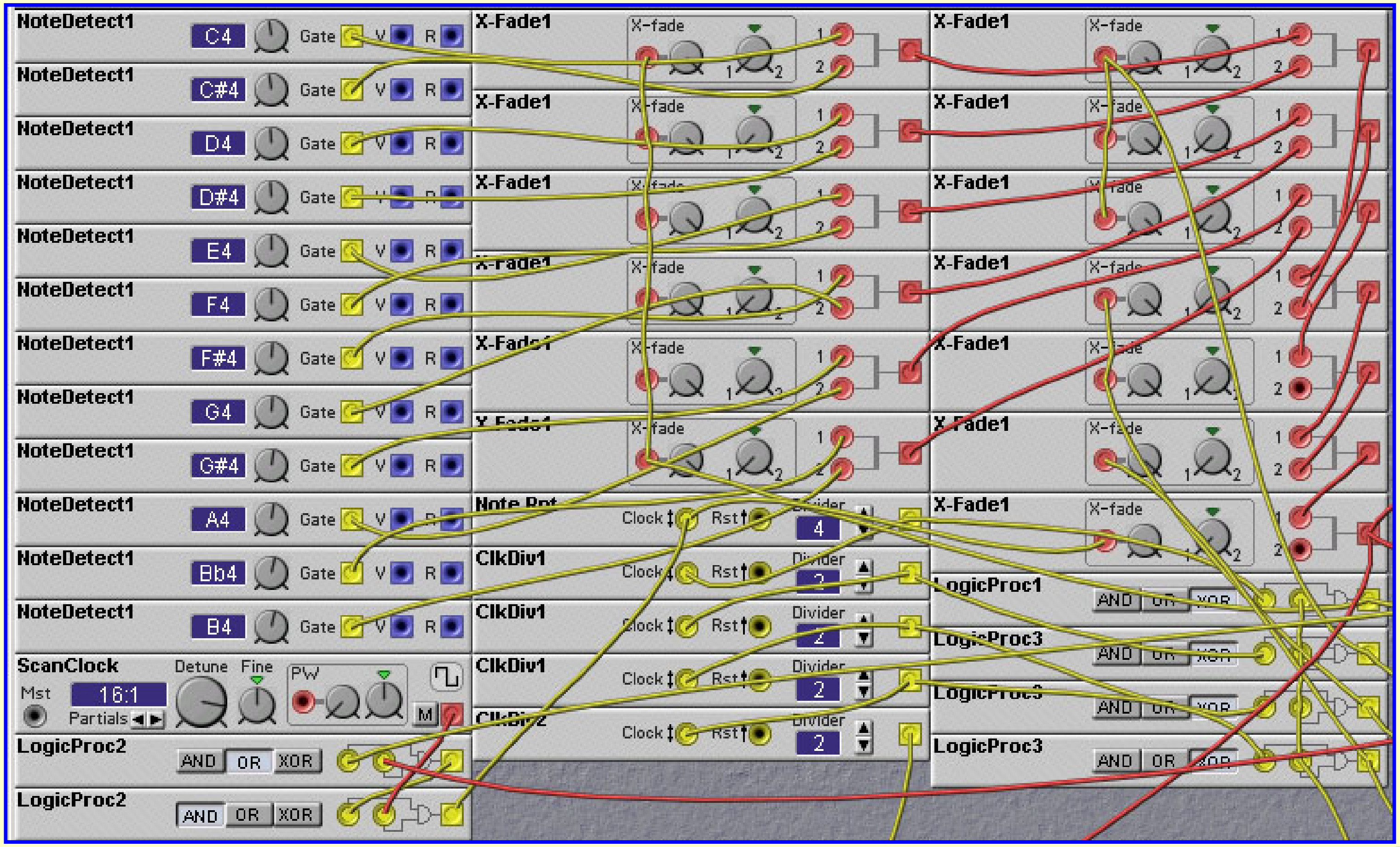


Figure 9.12. The portion of the arpeggiator patch that implements the scanning of one octave of keys (J. Clark).

When the scan detects a key down (i.e. when the output of the scan multiplexer goes high), the scan is halted until the arpeggiator clock goes low, at which point the scan is restarted and searches for the next key that is down. The envelope generators are triggered on the rising edge of the arpeggiator clock. Let's pause here and look at an example. Suppose there are three keys pressed, say the C#, F, and A keys of the octave we are scanning. Suppose the scan counter is counting up (we will look at up/down counting in a bit) from 0, which corresponds to note C of the octave. The output of the scan multiplexer will be low until the scan count reaches the count selecting the NoteDetector tuned to C#. Then the output of the multiplexer will go high. At this point the scan is stopped. Therefore the multiplexer output will stay high. The desired operation of an arpeggiator is to progress from one note to the next at times specified by some arpeggiation clock. Thus we restart the scan when we see the next rising edge of the arpeggiator clock. At this point the scan will continue until the next key that is down is found, in this case the F key. Then the scan stops again until the next rising edge of the arpeggiator clock. You might complain that there will be a bit of timing irregularity caused by the fact that there will be some time elapsed after the rising of the arpeggiator clock to the detection of the next key. So there will be a slight delay in the setting of the pitch to the new value from when the envelope generators were triggered by the arpeggiator clock. And you would be right, but the scan clock frequency is set to be quite high, so that this time delay is very small and unnoticeable in practice, and at worst adds a bit of pitch modulation.

The key scanning process is the difficult and tricky part of the arpeggiator design. The rest of the arpeggiator patch is just adding in the bells and whistles which make arpeggiators so much fun. As mentioned earlier, this arpeggiator is based on the one found on the Korg Mono/Poly synthesizer, which had a number of different modes of operation. The different features of the Nord Modulator implementation are as follows:
* The arpeggiation direction can be either UP, DOWN, or UP/DOWN. The direction is set by the XOR gates acting the output of the scan counter. In the UP/DOWN mode the input to the XOR gate is toggled (alternated between high and low values) at the end of each 1 octave scan. This toggling is implemented by a ClkDiv module acting on the Most Significant Bit of the scan counter. LogicProc3 is an XOR gate which combines the UP/DOWN toggle signal with the output of the "Up or Down" switch, which selects whether the

direction is UP or DOWN when UP/DOWN mode is not active (i.e. when the "Up and Down" switch is off).
* The arpeggiation pattern can be repeated in higher octaves. Either 1, 2 or 3 octave jumps can be selected. To see how this works, one should first understand how the pitch is determined. The scan count at the time when the scanning is stopped is converted to a pitch value by a "digital-to-analog" converter, consisting of four multiplier modules feeding a mixer. The multipliers multiply the counter bits by values of 1,2,4, and 8, and these products are added together to complete the conversion, and then fed into one of the pitch inputs of the master oscillator. In addition to this pitch value, another pitch input of the master oscillator is fed by an offset signal corresponding to the octave jumps. The octave offets are values of 0,12,24 or 36, depending on the octave jump setting. The time at which octave jumps are triggered is determined by the state of a counter (implemented by the ClkDiv module titled OctRpt in the patch). This selects the output of the octave jumps 1-4 switch. The period of this counter is set by knob 7. It determines the number of notes between jumps from one octave to the other. Interesting patterns can be obtained by setting this period to something different than the number of keys pressed.
In addition to the octave jumps, one can repeat notes within the arpeggiation pattern. The number of repeats of a note is set by a clock divider (the module entitled NoteRpt in the patch) placed in front of the scan counter. This counter has the effect of pausing the scan count until a given number of arpeggiator clock cycles have passed, thereby resulting in the repetition of notes.
* There is a ping-pong pan mode which merely alternates successive notes between the left and right outputs, giving the classic ping-pong effect.

[Prev (Speech Synthesis and Processing)][Next (Algorithmic Composition)][Table of Contents]

Chapter 10. Algorithmic Composition

Most of the readers of this document will no doubt be familiar with creating music using sequencing software or hardware. In this compositional process the composer determines the melody and the rhythm and transfers them to the sequencer by hand. But wouldn't it be nice if the computer could do some of the composing work? Is it even possible for a computer to do so? The computer generation of music is often referred to as *algorithmic composition* where the word algorithmic refers to the use of a computer program, or algorithm, to specify the music structures.

The Nord Modular contains some modules that are useful for composition. Some of these, such as the Sequencer modules, require human interaction to define the notes and rhythms. Others, such as the pattern generators, can run on their own. There are many Nord Modular patches in the archives, often referred to as "noodles", which need no human interaction to produce interesting music. Most of these noodles are carefully hand-crafted, however, and the music is clearly man-made. The noodles are often of a fixed nature, and cannot generate radically different musical creations. But, in spite of this, the Nord Modular IS capable of pure algorithmic composition, which can generate radically different music with little human intervention beyond setting a few parameters. In this chapter we will look at a few different types of algorithmic composition that can be implemented on the Nord Modular.

10.1 Chaos and Fractal Music

Mathematic chaos is a study of rather simple systems that produce rather complicated behaviour. This is perfect for making music on modular synthesizers, since we all know interesting music is complicated, but we can't implement very complicated equations with the limited resources of modular synths.

The patterns produced by chaotic systems bear some similarities to random noise, such as we use to make snare drums and so forth, but tend to have much more structure. The chaotic patterns tend to be "quasi-periodic", that is, the output of the system follows a regular pattern, but not exaclty. From cycle to cycle, the patterns are slightly different. Obviously, totally random noise is not too useful for specifying melodies and rhythms (although some may argue to the contrary). But chaotic patterns seem to have just the right mix of randomness and repeatability to produce interesting music. In the following section we present two Nord Modular patches which use different chaotic systems to produce self-generated music.

The first patch we will look at was designed by *Rob Hordijk*. It implements four simple non-linear ecosystems, defined by the formula that *Mitchell Feigenbaum* used to come to his 'Chaos'-theory. The equation that governs this chaotic system is

$$X(t+1) = G * X(t) * (1 - X(t))$$

This equation is known as the *Logistic Equation*. It was first used to model the population of animal species from year to year. In the equation above X(t) is the ratio of the actual population to the maximum population. Each iteration (e.g. from one year to the next) gives the new relative population in terms of the old one. The parameter G is the effective growth rate. The two terms in the equations model the fact that more animals will have more offspring (so growth is proportional to X), but will compete for resources (so that growth is also proportional to (1-X)). As the population increases the load on the environment increases reducing the availability of resources and limiting the growth rate. This is modeled by the (1-X(t)) term. For growth rates G less than 1, X(t) tends to 0. For growth rates between 1 and 3, X(t) tends to 1 - 1/G. Beyond G=3 a bifurcation occurs causing an oscillation from year to year. (corresponding to high and low populations in alternate years). Further bifurcations occur until at G = 3.53... chaotic dynamics set in. For values of G greater than 4 the system diverges (goes off to -infinity, or saturates at the most negative possible value).

What does all this animal husbandry have to do with the dog-eat-dog world of musical synthesis? Well, the chaotic behaviour exhibited by the logistic function can be used to produce interesting music. In the patch below the X values are used to play notes, but they could be used to control anything. In the patch there are four separate chaotic systems implemented. The systems can influence each other. They can be helped by their right neighbours to become more alive while their growth can be suppressed by their left neighbours. Sounds like real life, don't it? Well, actually it sounds like the Nord Modular, so don't worry, no need for peace keeping forces in your machine. Some settings can make all four systems slowly succumb, others can result in chaotic behaviour, and somewhere in between there should be "easy livin'". In figure 10.1 we show the behaviour of the logistic equation for various settings of the G parameter.

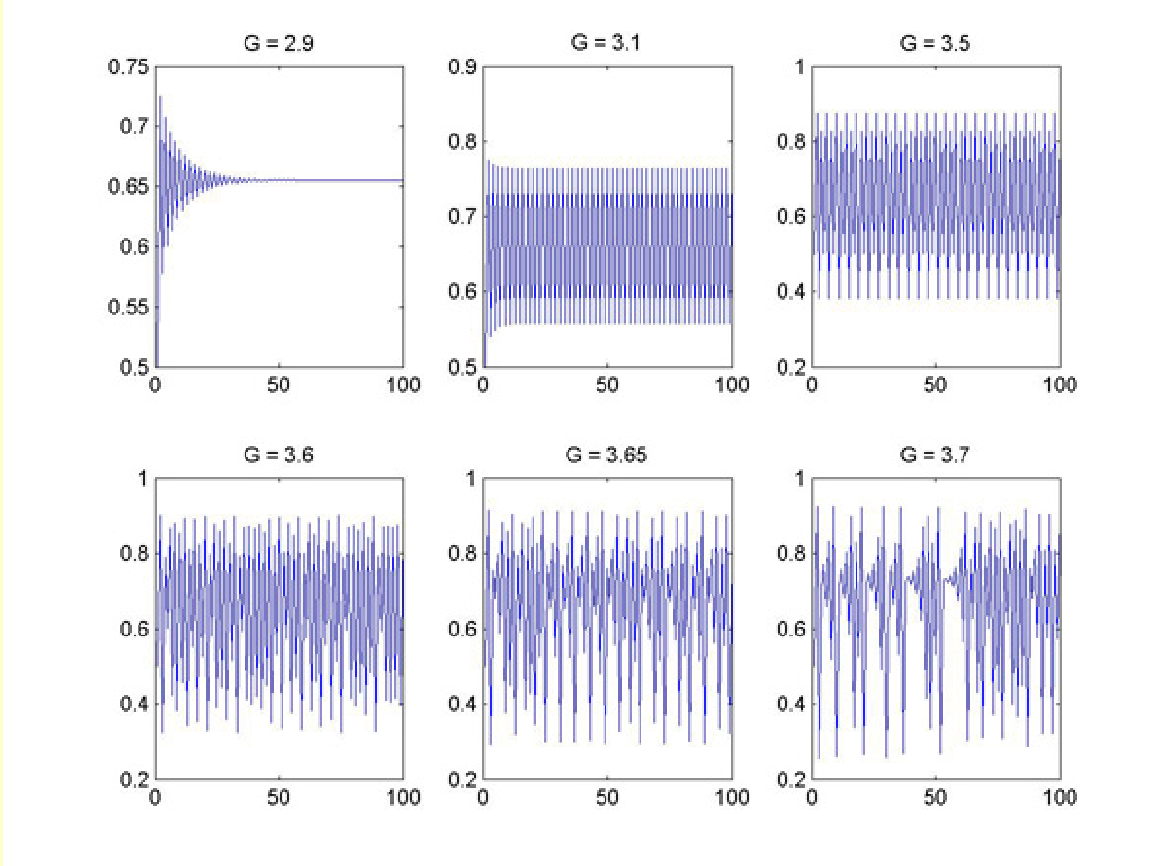
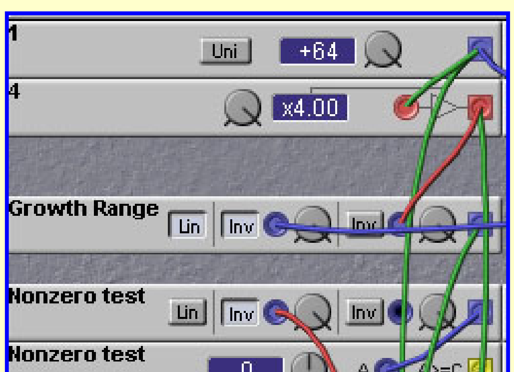
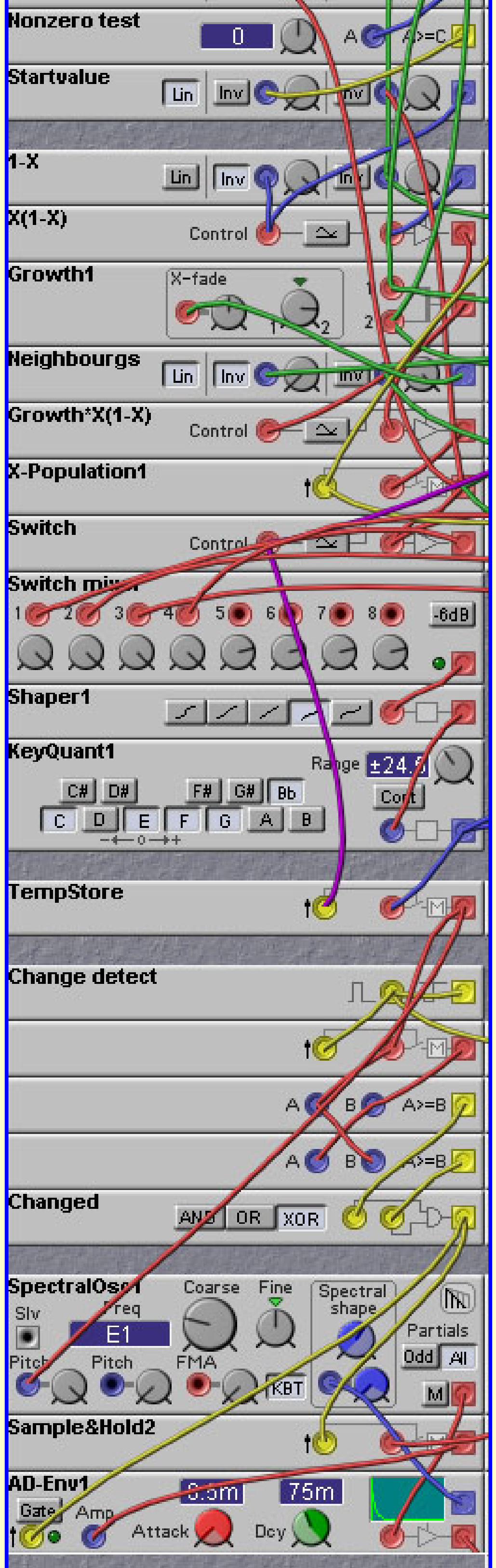


Figure 10.1. The behaviour of the logistic equation for various settings of the G parameter. For G less than 3, the system quickly settles to a fixed value. For G between 3 and 3.53 oscillations occur. For G greater than 3.53 chaotic dynamics result.

In the figure below we show 1/4 of the entire patch, corresponding to one of the four chaotic systems (clicking on the figure will give the entire patch, however).




Nonzero test
0
A
A>=C
Startvalue
Lin
Inv
1-X
Lin
Inv
X(1-X)
Control
Growth1
X-fade
1
2
Neighbourgs
Lin
Inv
Growth*X(1-X)
Control
X-Population1
Switch
Control
1
2
3
4
5
6
7
8
-6dB
Shaper1
KeyQuant1
Range
C#
D#
F#
G#
Bb
C
D
E
F
G
A
B
TempStore
Change detect
A
B
A>=B
A
B
A>=B
Changed
AND
OR
XOR
SpectralOsc1
Coarse
Fine
Spectral
shape
Slv
Freq
E1
Partials
Odd
All
Pitch
Pitch
FMA
KBT
M
Sample&Hold2
AD-Env1
75m
Gate
Amp
Attack
Dcy

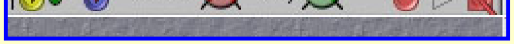


Figure 10.2. Chaotic system patch using the Feigenbaum attractor (R. Hordijk).

For the equation to work it can not start with a zero value for X, as nothing can grow out of nothing unless some "Act of God" is involved. So a provision is made that a small value is automatically added to X, only in the first cycle when X is zero due to patch loading conditions.

The following patch implements another type of Chaotic system, this one termed the Henon strange attractor. This system was discovered by the French astronomer Michel Henon while studying the dynamics of stars moving within galaxies. The equations for this system are:

$$x(n+1) = 1 + y(n) - a*x(n)*x(n)$$
$$y(n+1) = b*x(n)$$

The Henon system is 2-dimensional, unlike the logistic system described earlier, which was 1-dimensional. That means it produces two values which you can use to control various aspects of the music. As with the logistic equation, certain settings of the parameters a and b give rise to chaotic behaviour. An example is shown in figure 10.3, which plots the values of x and y for a setting of a=1.4, b=0.3. Notice that the points are distributed along a smooth curve, so that overall pattern of the values of x and y is appears non-random, and repeatable. At any given time, however, the values are essentially randomly distributed along this curve. Thus we have a "structured randomness", just the thing we are looking for in algorithmic music!

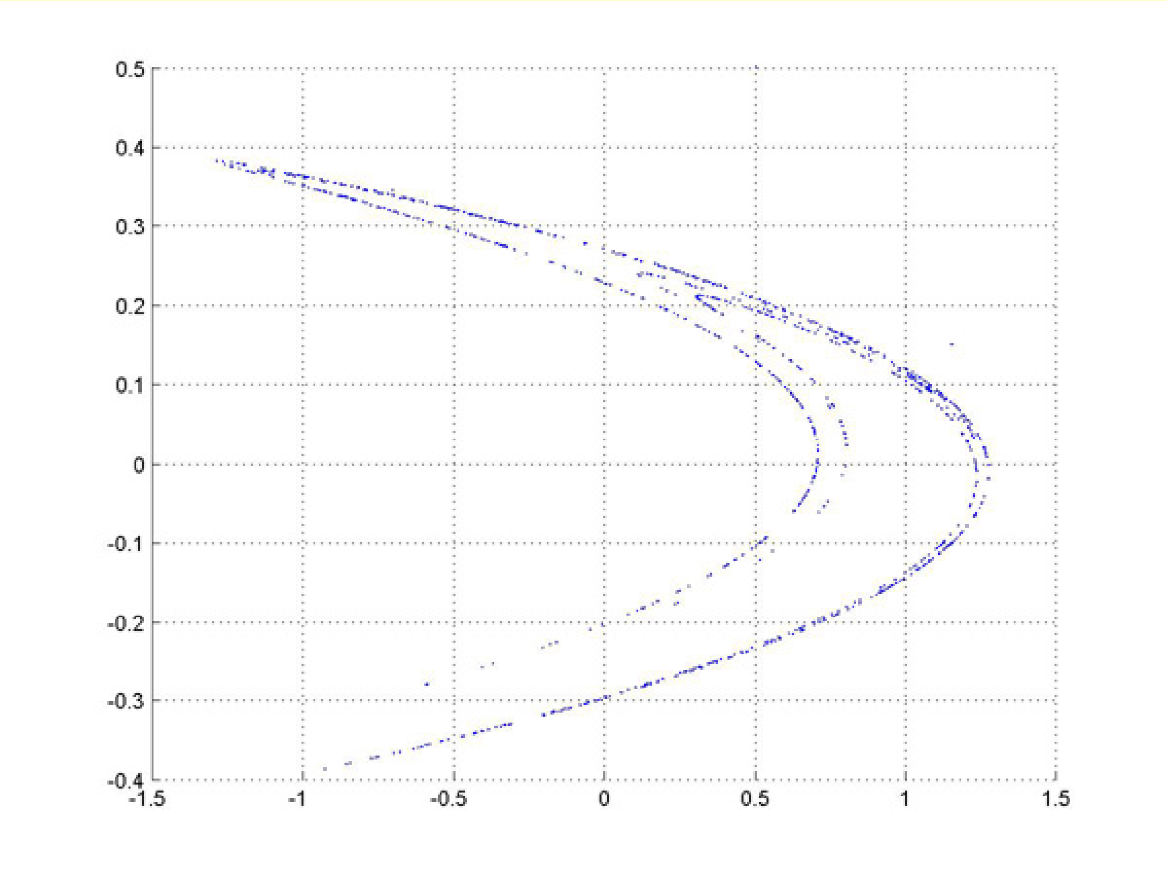


Figure 10.3. Behaviour of the Henon attractor for a setting of a=1.4, b=0.3.

The following patch uses the Henon equations to generate a quasi-periodic melody (in Eb major, but you can change this by changing the notes selected in the key quantizers).

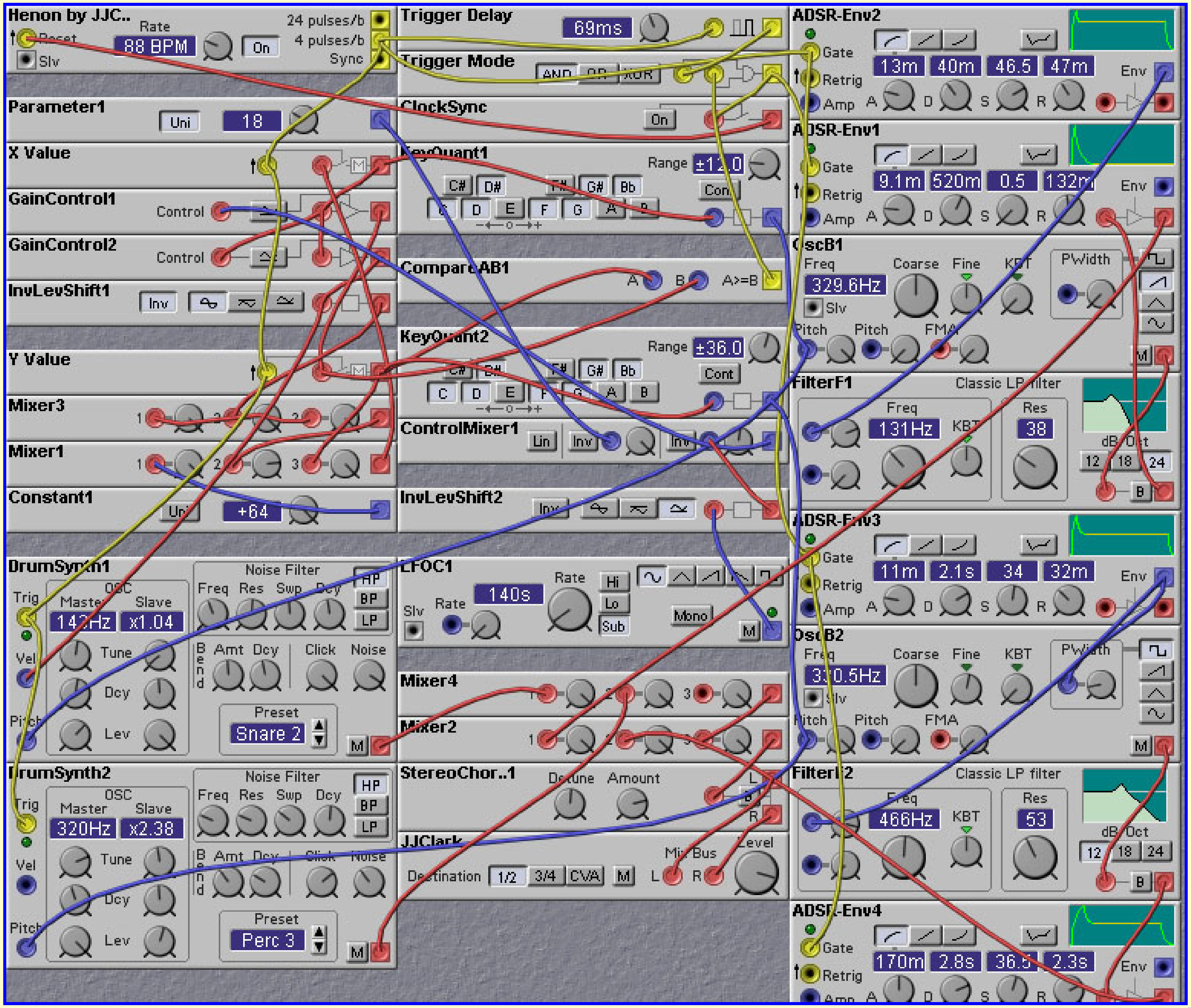


Figure 10.4. A chaotic noodle patch using the Henon strange attractor to vary the pitches and rhythms (J. Clark).

Knob 1 controls the dynamics, and determines whether the pattern is repetitive or chaotic.
Knob 2 controls the level of a LFO that adjusts the attractor parameter (Knob 1) to provide a constantly (slowly changing) dynamic. Turn knob 2 to zero to maintain a steady attractor type.
Knob 3 sets the tempo.

Knob 4 sets the delay in the triggering of the second voice.
Knob 5 is used to turn on or off the syncing of the master clock with the second voice trigger. Turning this on causes a change in the beat pattern.
Knob 6 is used to select one of three different modes for generating the second voice trigger.
You can try to play along with the pattern with your piano instead of doing all those tedious Hanon piano exercises!

There is a wealth of information on Chaos on the web and in the libraries. A good starting point would be to read the very accessible book "Chaos: making a new science" from James Gleick. It's a good introduction into chaotic behaviour.

The chaotic systems that we have talked about so far are of a class known as iterative systems, in which an operation is repeatedly applied in discrete times. One can also obtain chaotic behavior in continuous dynamical systems. These are much harder to analyze mathematically, but they are just as easy to use. While there is a great variety of dynamical systems that exhibit chaos, from the point of view of modular music synthesis one of the most useful is the coupling of three voltage controlled oscillators, connected in a ring. The idea is to have oscillator 1 frequency modulate oscillator 2, oscillator 2 frequency modulate oscillator 3, and oscillator 3 being fed back to frequency modulate oscillator 1. Depending on the modulation amount settings, such a circuit produces chaotic outputs for each of the oscillators. This approach can be used for audio-rate oscillators as well as for LFOs. The audio-rate circuit produces sounds that sound like (rather sickly) noise, which may be useful for some percussive sounds or simulating broken steam radiators. For algorithmic synthesis, however, chaotic LFO chains are much more useful. The outputs of the LFOs can be used to control sound parameters (pitch, filter cutoff, amplitude, etc) or to trigger events. An example is shown in the following patch:

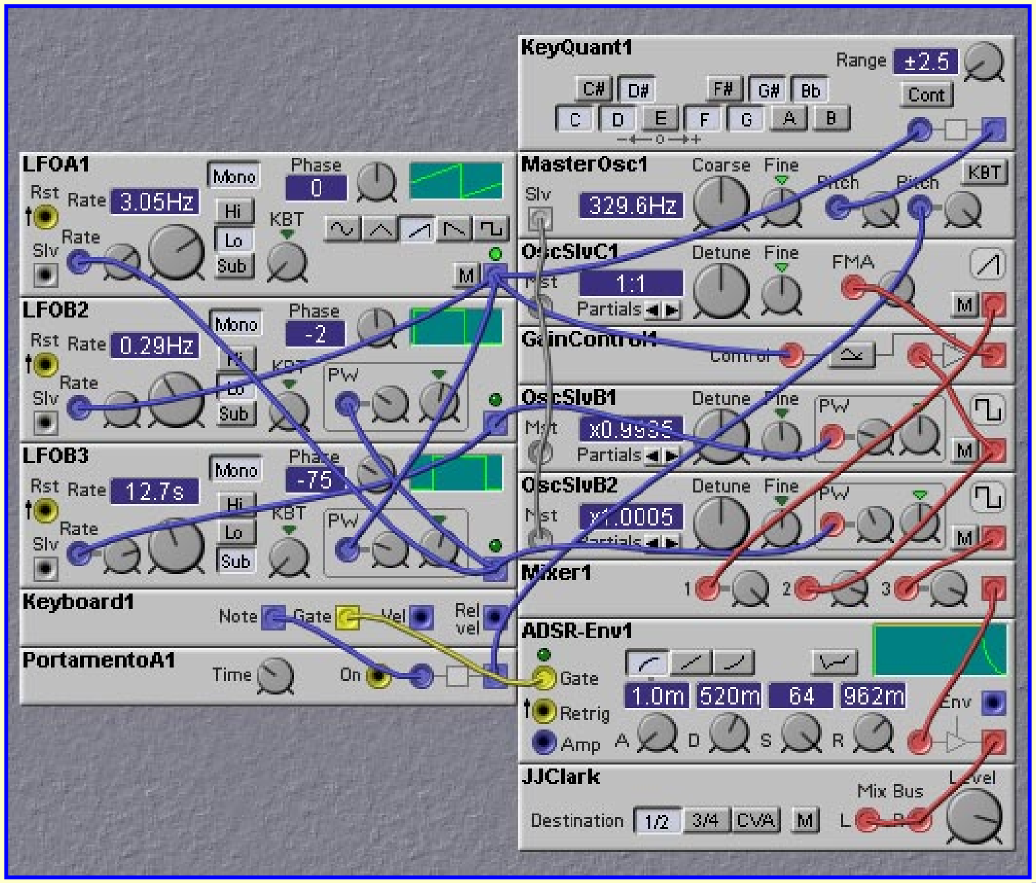


Figure 10.5. A chaotic noodle patch using 3 coupled LFOs (J. Clark).

In this patch the outputs of the LFOs are used to control (through a note quantizer) the note pitches, the pulse-widths of 2 audio oscillators, and the FM amount for one audio oscillator. Thus the pitch seems to warble in a quasi-random fashion, as does the timbre of the notes (due to the time-varying PWM and FM).

10.3 Cellular Automata

Cellular automata are instances of Finite State Machines, where the state of the system is represented as binary values in a set of cells. The cell values are updated regularly based on the values in neighboring cells, according to some rule. Perhaps the most famous example of a cellular automaton is that used in the "game of life", devised by Princeton mathematician John Conway.

In a "game of life" cellular automaton each cell carries a binary (0 or 1) value. If the value is 0 the cell is said to be "dead", otherwise if its value is "1", the cell is said to be "alive". Updates of cell values are based on the number of "living" neighbors. A typical rule for the updating process is to set a cell to the "alive" state is the number of living neighbors is less than a certain high threshold (not too crowded) and greater than a certain low threshold (not too lonely). Otherwise the cell is set to the "dead" state. For certain choices of the thresholds and initial population patterns, dynamic and persistent behaviour can arise in the state values of the overall cellular automata.

An example of the evolution of a simple 1-dimensional cellular automaton is shown in figure 10.5. The behaviour of the automata is set by a rule which maps the local state (the value of a cell and those of its neighbors) into a new state value. The rule can be described in a look-up table that gives a value of either 0 or 1 for every possible combination of neighborhood cell values. The automaton shown in figure 10.5 has 2 neighbors for each cell. The complexity that can be obtained from such simple rules is clearly evident.

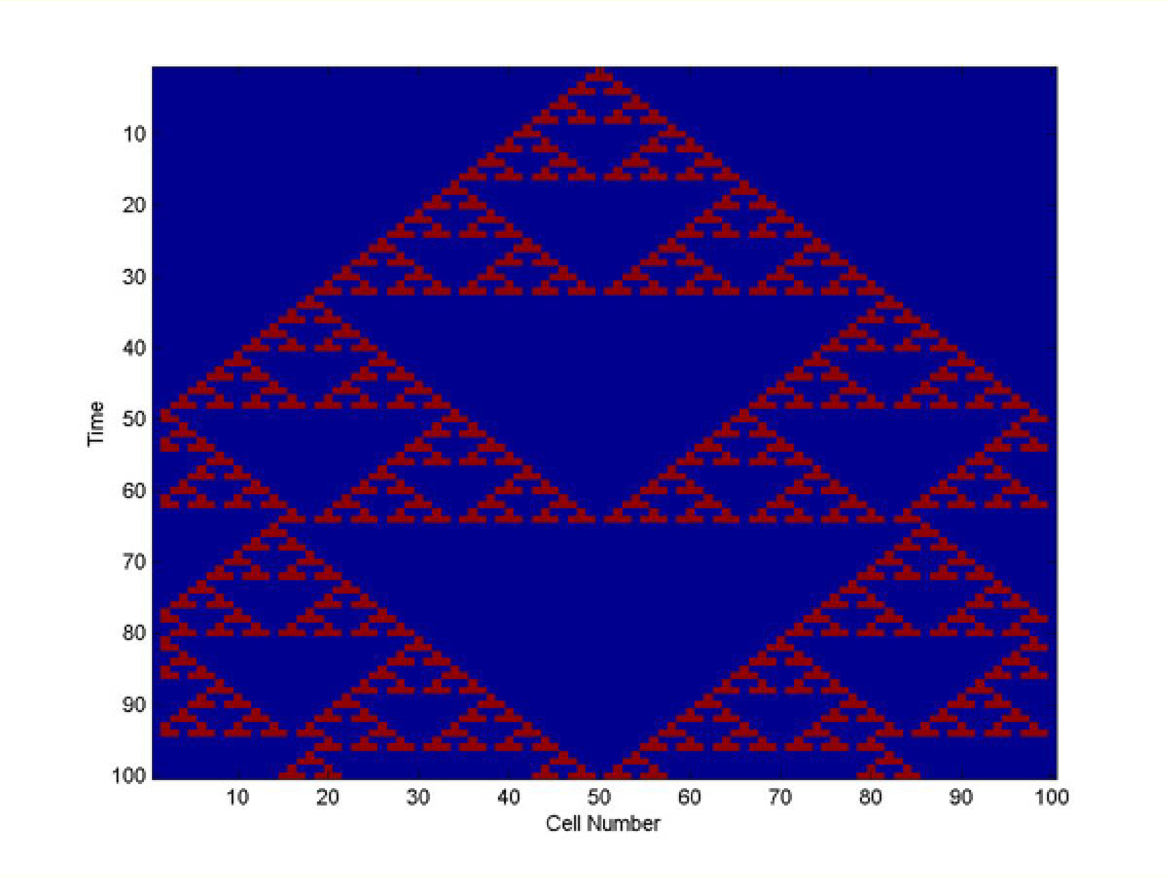


Figure 10.6. An example of the patterns generated by a 1-dimensional cellular automaton. The vertical dimension represents time and the horizontal dimension corresponds to the location of the cells. There are 2-neighbors for each cell.

If we implement a cellular automata in the Nord Modular we can use cell state values to control various parameters and generate interesting (?) music. The following patch illustrates the implementation of a simple Cellular Automaton. It implements a 1-D array, with each cell have 2 neighbors. A "Life" type of rule is used, where a cell state is set to one if there is only one active cell in its neighborhood (which includes itself). This is basically the same rule that was used to generate the pattern shown in figure 10.5. The cell state is stored in a Sample-Hold module and cells are updated whenever the Sample-Hold modules are clocked. The update values are computed by the logic gates attached to the S/H outputs. In this case the logic merely determines when there is exactly one logically high output in a neighborhood of three S/H outputs. The patch implements 6 cells, connected in a circular ring. The outputs of each cell are used to trigger an envelope generator. Of course, you could use these outputs for anything. I used them to trigger the envelope generators since the gate LEDs on the envelope generators gives a visual indication of the cellular automata states. The keyboard gate is used to briefly set one of the cells states to one. For the particular rule that is used, the cellular automaton will go through 4 active states and then stick and hold on the "all-zeroes" state. By changing the logic circuits that are used to update the state you can get different patterns.

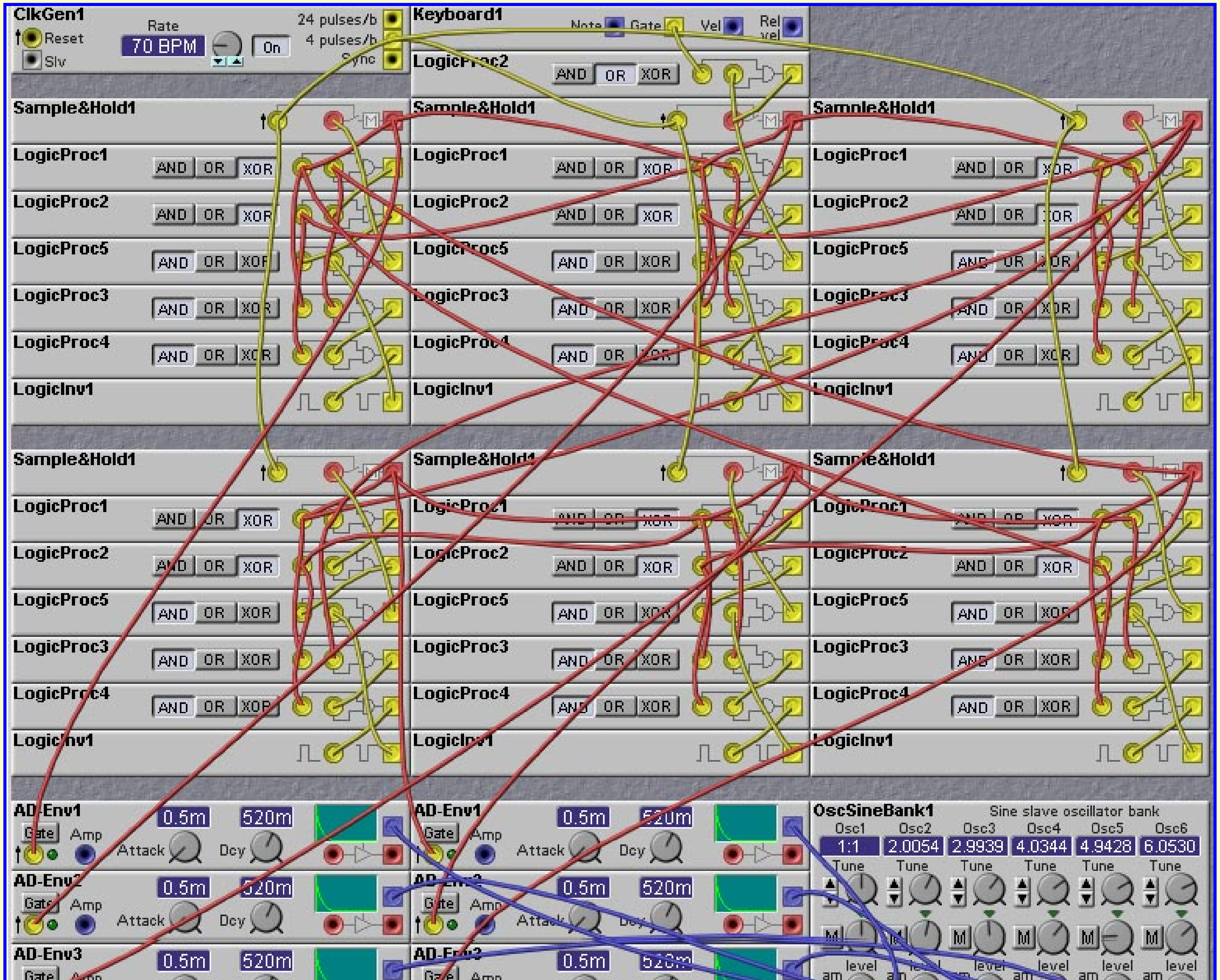
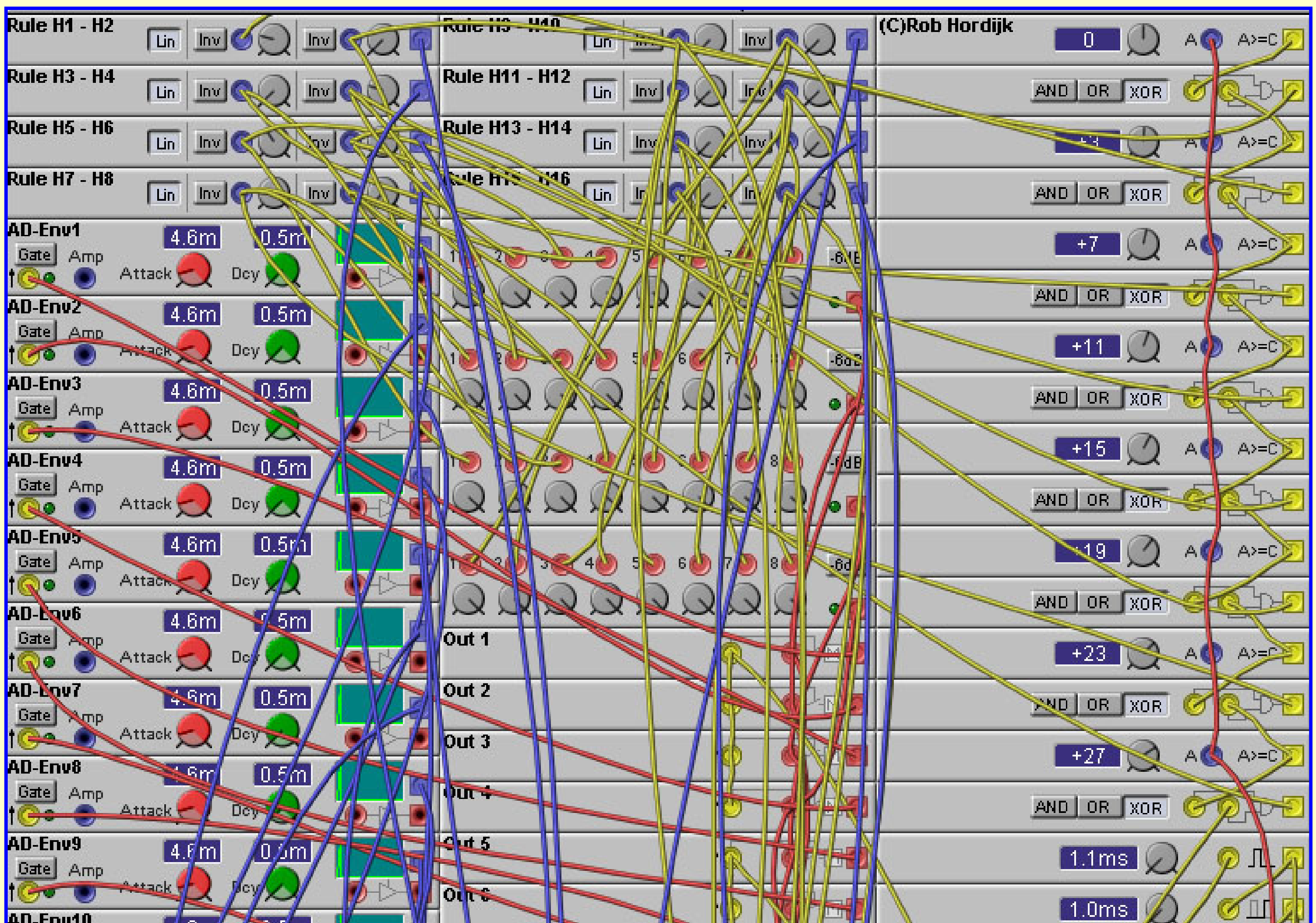


Figure 10.7. A simple 1-D cellular automata patch for the Nord Modular (J. Clark).

The cellular automata patch shown above uses logic gates to implement the state updating. This approach is rather inflexible, as changing the update rule requires changing the logic circuitry. Thus, one can't change rules on the fly, as it were. In most software implementations of cellular automata the updating is done via a look-up table, wherein the neighborhood state is used to index a look-up table memory which stores the appropriate update value. Changing the update procedure simply requires changing the values in the look-up table memory, something which could be done on the fly. Unfortunately, the Nord Modular does not contain any random-access, re-programmable memory modules that could be used for a look-up table. One could cobble together something out of sample-hold modules, but it would be very cumbersome. So, what to do? How can we make a programmable cellular automaton on the Nord Modular? Well, the always resourceful *Rob Hordijk* came up with a solution, which uses the knobs on the front panel of the Nord Modular as the programmable lookup table! His implementation is shown in the following figure. It doesn't immediately look like a cellular automaton, as the neighborhood structure is hidden away, but the overall behaviour is that of a cellular automaton.



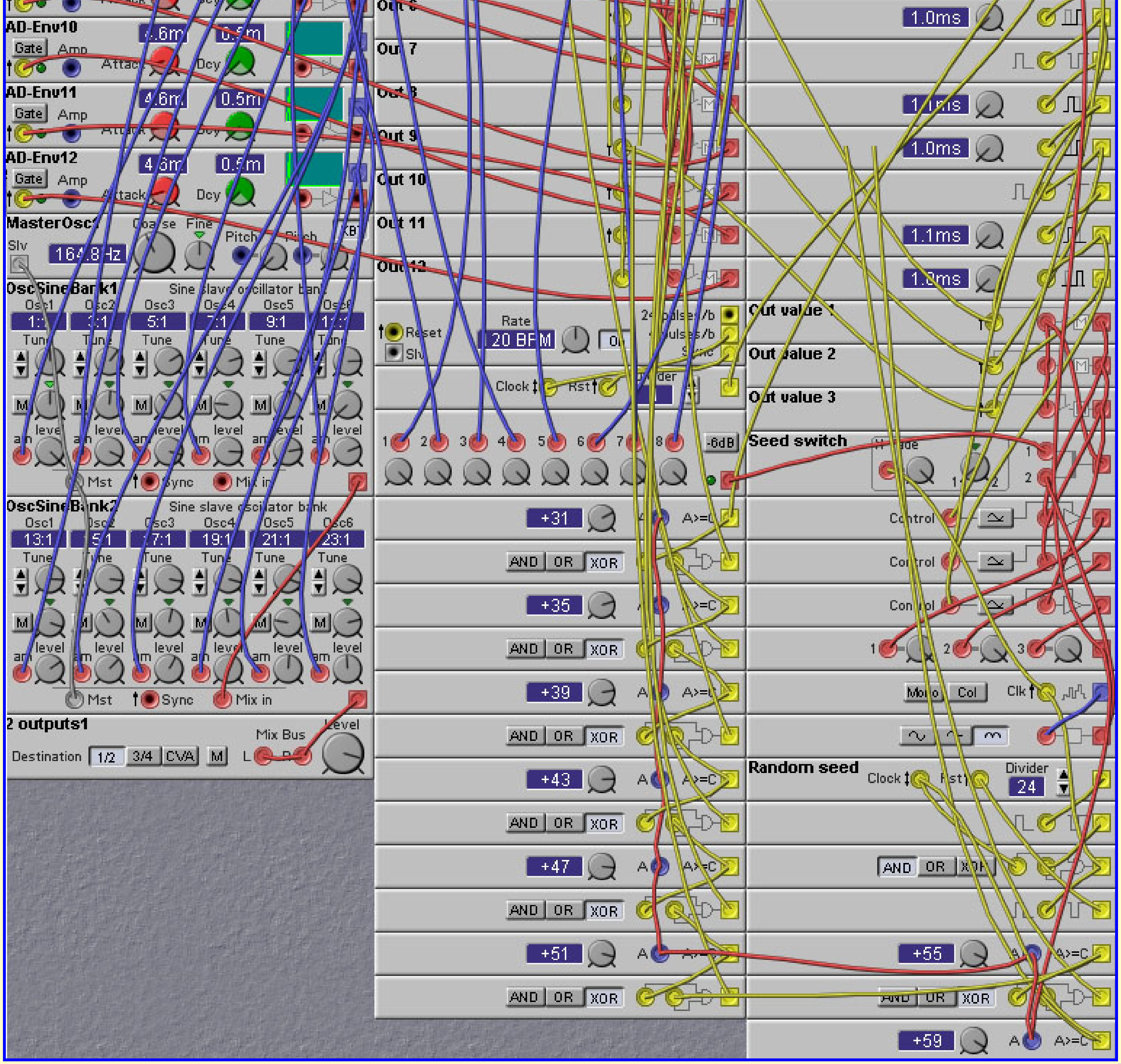


Figure 10.8. A cellular automata patch for the Nord Modular (R. Hordijk).

In this patch, four-bit 'nibbles', implicitly representing a neighborhood state, are used to address sixteen memory locations. These locations themselves each hold a nibble, which represent the updated neighborhood values. These locations are examined (looked up) on every clockpulse, to be used as the address in the next iteration. There are three usable nibbles in this patch, each with the same set of 'rules', that is they share the same lookup memory locations. The contents of the sixteen memory locations are in practise the sixteen knob values.

The 'nibbles' are conveniently stored in S&H modules. The nibbles are decoded using one-of-sixteen circuits (the group of compare and XOR modules) and used to address one of the sixteen knobs. Each knob delivers a new 'nibble' to store in the S&H on the next clockpulse. Also the sixteen outputs of the one-of-sixteen circuit are coded back to the real four bits of a nibble, using so called 'mickey mouse'-logic (the four eight-input mixers). We now have three types of output to work with, (1) the sixteen separate logic outputs of the XOR-modules, only one of these can be on, (2) the three S&H's that can have sixteen values between 0 and 64, in steps of 4, and (3) the four discrete bits that form the 'nibble' at the outputs of the eight-input mixers, whose values must be interpreted as 0 or anything greater than 0. What to do with the output values is up to you. The patch shown above uses the outputs to trigger envelope generators attached to sine wave oscillators of various frequencies, thereby producing different melodies.

To program a pattern in this patch follow these rules:

If knob1 is closed the next selected knob will be knob1 again. If knob1 points to knob2, that one will be the next knob and if that one points to knob5, which on it's turn points to knob1, the pattern will be knob1 -> knob2 -> knob5 -> knob1 -> knob2 -> knob5 -> knob1, etc. So here repetition occurs after three clockpulses, but in general the longest pattern will be sixteen clocks. So to make it more interesting knob16 is replaced by a randomvalue. If in a pattern knob16 is reached it jumps randomly to one of the other knobs. Now it starts to get interesting as in the example the patterns would never use knob3 and knob4 and all knobs after knob5. But if knob3 points to knob4 and knob4 points to knob16, and we will also let knob5 point to knob16 instead of knob1, and all other knobs point to either knob1 or knob 3, the pattern goes random, but not completely random, it starts to randomly play the small fragments all ending with knob16, sometimes preceded by a randomly chosen knob and sometimes jumping into the middle of the first two fragments. There is one 'disallowed' state in the knobsettings, if a knob points to itself the pattern 'hangs' on that knob -> the 'population' sadly dies and it's game over.

On loading the patch in a NM slot, the S&H modules will be initialised with values of zero, so the three nibbles all point to knob1. But for every nibble, when knob16 is reached a different random value for each nibble is generated, so the combination of the three nibbles gives a more varied pattern than only one nibble, but still fragments will be more or less recognizable according to the knobsettings. So knob16 can actually be deassigned in this particular patch, as tweaking it doesn't do anything since it is replaced by a random value.

The 'rule'-values or 'to what value should a knob be set to make it jump to which other one' are:

Jump to Knob1: 0-7
Jump to Knob2: 8-15
Jump to Knob3: 16-23
Jump to Knob4: 24-31
Jump to Knob5: 32-39
Jump to Knob6: 40-47
Jump to Knob7: 48-55
Jump to Knob8: 56-63
Jump to Knob9: 64-71
Jump to Knob10: 72-79
Jump to Knob11: 80-87
Jump to Knob12: 88-95
Jump to Knob13: 96-103
Jump to Knob14: 104-111
Jump to Knob15: 112-119
Jump to Knob16: 120-127

10.4 Cooking Noodles

One of the favorite pastimes of Nord Modular programmers is the "cooking of noodles". Noodles refer to self-running patches that have some (hopefully interesting) temporal variation in the sound. Some noodles are melodic, some are extended "drones", where the variation is timbral in nature, and some are just plain wierd.

In this section I will try to give some tips for creating your own noodles, but these should only be considered as examples, and shouldn't limit your own creative explorations!.

First off, you should decide what sort of noodle you want to create. That is, do you want a melodic noodle? Do you want a rythmic noodle? Do you want a timbral noodle? A combination of these? None of the above? This will tell you what type of parameters will be time-varying. Time-variation is the key in cooking noodles. As Jan Punter, one of the master noodle chefs for the Nord Modular, says, noodle makers should concentrate on the "blue wires" in the Nord Modular patch. That is, the control structures are paramount. The actual sound generation is less important, although of course you do want to have interesting sounds for the most part.

You will need to have some mechanism(s) for producing the temporal variations in the control signals. The most basic ways of doing this are to use LFOs, Sample-Hold modules sampling noise, the Sequencer modules, and the Random generators. By themselves, these modules give uninteresting noodles, as they are either too static (e.g. with the sequencer or LFO modules) or too random (e.g. with the random generators or with sample-hold noise). Such noodles sound stereotyped and trite. Rise above the masses! More interesting variation can be obtained using the chaotic control signal generation techniques described earlier in this chapter.

Some slow variation is possible by working with the audio-chain as well. Feedback, as we have seen in the case of three coupled oscillators, can give rise to chaotic behavior. This is not limited to feedback to frequency modulation inputs, however. Chaotic and unpredictable behaviour can result from all sorts of feedback, whether it be amplitude modulation feedback, filter cutoff modulation, etc. Especially good for creating unpredictable havoc are the delay module (inserted into feedback paths or use feedback to control delay times) and feedback of audio signals to sync inputs on oscillators. These provide very interesting timbral variations that seem to occur in bursts.

Another useful technique is to use crossfade modules to dynamically select different control signals or audio pathways. The crossfade modules themselves can be controlled by chaotic or semi-periodic signals to give noodles that are locally quite well-behaved but from time-to-time change character abruptly. Logic circuits can be used to control the crossfaders as well, for example using counters, flip-flops or even event sequencers, to control the evolution of the noodle. Sample-Hold modules can be used to "freeze" certain control signals for a time.

Experiment with every module and with every type of parameter. For example, I find that using the AHD and MOD envelope modules can be very effective in noodling, since they have modulatable timing parameters. The digitizer module also finds its way into many noodles. Noodle cookers should have an aversion to leaving any input unconnected. An unconnected input is a wasted opportunity for interesting variation!

Most good noodles have a certain "feel" to them. Some are jazzy, some bluesy, and others have a middle-eastern feel. You need to keep tight control over such noodles, to ensure that they don't stray from their intended groove. To do this, use mixers and attenuators to keep levels in a certain range, and employ quantizers. Note quantizers can be used to impose scales (e.g. the blues scale) on a noodle. Digitizer modules can be used to limit the range of control signal levels. The following patch shows a noodle that has a definite feel to it, that of an aging 70's prog-rocker aimlessly noodling away on his synth. This patch is an expansion of the patch shown in figure 10.5. A chain of 3 coupled LFOs is used to generate three chaotic control signals, which are used to control the pitch and timbre of the notes. One of the chaotic control signals is used to trigger the amplitude envelope, thereby impose a quasi-random rythm. The 70's synth feel is created through the use of portamento and resonant filtering, as well as by the imposition of a minor scale through use of a note quantizer.

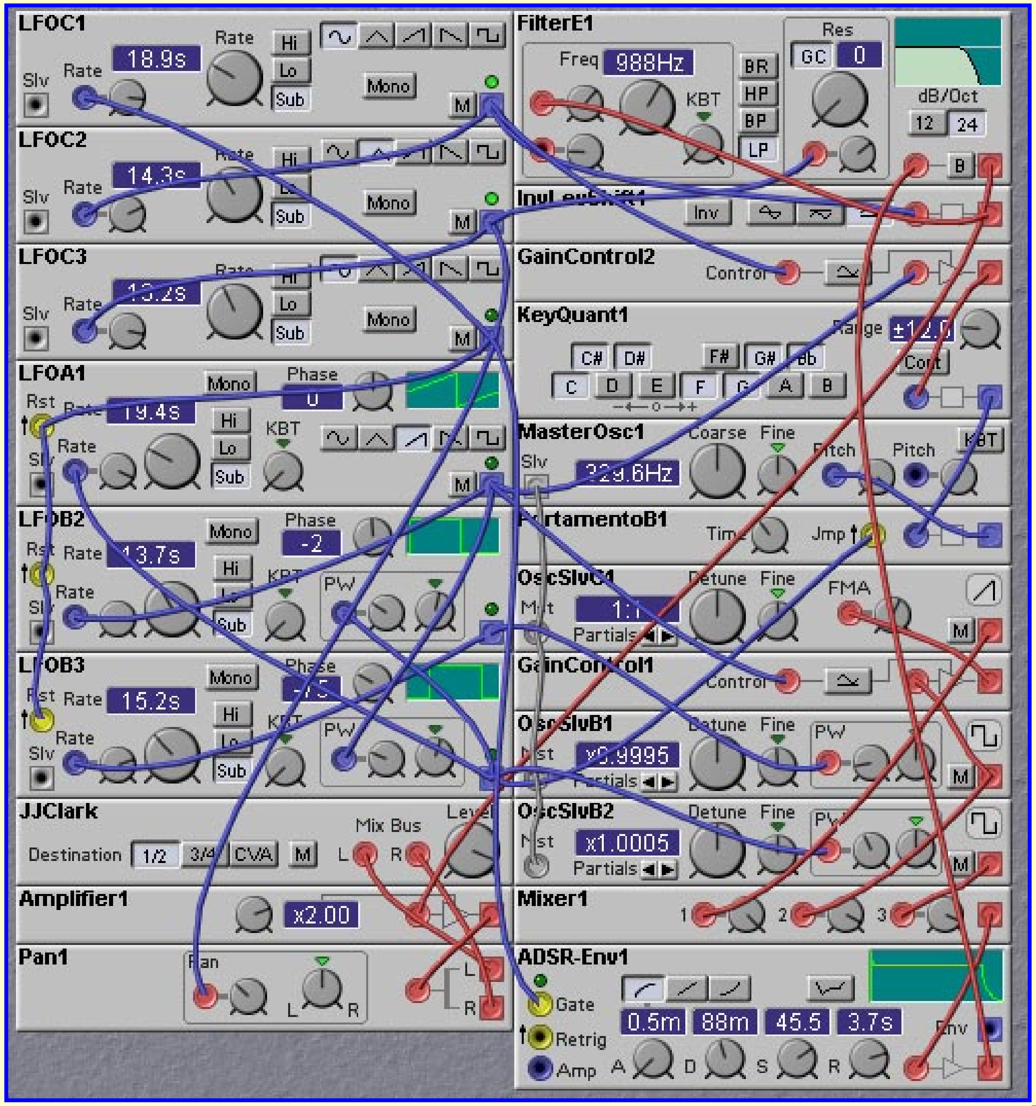


Figure 10.9. A chaotic noodle patch using 3 coupled LFOs. The sound is designed to give the feel of a 70's prog-rocker aimlessly noodling away on his synth keyboard (J. Clark).

One of my most frequently employed methods for getting interesting noodles is to take somebody else's posted noodle patch and "mod" it (modify, hack, deconstruct,...) to yield something completely different (yes, I know, I am lazy). I usually do this by re-wiring some of the control signals, changing some of the parameters (e.g. cutoff frequencies, osc frequencies, etc), adding in osc sync, and inserting feedback paths. If there are sequencer modules in the original patch their settings should be altered (e.g. melodies and rhythms changed). An example of this is shown in the following patch, which is a mod of K. Lex. Pattyrson's patch KrypixMia.pch. K. Lex (who goes by many on-line aliases) is one of the masters of noodle programming, and it is worth the time to download his patches and give them a listen.

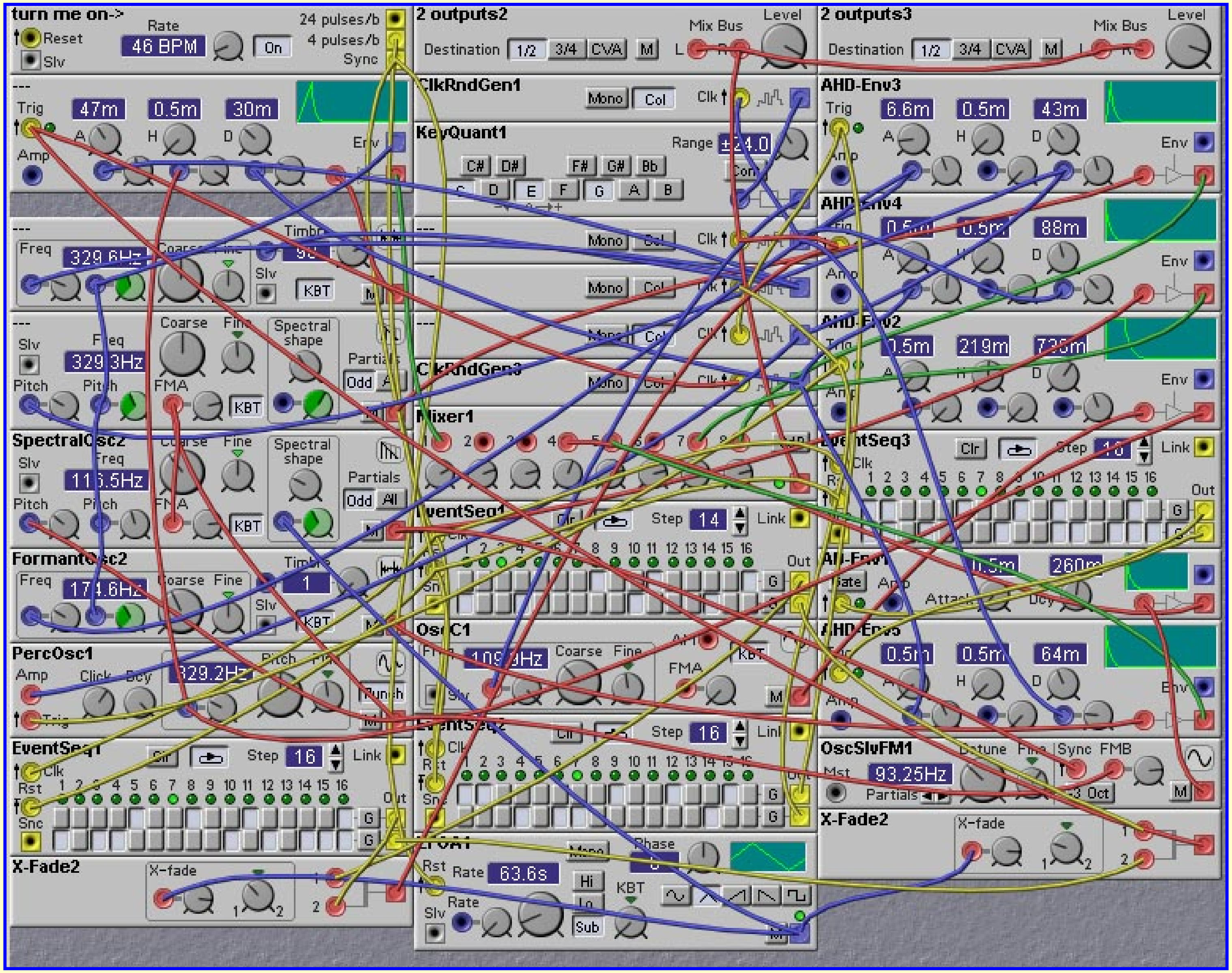


Figure 10.10. A sonically complex noodle patch adapted from another noodle patch (J. Clark from L. Pattyrson).

You can see by the mess of wires in this patch that noodles can be quite complex. In fact this noodle patch is one of the simpler ones out there. Most are even more complex. Even so, the best way to learn how to create your own noodles is to look at other people's noodle and see how they do it. Because of their complexity it can be difficult to understand what is going on. You should try to focus on the control signal structure and see what is modulating what. Also look for feedback, as this can also significantly affect the temporal variations. The sound generation chain itself is not so important from the perspective of the noodle maker.

[Prev (Using the Logic Modules)][Next (Reverb and Echo Effects)][Table of Contents]

Chapter 11. Reverb and Echo Effects

One of the most frequently asked questions regarding the Clavia Nord Modular is how to make a Reverb patch. Well, the short answer is that it cannot be done. The reason for this is that high-quality reverbs and long echoes require delay lines that are much longer than the ones available on the Nord Modular. This is due to memory limitations on the Nord Modular.

So then, what is a poor synthesist to do?? We need our reverbs! There are a number of options. The best option is to use an external effects unit to provide reverb and echo. You can use the analog outputs of the Nord Modular to route signals to the external effects, and can use the analog inputs of the NM to feed the processed signals back in (especially useful with echo effects).

But is it possible to do *something* with the Nord Modular, to produce reverb or echo effects? Well, yes, there are some things that can be done. I will discuss these in the following sections.

11.1 Synthetic Echo and Reverb

The Nord Modular is often used to generate sounds internally, rather than to process external sounds. In this case we can modify the sound generation process to directly produce the required echoes and reverberations. To make this clear, consider the following simple example - suppose we want to generate a simple single oscillator sound, but with an adjustable echo effect. Instead of constructing a patch using a single VCA and envelope generator, with a long delay line, we could use multiple envelope generators (or a single multi-stage envelope) to create the echoes. You could also use a single envelope, but retrigger it repeatedly to produce the echoes. You should decrease the overall amplitude of the sound, perhaps with another envelope, to mimic the decay of the echoes. This technique is used in the following patch (by *Martin Sommerville*). In this patch, an event sequencer is used to specify when the echoes are generated. This allows an irregular spacing of the echoes, giving an effect similar to reverberation.

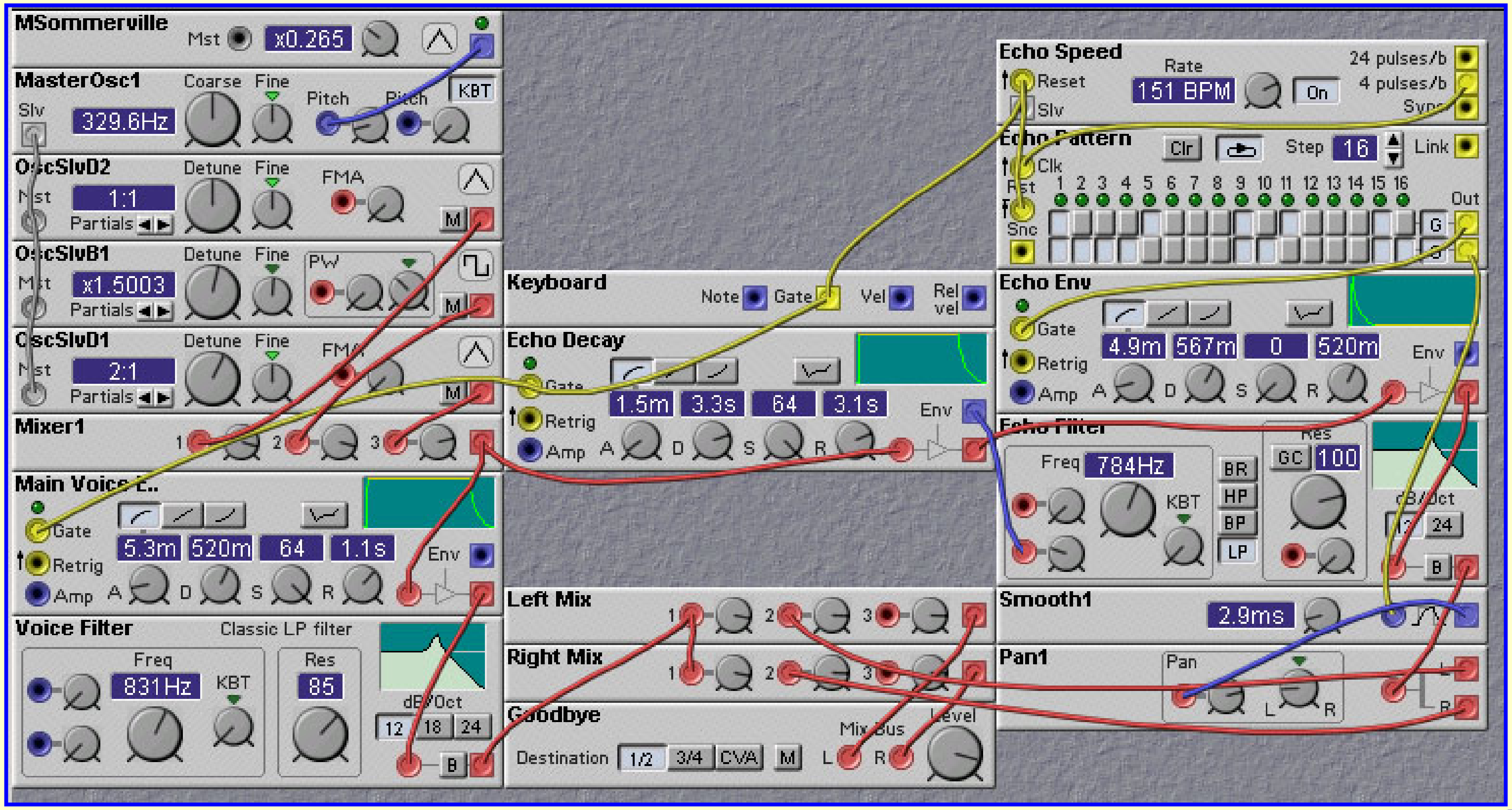


Figure 11.1. Echo patch using a retriggered envelope (M. Sommerville).

Logic delay modules can also be used in creating synthetic echo and reverb sounds. This is done in the following patch, also developed by *Martin Sommerville*. In this patch a set of eight logic delay modules are used to create eight echoes of the keyboard gate signal. These are weighted by the attenuator controls of the mixer module to provide a falloff in amplitude, and frequency content, of the echoes with time. The delayed pulses are used to gate the envelope of the sound.

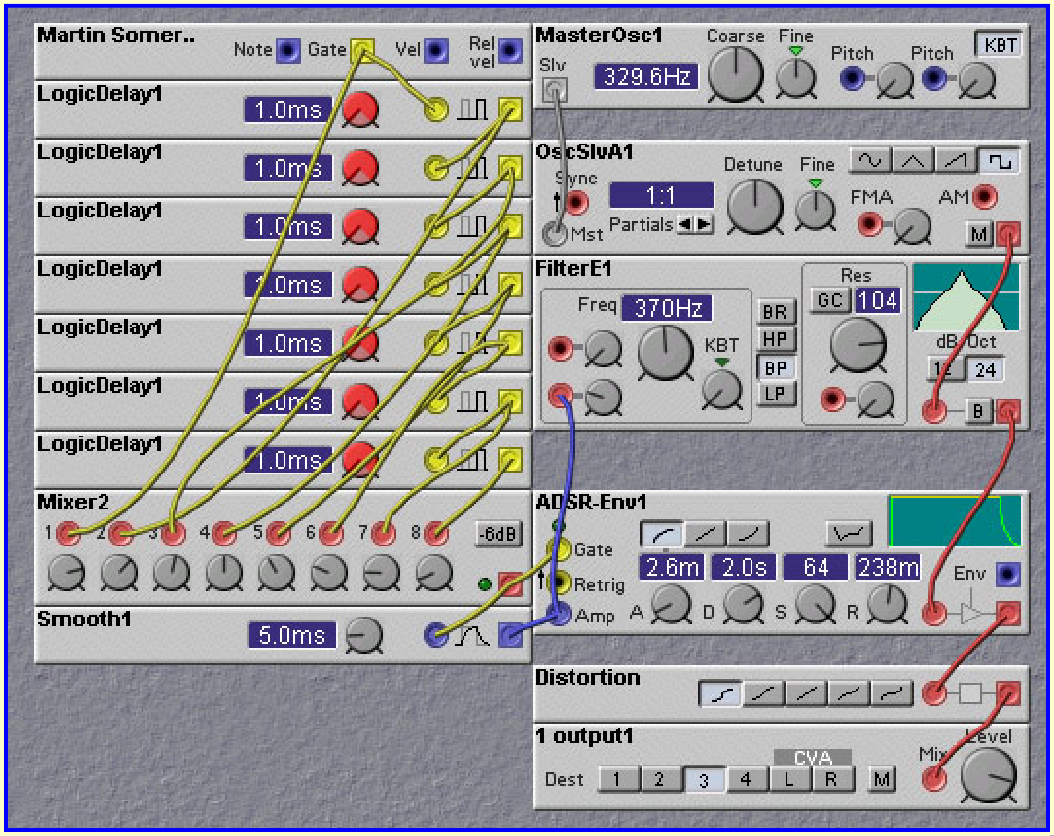


Figure 11.2. Echo patch using logic delay modules (M. Sommerville).

Similar approaches can be used to create synthetic reverb sounds. In a reverb sound one adds echoes of the primary sound. Unlike a standard echo effect, in a reverb effect the echo times become varied (modeling the various pathways the sound takes as it bounces around the walls of the room) and the sound of the individual echoes become duller and duller (modeling the loss of high frequency information each time sound bounces off of a wall). This approach is shown in the next patch.

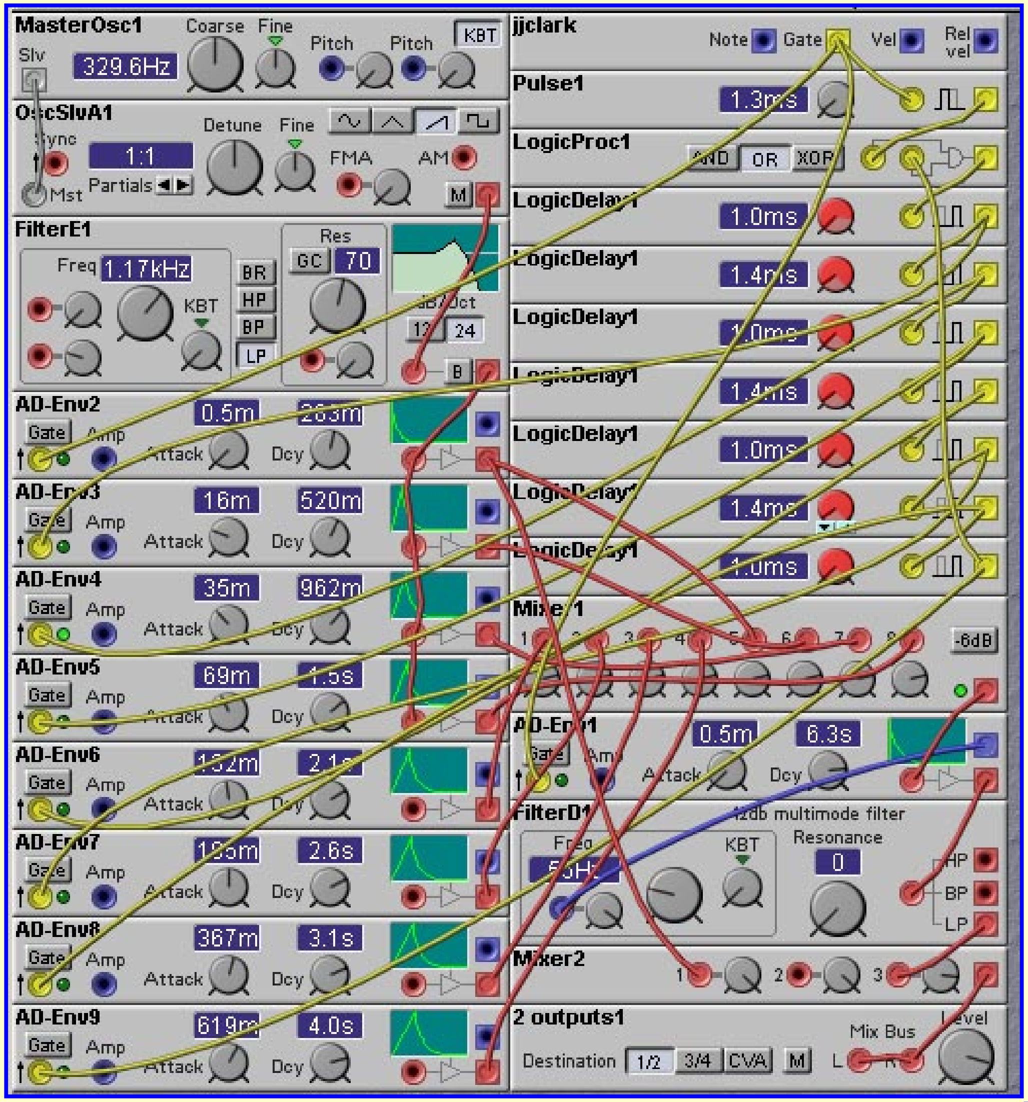


Figure 11.3. Reverb patch using multiple envelopes (J. Clark).

The next patch, by *Rob Hordijk*, is slightly more complicated. It creates a single echo, using a delayed envelope, and then passes the echo through a series of modules which diffuses (or spreads out in time) the echo.

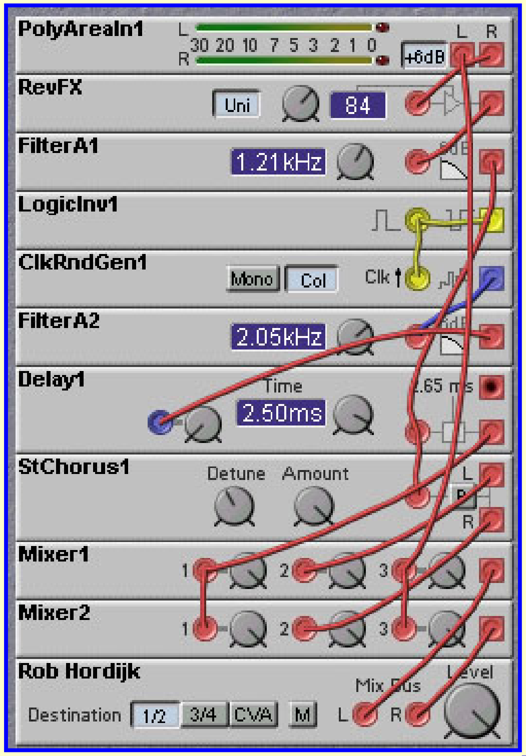


Figure 11.4. Reverb patch using a single diffused echo (R. Hordijk).

The diffusion is done by a delay line whose delay time is modulated by a randomly varying signal. The LogicInv1 module is wired in a positive feedback arrangement, which produces a high frequency square wave which is used to clock the ClkRndGen1 module. This is a cheap way to generate a clock signal. A chorus is also applied to the echo signal and added to the diffused signal. This thickens up the sound some more. Note that the diffusion is done in the Poly area. This means that the diffusion acts on all of the notes being played at the same time. Thus the reverb effect is on the entire sound being played, rather than on the individual notes.

11.2 Short-Time Reverb

One can use the audio delay line modules of the Nord Modular to create echoes and reverb. This is the obvious strategy, but one must remember that the delay line module's maximum delay time is quite short, only 2.65 msec. You can only string 10 of them together without exceeding the Nord Modular's memory limits. Thus it would seem that only a 26.5 msec delayline could be made, much too short for a useable echo effect. This short time can be used to make a reverb effect, albeit one which models being in a *very* small room. An example of this is shown in the following patch, made by *Kaspar Thommen*.

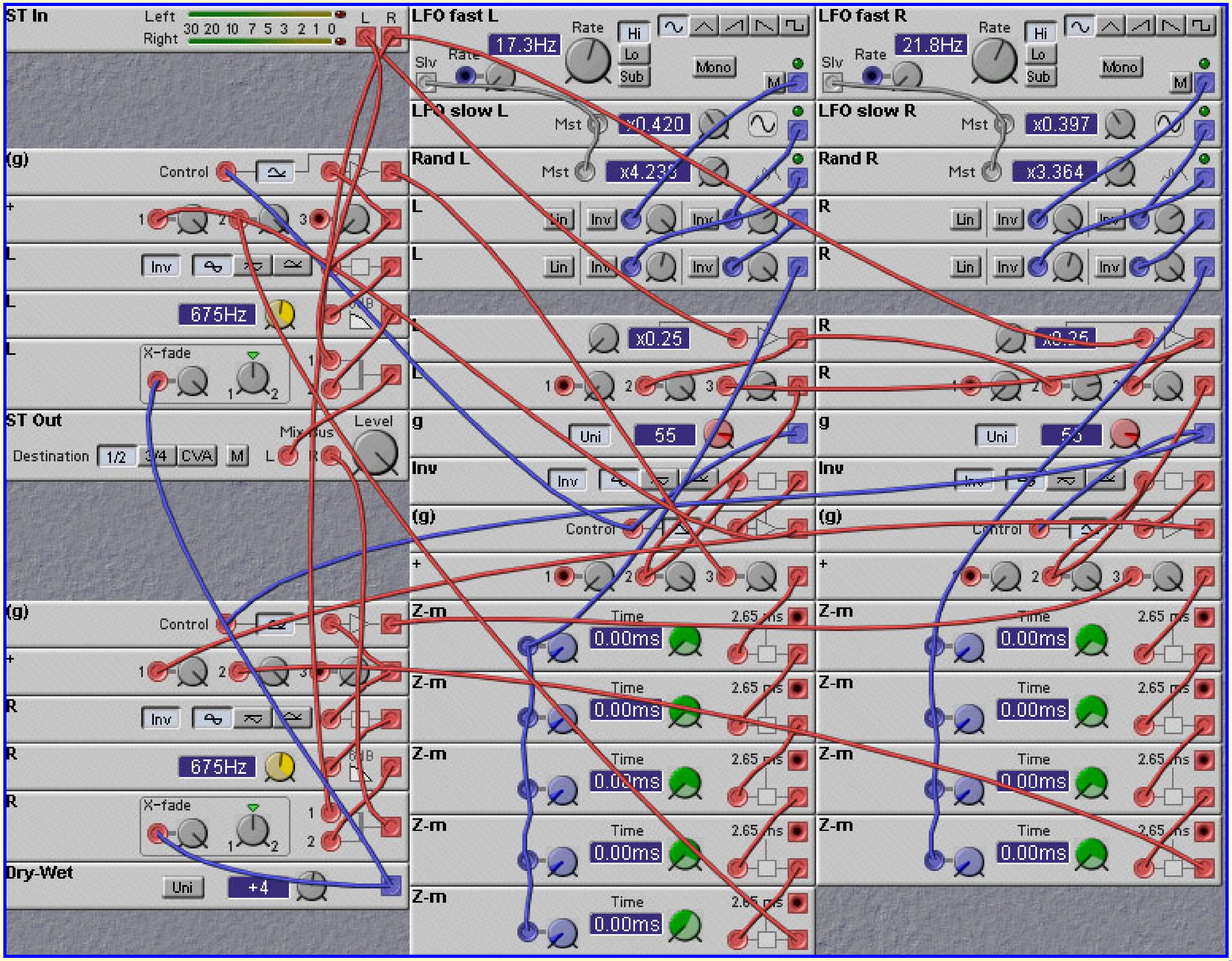
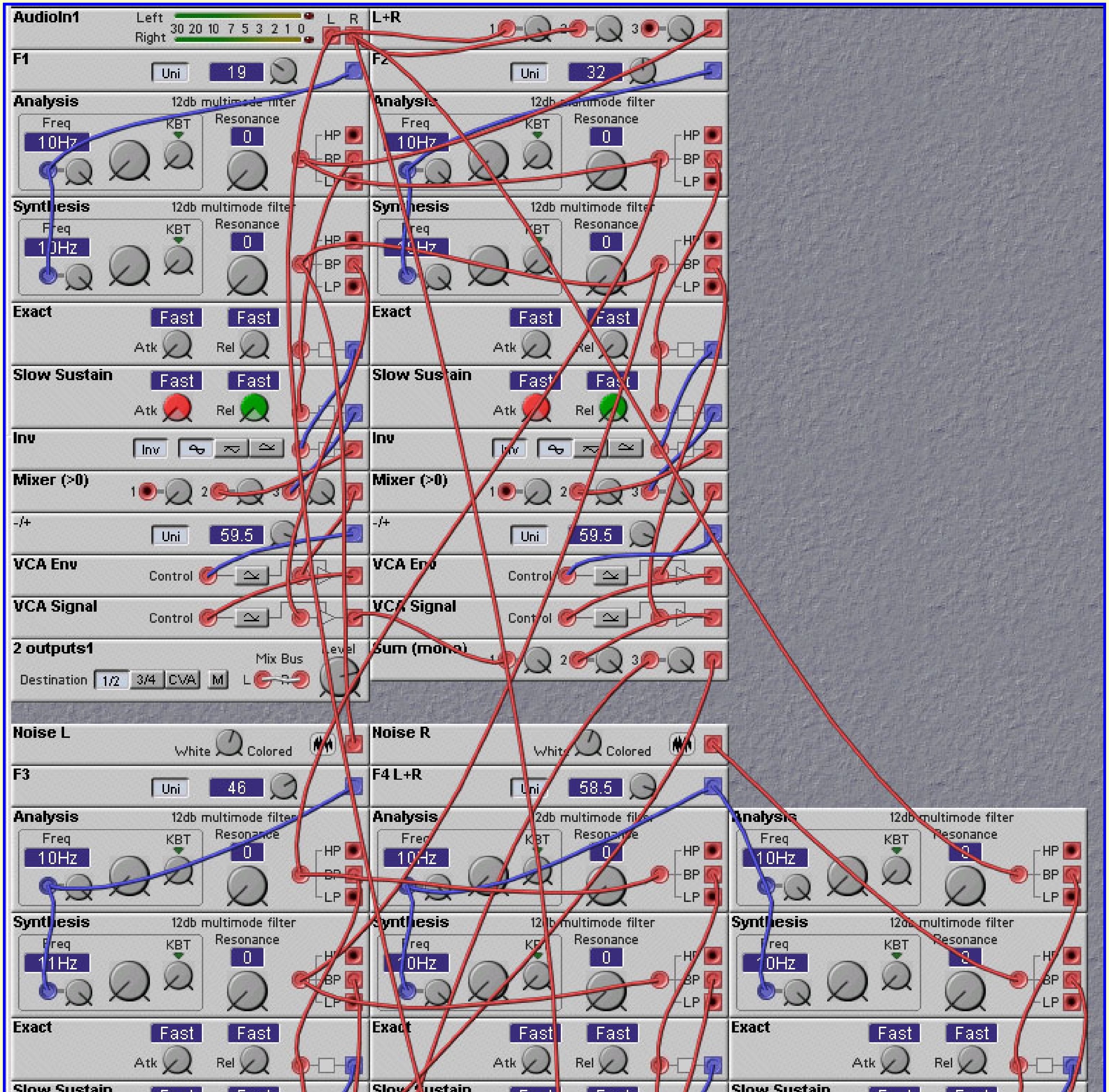


Figure 11.5. A very short-time reverb patch using the audio delay line modules (K. Thommen).

11.3 Low-Fidelity Echo and Reverb

If we are willing to sacrifice a bit (a lot!) of sound quality there are some tricks that can be played to obtain longer delay times with the Nord Modular. These tricks can then be employed to produce low-fi echo and reverb effects.

The following patch, designed by *Kaspar Thommen*, is a clever way to obtain a reverb effect. It basically uses a vocoding process to analyze the input signal, delay the analysis signals with envelope followers, and then resynthesize. This results in delayed versions of the input signal, which can be added to the original input to achieve the reverb effect. The result is not perfect, since the resynthesis of the input is rather crude. But it is still usable for some sounds, like drums. Perhaps if a future update of the Nord Modular software ever provides a vocoder with accessible analysis signals, this patch could be improved.



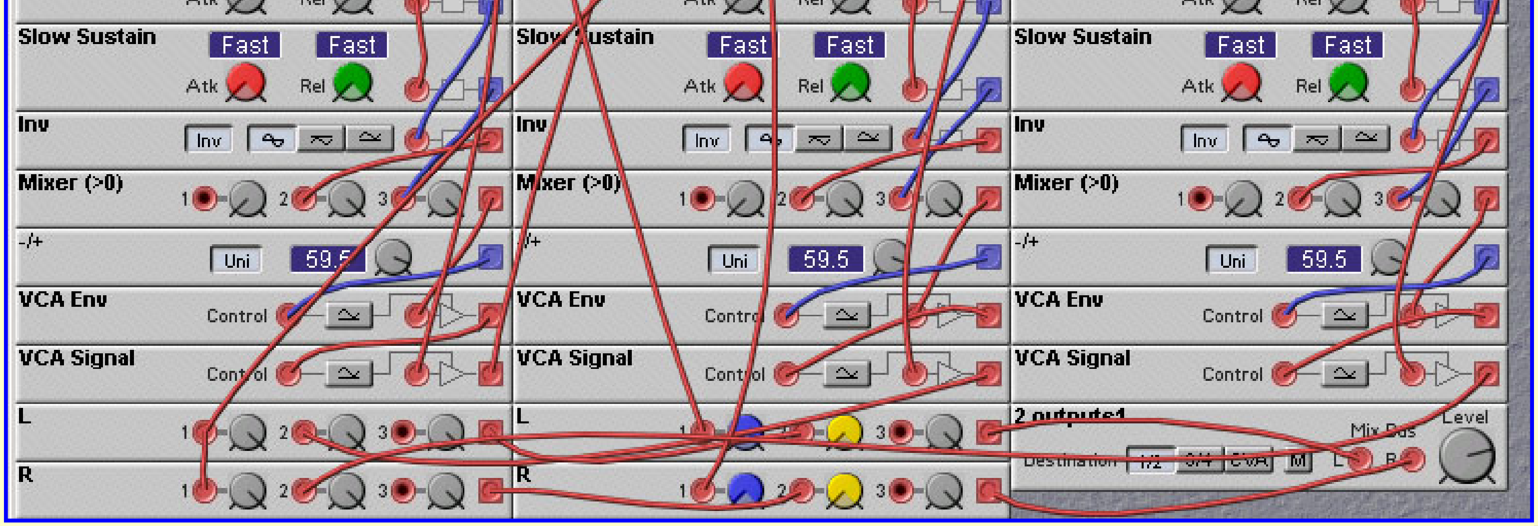
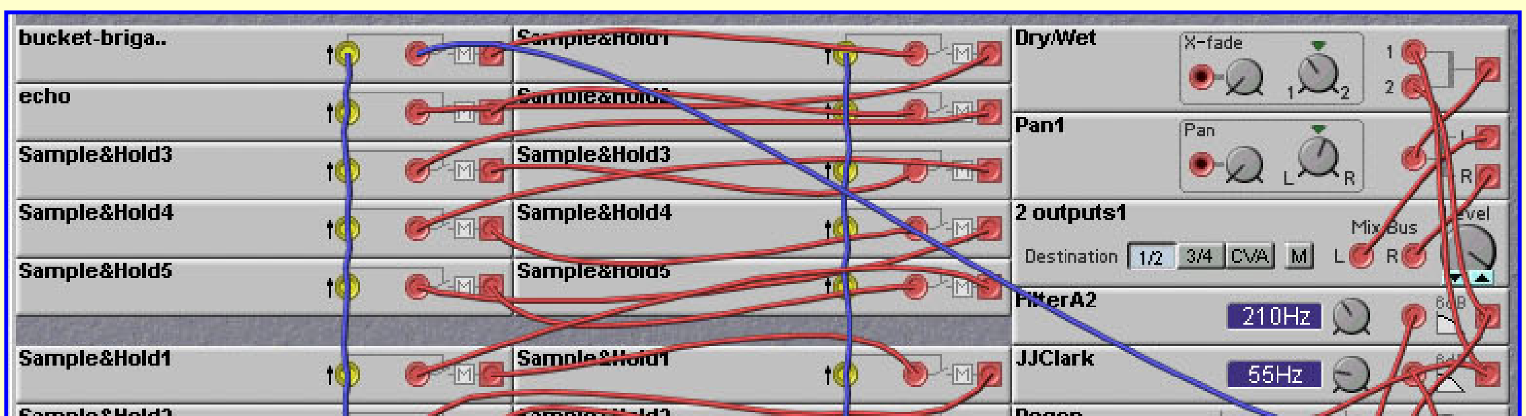


Figure 11.6. A low-fi reverb patch using analysis/resynthesis of the input (K. Thommen).

It is possible to make longer delays by trading off sampling rate for delay time. While the Nord Modular delay line modules do not have adjustable sampling rates, we can create our own delay lines using strings of Sample/Hold modules. A string of two S/H modules acts as a delay. Suppose we sample the input to the first S/H and hold it. Then, say T milliseconds later, we sample the output of this S/H with the second S/H. Only then will the input value find its way to the output of the second S/H. Thus there will be a delay of T milliseconds. Of course, the price we pay is that the sampling period of the output is also reduced to T milliseconds. But S/H modules are less resource hungry than the audio delay lines, so we can use more of them. In fact, you can get 98 of them before exceeding the Nord Modular resource limitations. Thus we can string together a whole bunch of these two-S/H delaylines to get one long delay line. This type of delay line structure is known as a *Bucket-Brigade* delay line (BBD). The name comes from the observation that the action of the sample/hold modules is much like that of the buckets of water used in old fire-fighting brigades, where water was carried in buckets from person to person in a line from the water source to the fire. BBDs are used in many analog chorus and delay effects found on synthesizers and guitar effects boxes. They are usually implemented using Switched-Capacitor or Charge-Coupled-Device (CCD) technology.

The following patch shows a bucket-brigade delay line being used in an echo circuit. It uses 90 S/H modules, giving a delay time of 89 times the sampling period (note: only part of the S/H delay line is shown in the figure below, to save space - download the patch file to see the entire circuit). So, if the sampling period is 5 msec (sample rate of 200 Hz) then the time between echoes is 445 msec. More than enough for a useful echo effect. The main drawback of this patch is that the input signal must be limited in high frequency content to under 1/2 the sampling rate (e.g. 100 Hz in the above example). So the patch is best used for low frequency sounds such as bass drums. To make sure that the input signal's high frequency content is limited, we filter it with a lowpass filter. This is known as anti-aliasing in signal processing terminology, as it prevents the "aliasing" of high frequency components back into lower frequencies when a signal is sampled. Note also the use of control mixers instead of audio mixers for combining audio signals. This reduces the DSP usage and causes very little signal degradation as the signal frequency is low enough to be handled by the lower control sampling rate.



Sample&Hold1
Sample&Hold2
Sample&Hold3
Sample&Hold4
Sample&Hold5
Regen
Lin
Inv
Inv
Input Sel
Input
1
2
3
4
M
AudioIn1
Left
Right
30 20 10 7 5 3 2 1 0
L
R

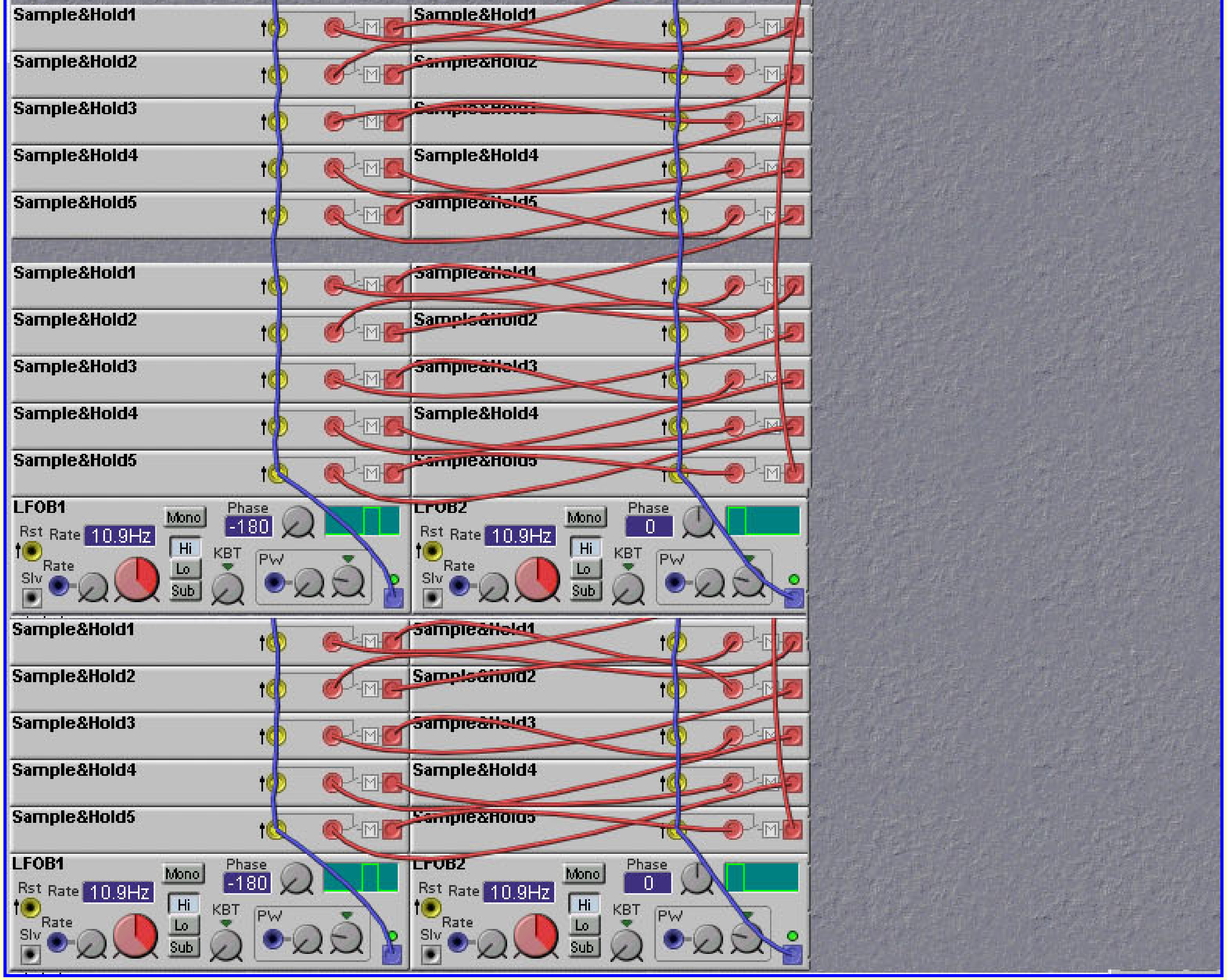


Figure 11.7. A low-frequency echo patch using a bucket-brigade delay line made with Sample/Hold modules (J. Clark).

Finally, the audio delay line modules *can* be coaxed into giving more delay. The trick is to *recirculate* the signal through the delay line many times. But wouldn't this just overwrite whatever is coming into the delay line? Not if we subsample the input, and *interleave* it with the delayed signal(s). Interleaving is a technique used in many communication systems (where it is called Time-Division-Multiple-Access, or TDMA) to send many relatively low frequency speech signals over a single high bandwidth communication channel. In our application, we make use of the fact that the audio delay line actually operates at quite a high sampling rate (96KHz). Suppose that we only needed a sampling rate of 9.6KHz. Then, theoretically, we could cram ten of these lower frequency signals into the delay line without them getting in the way of each other. So the idea behind the recirculating delay line is to use one slot for the input signal and the other nine (or however many slots you divide the line into) to hold the recirculating signals. Since, in this example, we have room for nine recirculating signals, the total delay time will be 9 times the nominal delay time of the audio delay line. The following patch does just this - it creates a recirculating delay line, where a crossfader is used to switch the input to the audio delay line module between either the input or the output at appropriate times to do the recirculation. The sample/hold module is used to sample the output at the right time - we only want to sample the output when the data has finished recirculating the desired number of times. The way to ensure this is to sample the output just before reading in the next input sample. The patch is just for demonstration purposes. To alter it for practical use you should remove the test input signal generation modules (the VCO, envelope etc) and replace it with an audio input module. Also add in a crossfader between the input signal and the output of the echo unit to provide a wet/dry mix control.

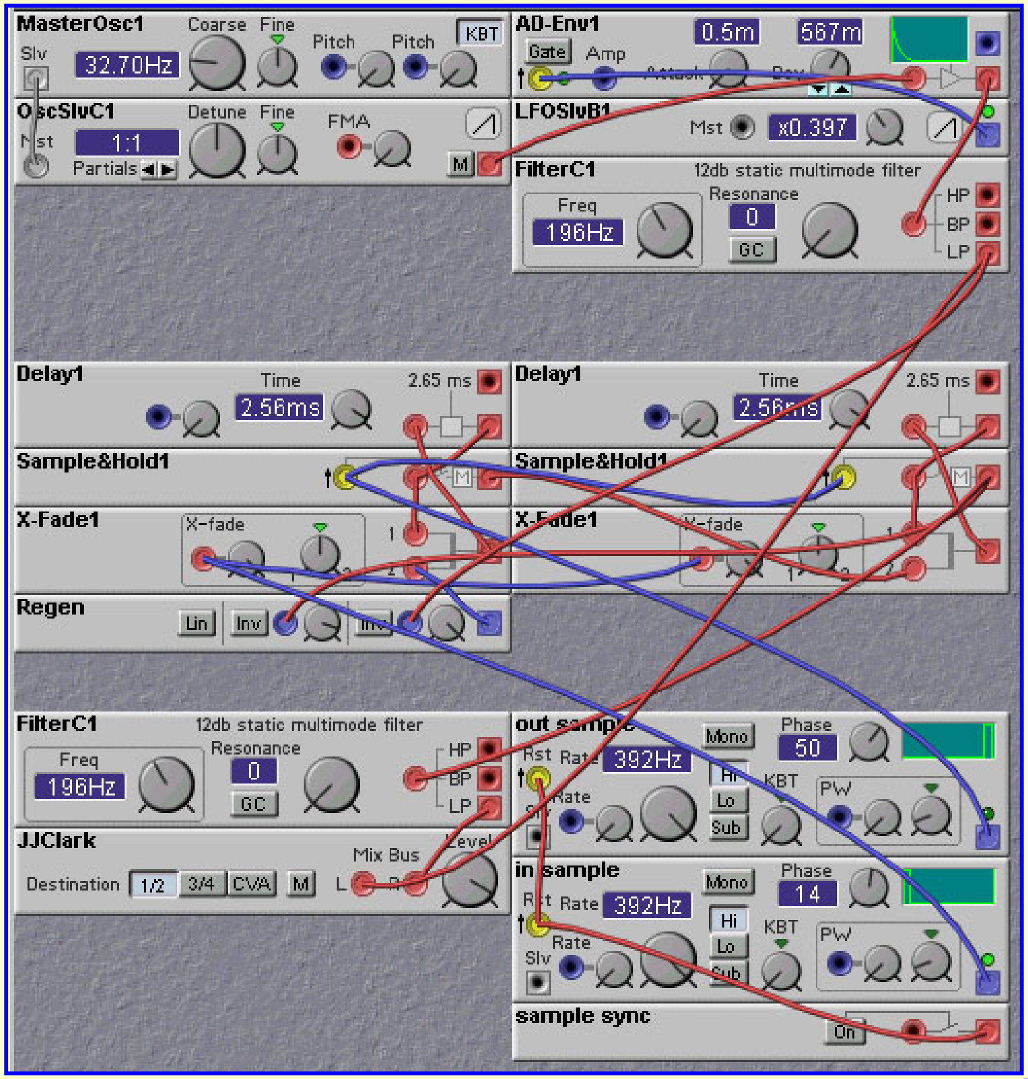


Figure 11.8. A low-fi echo patch using a recirculating delay line (J. Clark).

This patch uses fewer resources than the bucket-brigade echo patch, but it suffers from the same drawbacks, namely the limited frequency response. It is also very difficult to adjust the echo time. This is because the delay time of the recirculating delay line depends critically on the sampling rate relative to the delay time of the individual audio delay line modules. For most ratios proper interleaving is not possible, and the recirculating signals interfere and overwrite each other. So careful adjustment of the sampling rate and of the delay line module delay time is needed for proper operation. The following table lists the total delay values of a single recirculating delay stage (there are two in the patch shown above) attained for various delayline delay times, with a sample rate of 392Hz:

Delayline Delay Times (msec)	Total Delay (msec)

1.31	23
1.29	39
2.65	54
2.60	88
2.58	129
2.56	245

Table 11.1. Total delay of the recirculating delay stage as a function of delayline module delay times.

You can see that the total delay time does not vary linearly with the delay module time, as you might expect. This is because the number of recirculations is equal to:

M = (T_R*T_D/GCD(T_R,T_D)) - 1

where **T_R** is the period of the recirculation clock (in units of the audio signal sample period), and **GCD(T_R,T_D)** is the greatest common divisor of **T_R** and **T_D** (the delay time of the audio delay line module). Here is a simple example. Suppose that T_D = 4, and T_R = 6 (where the time units are the audio rate sampling period, which is 1/96,000Hz = 10.417 microseconds). Since the GCD of 4 and 6 is 2, the total delay time is M = 4*6/2-1 = 11. We show the contents of the delay line after each time unit in the following table:

0000 (initially delay line is all zero)
1000 <- 1 (first input)
0100 (recirculate)
0010 (recirculate)
0001 (recirculate)
1000 (recirculate)
0100 -> 0 (sample output and recirculate)
2010 <- 2 (second input)
0201 (recirculate)
1020 (recirculate)
0102 (recirculate)
2010 (recirculate)
0201 -> 1 (first input makes it to the output after 11 time steps)
3020 <- 3 (third input)
0302 (recirculate)
etc...

Table 11.2. Simulation of a simple recirculating delayline. External inputs are entered into the delayline every 6 time steps, and at all other time steps the output is recirculated back to the input. The delayline output is sampled on the time step before the input time. Inputs are seen to arrive at the output 11 timesteps after being fed in.

The output is sampled one time unit before the input is sampled (and fed into the delay line). The total delay for this example is seen to be 11 time units. This is equal to (T_D*T_R/GCD(T_D,T_R))-1 = (4*6/GCD(4,6))-1 = (4*6/2)-1 = 11.

For a more detailed example, suppose the recirculation clock period T_R is 1/392Hz = 2.55 msec, and the audio delay line delay time is 1.31 msec. In order to compute the GCD we need to scale these times by the audio signal sampling period. This gives (rounding to the nearest integer) T_D=125 and T_R=245). Then, GCD is the greatest common divisor of 245 and 125. This is GCD=5. Thus the total delay time will be T = (245*125/5-1) = 6124 time units, or 63.79 msec. Note that this value is much higher than the experimentally obtained value of 23 msec listed in the table above. Why the difference? Well, part of the problem is that the delay line delay time listed in the table is taken right off of the control knob on the delay line module. This value is not the exact value. The formula for the total delay time is very sensitive, so a little inaccuracy can cause a big change. Also, the formula given above assumes that the input and recirculation sampling pulse width is equal to one audio signal sampling period. If the sampling pulse is fairly wide, the number of recirculation slots will be lower than the maximum value (which is 256). In the above patch we use a fairly wide pulse (equal to about 25 audio sample periods), since it is very difficult to get a pulse with a width equal to one audio signal sample period, due to antialiasing of the oscillators. So the lesson to be learned is to throw away the math and experiment! (or find a more accurate mathematical model!) The maximum theoretical delay time that can be attained with a recirculation clock frequency of 392 Hz is 648 milliseconds. The best we can get with our patch is a bit less than half that. Using a slower recirculation clock will give longer delays but will result in lower fidelity. One should also note that the fidelity of this patch is compromised even further by the interpolation being done by the audio delay line modules, which tends to blur together neighboring recirculation slots. Since these slots are not neccessarily filled by segments of sound adjacent in time, all sorts of nasty filtering effects can be caused. Presumably, if there is no modulation of the module's delay time (through the delay time control input) then there will be little or no interpolation done (since the delay time will be an integer multiple of the audio signal sampling rate). These filtering effects are also exacerbated by the fact that the input and recirculation sampling pulses have a non-zero width. For large numbers of recirculation slots this means that neighboring slots may overlap. You can try to reduce the width of the sampling pulses, but I found that doing so causes other problems.

A plot of the output of the recirculating delay line echo patch is shown in the figure below. You can see that the echo time is about 0.5 seconds, and that the signal is degraded somewhat.

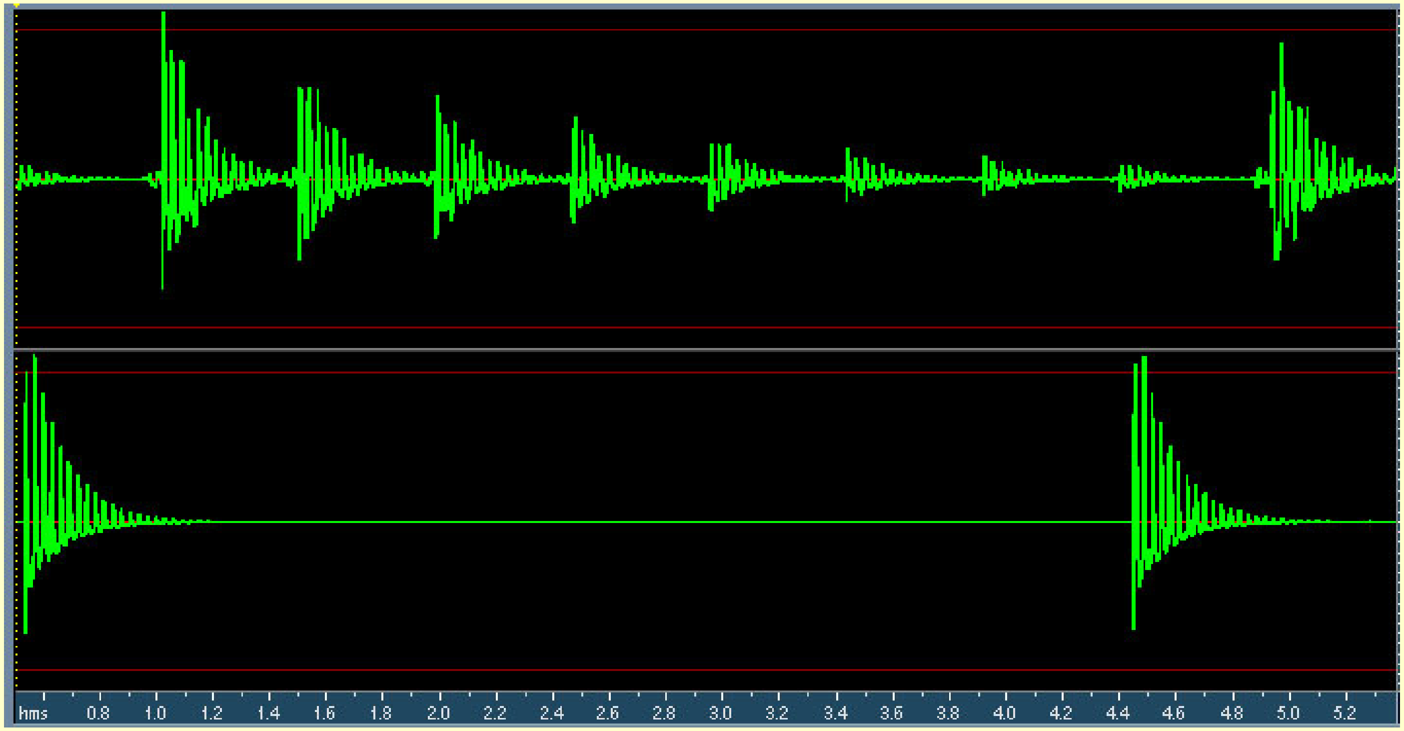


Figure 11.9. The response of the recirculating delay line patch shown above to a transient input signal. The period of the echoes is about 0.5 seconds.

The following figure shows a reverb patch based on the recirculating delay line technique. It uses the well-known *Gardner* reverb algorithm developed by William Gardner in his 1992 MIT Master's thesis. Details of the Gardner algorithm are available in many places on the web. The sound of this patch is quite bad, partly due to the low frequency response, partly due to the distorting effects of overlapping recirculation slots, and partly to the difficulty of finding suitable delay time settings to match those suggested by Gardner.

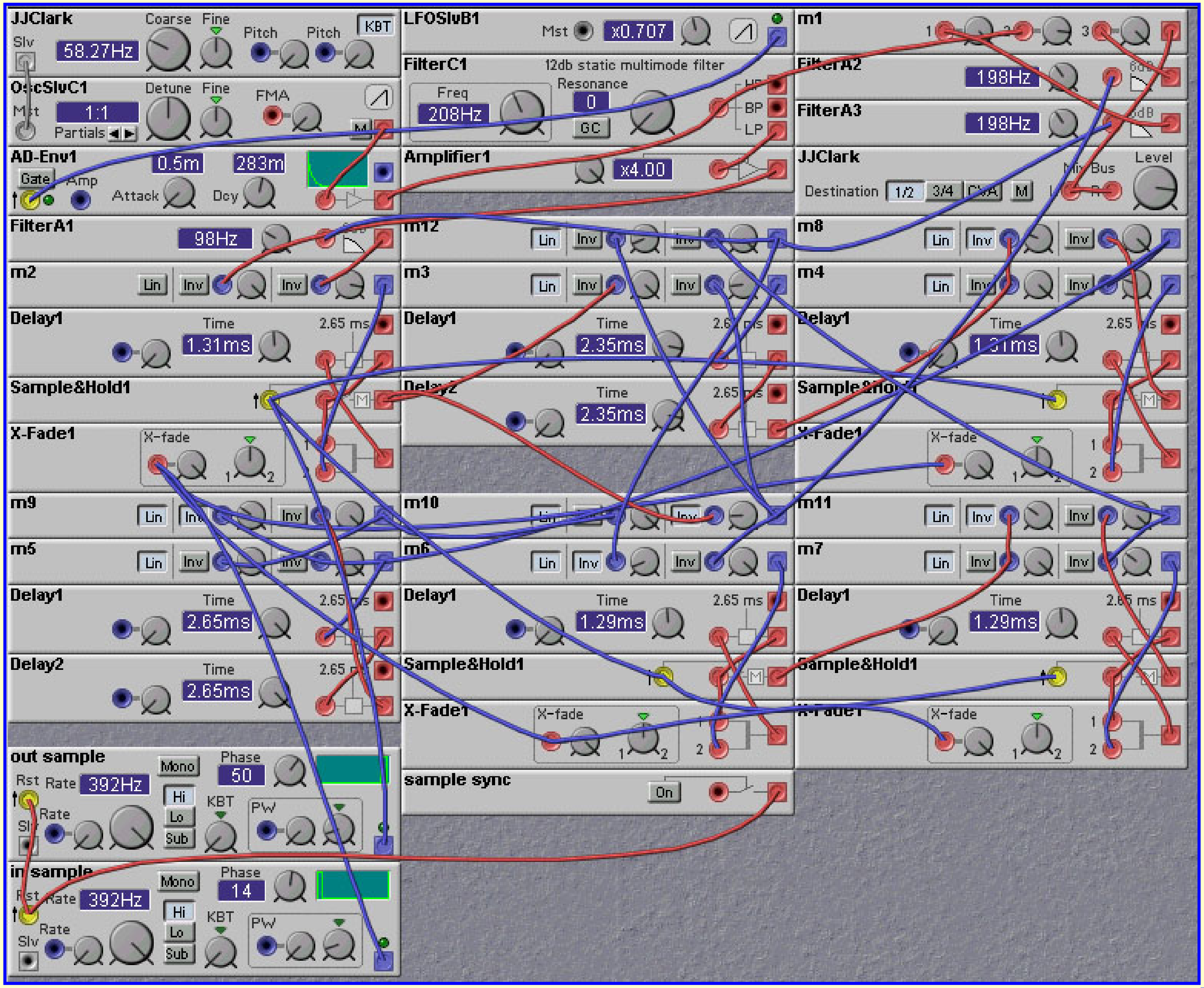


Figure 11.10. A low-fi reverb patch using a recirculating delay line (J. Clark).

[Prev (Algorithmic Composition)][Next (Distortion Effects)][Table of Contents]

Chapter 12. Distortion Effects

The general idea behind distortion effects is to add harmonics to a sound. Either nonlinear functions or time-variant functions are needed to do this. Purely linear time-invariant functions, such as those implemented by standard filters, can only attenuate or enhance existing harmonics and not produce new ones.

12.1 Distortion using Nonlinearities

The most common approach to implementing distortion is to use a monotonic nonlinearity (one where the output level is an increasing function of the input level). The Nord Modular has a number of modules that implement such monotonic nonlinearities - the Shaper, Clip, Overdrive, and Quantizer modules.

Less common is the use of non-monotonic nonlinearities to create distortion. In the Nord Modular the WaveWrapper module is characterized by a non-monotonic nonlinearity. The distortion provided by this module is more extreme than that produced by the monotonic nonlinearities.

The following patch, designed by *Wan Kemper*, includes almost all Nord Modular modules that can distort audio signals, including the Shaper, Clip, Overdrive, WaveWrapper and Quantizer modules. These are strung in a series beginning with the most gentle distortion (the Shaper module) leading to more and more severe distortion (WaveWrapper and Quantizer). This ordering allows a wide range of distortion from subtle to radical.

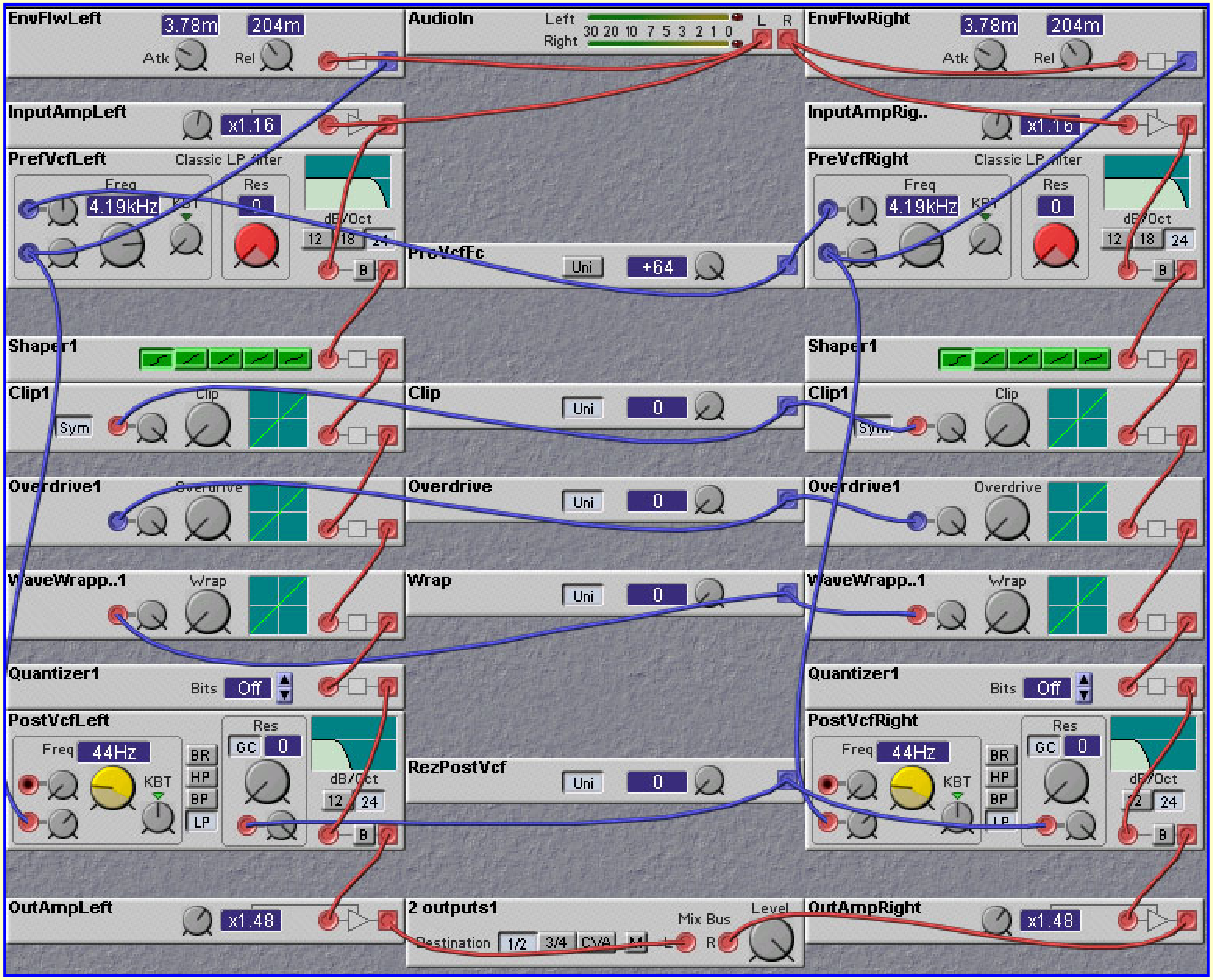


Figure 12.1. Serial distortion patch, including many different nonlinear modules to distort the input (W. Kemper).

The Nord Modular amplifier module can also produce a nice asymmetrical clipping if it is overdriven, as can many other modules, such as the mixer modules, and even the phaser module.

12.2 Multiband Distortion

Distortion always seems to give nice results when applied to harmonically simple sounds such as the decay of a feedback electric guitar, which is almost sinusoidal. Passing a spectrally complex sound through a distortion module usually results in obnoxious noise. The reason for this is a phenomenon called intermodulation. Intermodulation occurs when a signal consisting of multiple sinusoids with non-harmonically related frequencies are passed through a nonlinearity. In addition to the generation of the harmonics of each of these sinusoids, components are generated whose frequencies are harmonically related to the sum and differences of the original frequencies. In general, intermodulation results in lots of new spectral components whose frequencies are not harmonically related to any of the original input frequencies. So the sound that is created can be very harsh and objectionable. So, the trick is to try and reduce the amount of intermodulation while still retaining the creation of new harmonics. One way to do this is to split the signal into a number of separate frequency bands and apply different amounts and types of distortion to each band. This limits the intermodulation components to just those frequencies that are passed through a given band. You can tailor the distortion levels in each band to suit the particular types of musical input. For example, in dance club sounds the bass is often very hot, so the amount of distortion applied to the lower frequencies should be minimal, otherwise it can overwhelm the overall sound.

One of the most popular multi-band distortion units is the DIY (do-it-yourself) circuit known as the Quadrafuzz. This was designed by Craig Anderton and appears in his book "Projects for Guitarists" (GPI Books). A version of this circuit is manufactured (in kit form) by PAIA Electronics. The quadrafuzz, as its name would imply, splits the incoming signal into four bands, each approximately an octave in width. The outputs of each filter are passed through clipping style distortion circuits. The outputs of each distortion unit are then mixed together and passed through a final low pass filter. A Nord Modular implementation of the Quadrafuzz is shown in figure 12.2. It uses Overdrive modules rather than clipping modules and uses a lowpass filter for the lowest band and a highpass filter for the highest band, but otherwise the approach is very similar to the Anderton Quadrafuzz.

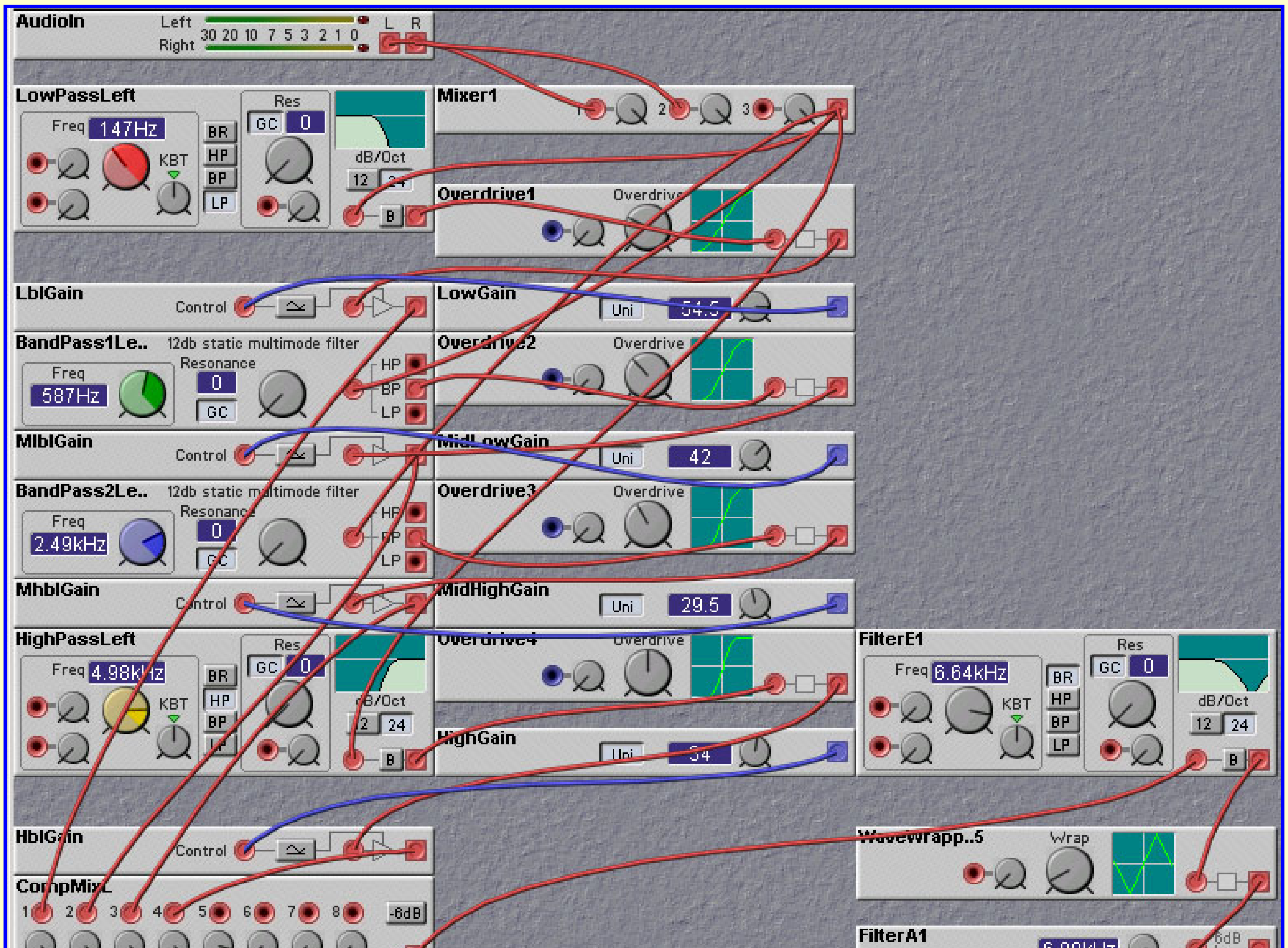


Figure 12.2. Four-band distortion (W. Kemper).

You don't want to take the multi-band distortion to extremes, however. It may seem like a good idea to use a large number of narrow bands and tailor the distortion for each band. The problem with this is that the harmonics that are generated are dominated by the harmonics of the filter band center frequencies, which are fixed, rather than the harmonics of the input frequencies. This leads to a rather static, hollow sound, akin to some vocoder sounds. It is an interesting effect in its own right, perhaps, but not what one would expect from a distortion effect. Thus, you should limit yourself to using a few bands, using rather broad filter banks. Although using a very large number of bands, say with an FFT-based harmonic re-synthesis approach might give a good result - a distortion with very little intermodulation components.

An effect related to distortion is the so-called "Exciter". This type of circuit aims to add harmonics to the sound in a way that does not sound like distortion, but rather as a *restoration* of high frequency detail that may have been filtered out somewhere along the line. Such circuits are often said to increase "presence" or "clarity" or add "sparkle" to sounds. The most famous of this type of effect is the Aphex Aural Exciter, which was first produced in the mid-70's. The Aphex effect highpass filters and phase shifts the input signal followed by a patented "Transient Discriminate Harmonics Generator" which creates varying amounts of harmonics depending on the dynamics of the input. Clearly, there are a lot of practical details left out of this description of the exciter's operation, and making a patch that duplicates the Aphex sound is challenging to say the least. But the basic idea, that of adding high frequency harmonics to the sound is not too hard to accomplish, as demonstrated with the following patches.

The first patch applies a highpass filter to the input signal and passes the filtered result to an overdrive module. The output of this distortion is then passed through another highpass filter, to allow selection of only the higher harmonics, if desired. The cutoff frequency of the two highpass filters are adjustable, thereby giving some control over the generation of the higher harmonics.

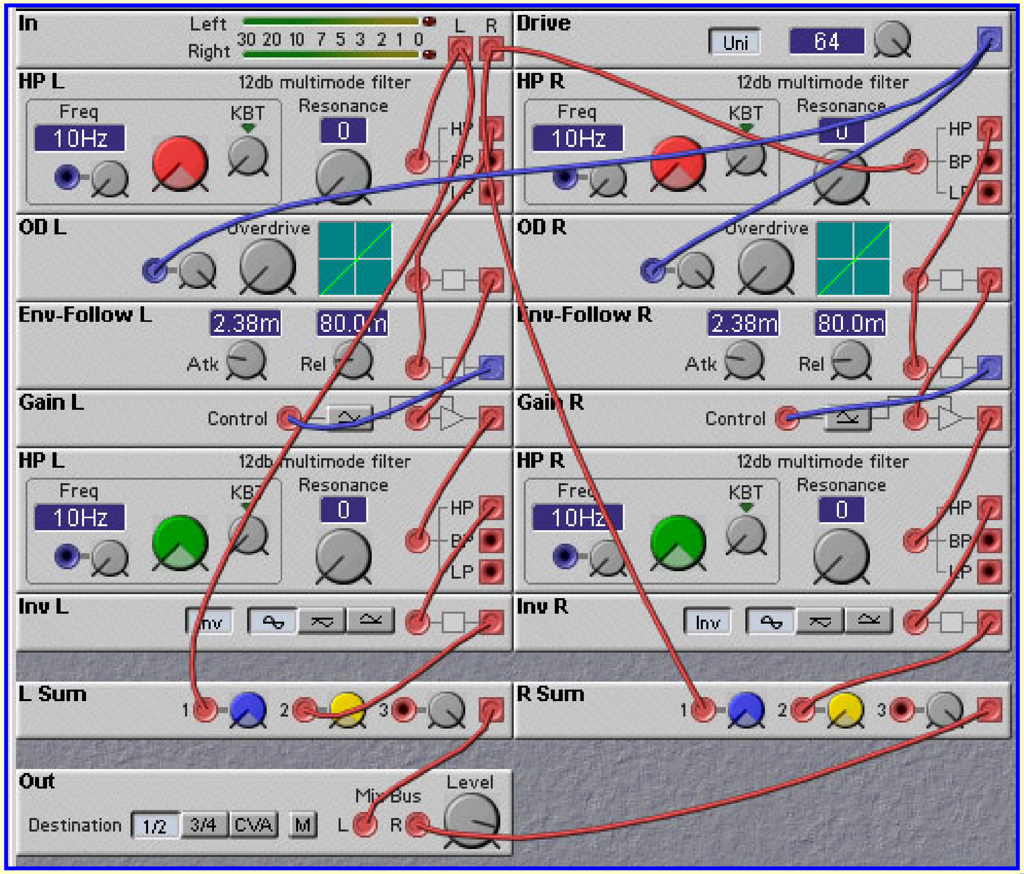


Figure 12.3. An "exciter" patch, which adds high frequency harmonics to the input (K. Thommen).

The second exciter patch is much like the first, but also contains a "big bottom" circuit, which aims to emphasize low frequencies sounds. The patch was designed by Rob Hordijk, who modeled it after the Behringer Enhancer. Aphex introduced the idea of the "big bottom" circuit as an addition to their exciter module. In the following patch the big bottom effect is obtained by adding a delayed version of a highpass filtered input (with a relatively low cutoff frequency) to the output. The delay does not affect the low frequencies very much (as they are not passed through by the highpass filter) but cause cancellations of some of the higher frequency components. This causes a perceived boost in the low frequencies relative to the higher, without affecting the overall amplitude significantly. This is important when driving amps or speakers that might not be able to handle extreme bass boosting or properly render sub-harmonics. Both of these exciter patches lack the (proprietary and secret) dynamics processing of the Aphex products.

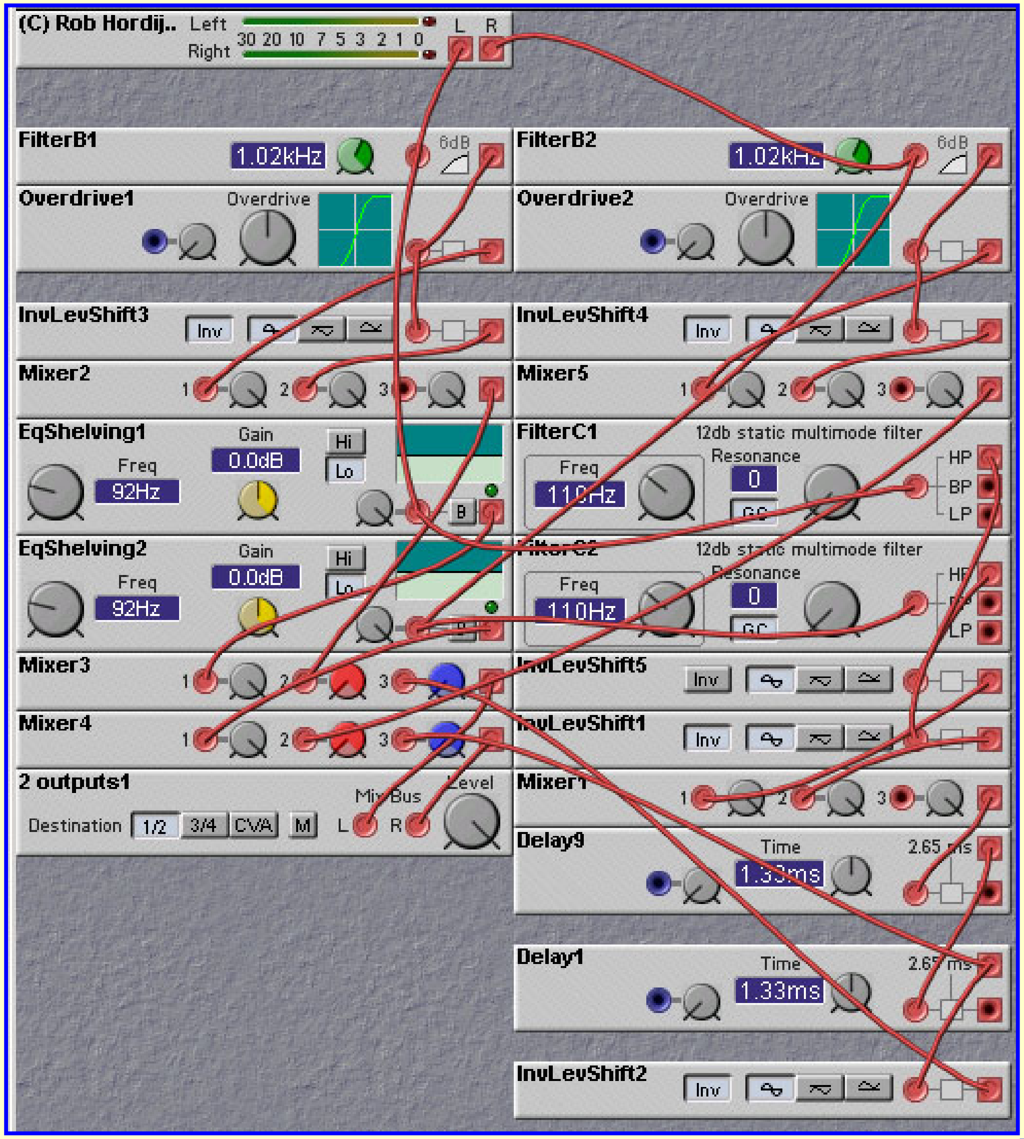


Figure 12.4. Another "exciter" patch (R. Hordijk).

A common criticism of "stomp-box" distortion units is that they don't sound very good when applied to synthesizer sounds. Some would take this to mean that the designers of distortion boxes just concentrate on getting them to sound good for guitars. Part of the problem is that guitar waveforms have some nice properties which permit good results to be obtained from applying distortion, properties which most synthesizer sounds don't have. These properties are: simple harmonic structure and smooth, controllable, decay. As we have noted above, the primary effect of distortion is to add harmonics. If a sound already has a lot of harmonics the distortion will just alter the level of the harmonics. If there are inharmonic frequency components, the harmonics added by distortion will tend to emphasize the inharmonic nature of the sound. Both of these effects will tend to result in a "harsh" sound.

We can do some things to improve the sound provided by passing a synthesizer sound through a distortion effect. The first of these is to restrict the harmonic structure by applying distortion separately to each voice in a polyphonic patch, rather than to the sum of all the voices. That is, you should apply distortion in the mono area rather than in the poly area. By summing the individual voices *after* distortion one avoids the intermodulation that you get if you summed the voices before distortion.

12.3 Polynomial Distortion

Any nonlinearity has the potential to create new harmonics in a signal. For most nonlinearities, there are actually an infinite number of harmonics generated. Of course, most of these will be filtered out by the high frequency cutoff of the analog signal chain in your system, and aliased to lower frequencies by the digital signal chain. In some situations, the user may wish to have greater control over the production of harmonics and, in particular, perhaps restrict the creation of new harmonics to a small number.

The creation of a fixed, finite, number of harmonics can be achieved with distortion functions whose input/output mapping can be expressed by a finite order polynomial. Chebyshev polynomials have the property that Tn(cos(theta)) = cos(n*theta). So if you make a linear combination of these polynomials, you keep a harmonic sound and you add frequencies in the spectrum.

You can find different Chebyshev's polynomials with the following recurrence relations:
T0(t)= 1
T1(t)= t
Tn+1(t)-2*t*Tn(t)+Tn-1(t)=0
The kind of distortion created by passing an audio signal through such a polynomial can be quite musical because it is not too harsh. The limited nature of the nonlinearity means that the number of intermodulation products is likewise minimized.

By adding together outputs of a number of different Chebyshev distortion units having different orders (eg. T0, T1, T2, etc), we can tailor how much of the various harmonic orders we obtain. This is useful in emulating various analog distortion modules. For example, in many tube amps, such as Marshall amps, distortion mainly produces even order harmonics, with odd orders coming in only at extremely high input levels. Some lower quality tube amps may have significant 3rd order harmonics. Push-pull tube amps can cancel out even order distortions, leaving mainly 3rd order harmonics. The solid-state "Fuzz-Face" distortion units distort asymmetrically resulting in significant 2nd order harmonics along with significant 3rd order harmonics, and some 4th and 5th order harmonics. The solid-state Electro-Harmonix Big-Muff PI circuit is symmetric and produces mainly 5th and 7th order harmonics.

The following patch illustrates how Chebyshev distortion can be implemented on the Nord Modular. It implements a sum of the first four (counting from 0) Chebyshev polynomials:
C(t) = T3(t)+T2(t)+T1(t)+T0(t) = (4t^3-3t)+(2t^2-1)+t+1.
It should therefore produce the first, second, and third harmonics of the input signal, as well as passing the input signal itself.

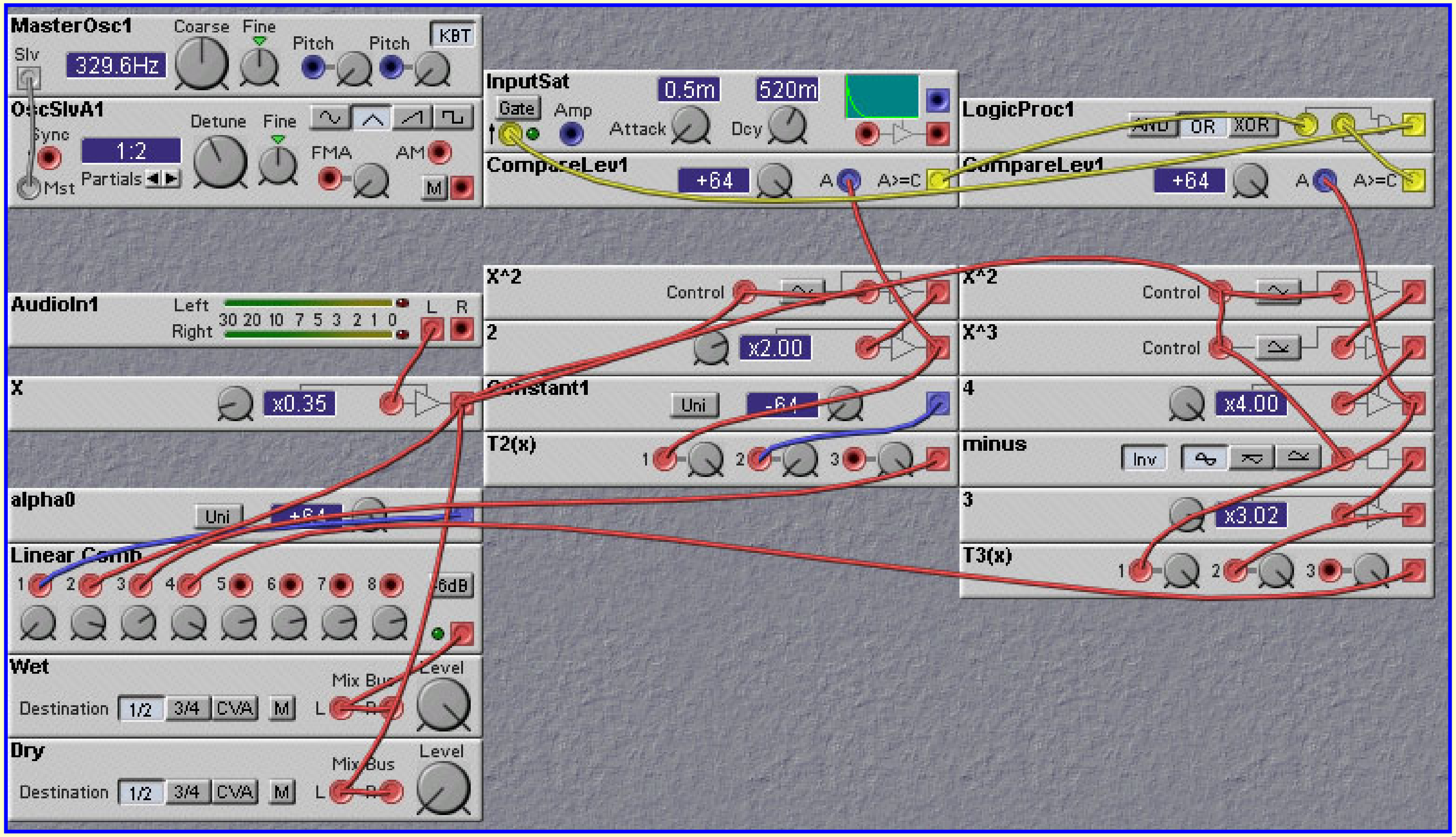


Figure 12.5. Chebyshev polynomial distortion technique (Thomas).

A problem with Chebychev distortion with the higher order polynomials is that, if you are not careful with signal levels in the patch, the multiplication operations can cause the some internal signals to become too large and be clipped. If this happens, more harmonics are generated than you want, and the sound can become muddy. In the patch shown above, an amplifier is used on the input to try to keep the internal levels low enough to prevent internal clipping. This amplifier (or attenuator) has the effect of reducing the dynamic range of the input signal, causing increased quantization noise, especially at low input levels.

Another problem is that the Chebyshev polynomials as defined by the recurrence relation given above only produce single, clean n-th harmonics if the amplitude of the input sinewave is exactly 1. A non-unit amplitude will result in more than one harmonic being generated. It is a good idea to try and limit the dynamic range of the input signal, for example using compression. In addition to creating more harmonics than the single one, a non-unit amplitude sinewave passed through an even-order Chebyshev distortion will also result in a DC component being added to the waveform. This DC offset may cause saturation or clipping, especially if the signal is fed back. Thus it may be a good idea to follow this type of distortion with a DC blocker, such as a high pass filter (with a very low cutoff frequency).

Another polynomial distortion approach is to use X*ABS(X) instead of X^2, e.g. X-a(X*ABS(X)). It is not a proper Chebyshev function but yields a similar sonic result. The ABS (Absolute Value) function can be conveniently made with a Diode module set to Full.

12.4 Distortion using Time-variant Systems

It is a little-known fact that distortion can be generated using linear systems. These systems, however, have to be time-varying. That is, their input/output mapping function, while linear, has to change with time.

If the time-variation of the system parameters is periodic, and this period is a harmonic of the input signal fundamental frequency, then the output of the time-varying system will contain harmonic components. If the period of the time-varying system is not harmonically related to the input signal, or if the time-variation is not periodic, then the output signal will have non-harmonic components. While the creation of non-harmonic components may produce interesting results, it is not what is usually referred to as distortion, so we will not consider it further here. We will restrict our notion of distortion to be those systems which create new harmonics. Thus we will only look at time-variant systems where the time variation is periodic and related to the fundamental of the input signal.

You might argue that these systems are really nonlinear rather than linear, because the transfer function between the input and output can be written as a nonlinear function. And you would be right. But, nevertheless, it is often worthwhile thinking of systems as time-variant rather than nonlinear. Consider the following distortion patch in which the main signal path is directly through a linear element, in this case a filter. The filter cutoff frequency is varied by a control signal derived from the input signal. Since the time-variation is periodic (for a periodic input), the output of the system is harmonic, and by our definition, is a distorted version of the input signal. Try it on an external input signal, such as voice, and compare the results to other distortion techniques!

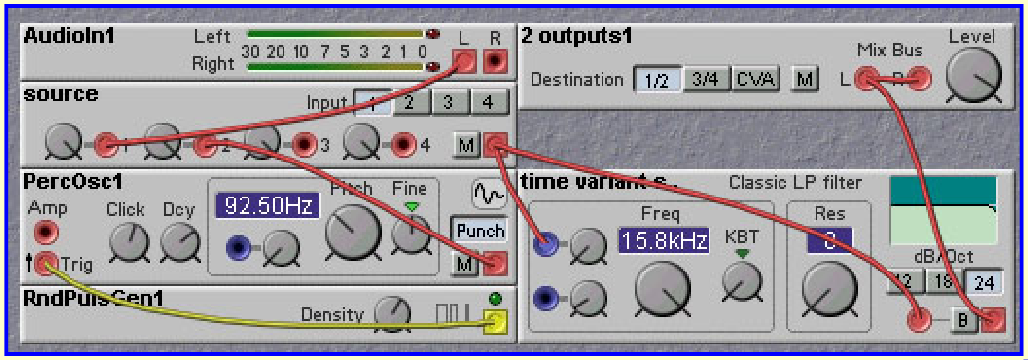


Figure 12.6. Linear time variant systems (a time varying filter as distortion producers (J. Clark).

[Prev (Reverb and Echo Effects)][Next (Frequency and Pitch Shifting)][Table of Contents]

Chapter 13. Frequency and Pitch Shifting

An effect that is lots of fun at parties is pitch shifting. Kids like making their voices deep like adults, and adults like making their voices high-pitched like children!

There are actually two forms of spectral shifting that are commonly encountered - *pitchshifting* and *frequency shifting*. These are related, but different, effects. With pitchshifting the frequencies of signal components retain their harmonic relationships. For example, a signal with a 1KHz fundamental and two harmonics at 2KHz and 5KHz, could be pitchshifted upwards by a factor of 2.5, and the signal components would now have frequencies of 2.5KHz, 5.0KHz, and 12.5KHz. The ratios of the frequencies of the signal components are seen to be the same as they were before the pitch shifting.

In frequency shifting, on the other hand, harmonic relationships between signal components are not preserved. To use the same example as before, if we shift our signal up in frequency (not pitch) by 1.5KHz, the resulting signal will have components whose frequencies are 2.5KHz, 3.5KHz, and 6.5KHz. The ratios of these frequencies are not the same as they were before the frequency shifting. The sound of a frequency shifted signal can be much different than that of a pitchshifted signal. But both techniques can give pleasing and musically useful results!

13.1 Frequency or Spectrum Shifting

with Rob Hordijk

Spectrum shifting (technically called "single sideband modulation") can be done with two sinewave oscillators that are in quadrature (90 degrees out of phase), a phase-shifting allpass filter that shifts all frequencies 90 degrees with respect to the input, and a couple of ringmodulators. The ringmodulators multiply the two phaseshifted signals with the two sinewaves. The outputs of the ringmodulators are then summed. The effect is that all frequencies (=harmonics) are shifted up or down by a fixed number of Hz, this number being the frequency of the two sinewaves. This means that the stronger the effect the more the harmonics lose their harmonic relation. (This in contrast to pitch shifting!) So the effect is that you "morph" a sound between the dry sound and a "ringmodulation" effect sound at extreme shifts. A shift of a few Hz can e.g. make a static synthetic sound more lively. The difficulty in constructing such a frequency shifter is to make the phase-shifting filter (usually called a "Hilbert Transformer"). Such a filter usually only gives the desired phase-shifting properties over a limited frequency range. So, one must use a bank of bandpass filters, each followed by a phase-shifting filter giving approximately 90 degrees phaseshift for that band. If you use this multiple band approach you can make things really interesting if you start to randomly modulate the spectrum shifts in each band. This can give very beautiful "very big choir" chorus effects. If you use two spectrumshifters in each band, one shifting in the reversed direction of the other and mix those to left and right channels, the effect becomes very spatial as well. Professional effects units would employ thirty-three 1/3 octave bandpass filters, sixty-six spectrum shifters and thirty-three random modulators. (Actually one can modify a spectrum shifter to give two outputs, one shifted up and the other shifted down, so only 33 are needed). In addition you can give every band a randomly modulated delay of 10 msec with random predelays between 0 and 300msec for each band. This shifts the formants in time in relation to each other and thickens the effect even more. Its clear that you need a lot of DSP power and memory for the 33 delaylines to do this!

Some attempts by some people have been made to make a frequency shifter patch on the Nord Modular, but due to lack of a good allpass phase-shifting filter the results were not of very high "quality". Still interesting sounding, though. One such patch is shown below, designed by *Rob Hordijk*. This patch just shifts frequencies downwards. In order to shift frequencies upwards, you need to swap the outputs of the two quadrature sine wave oscillators (OscSlvFM1 and OscSlvFM2). With many more and much sharper filters the effect could be much more accurate, but the patch is especially nice with strings, female vocals (fun!) and brushed cymbals. Only the five frequency bands where the center frequencies are exactly 90 degrees out of phase get shifted by the required amounts, and the places in the spectrum where the phaseshifts do not match seem to come through slightly ringmodulated, which produces a tremolo with only minor shifts, giving some strange eerie phasing to sounds.

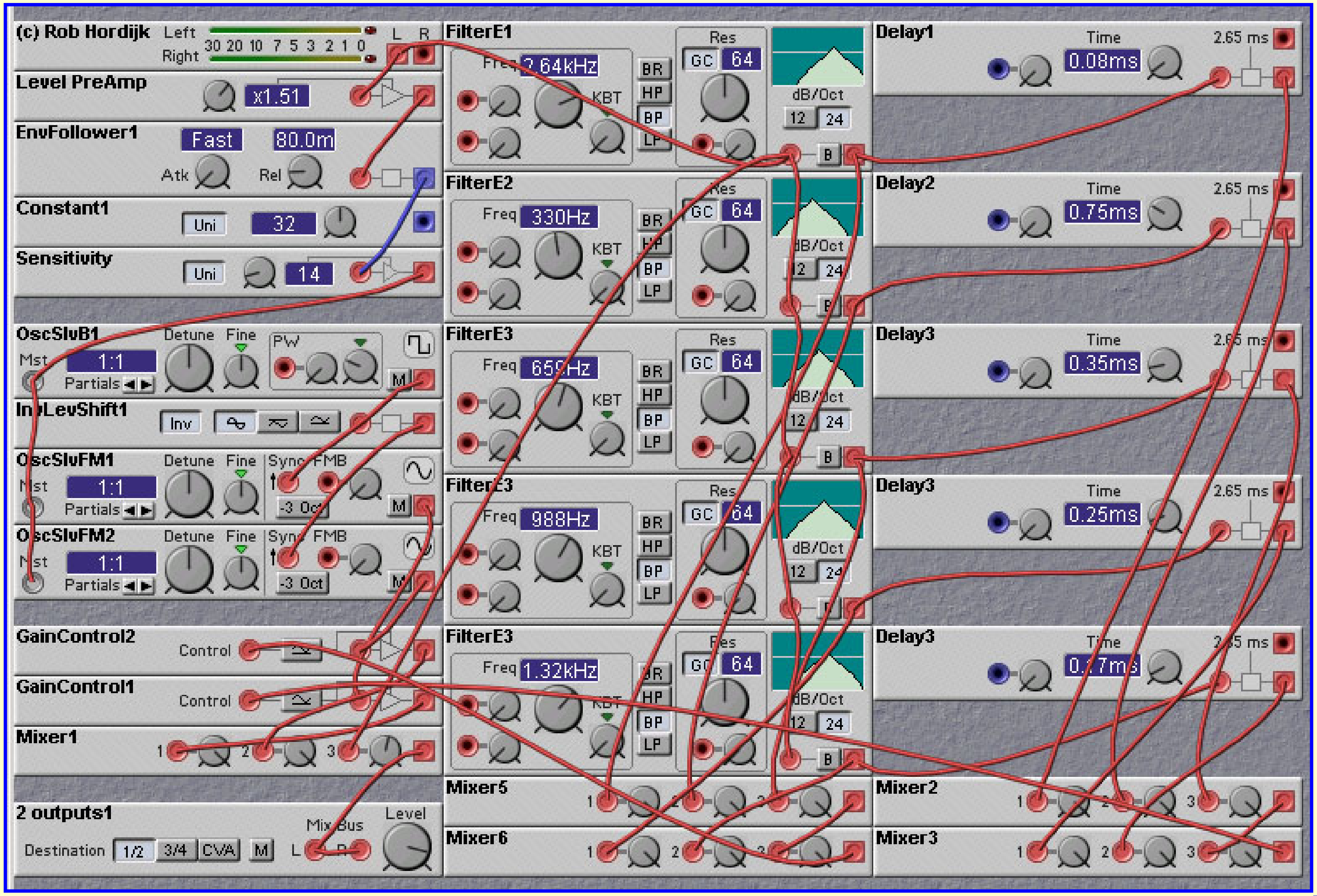


Figure 13.1. A 5-band spectrum shifter (R. Hordijk).

The following patch constructed by *Tim Walters*, shows a frequency shifter based on a Single-Sideband-Modulation (SSB) technique developed by Donald K. Weaver in the paper "A Third Method of Generation and Detection of Single-Sideband Signals", *Proceedings of the IRE* December, 1956, pages 1703-1705. A block diagram of the Weaver technique is shown below.

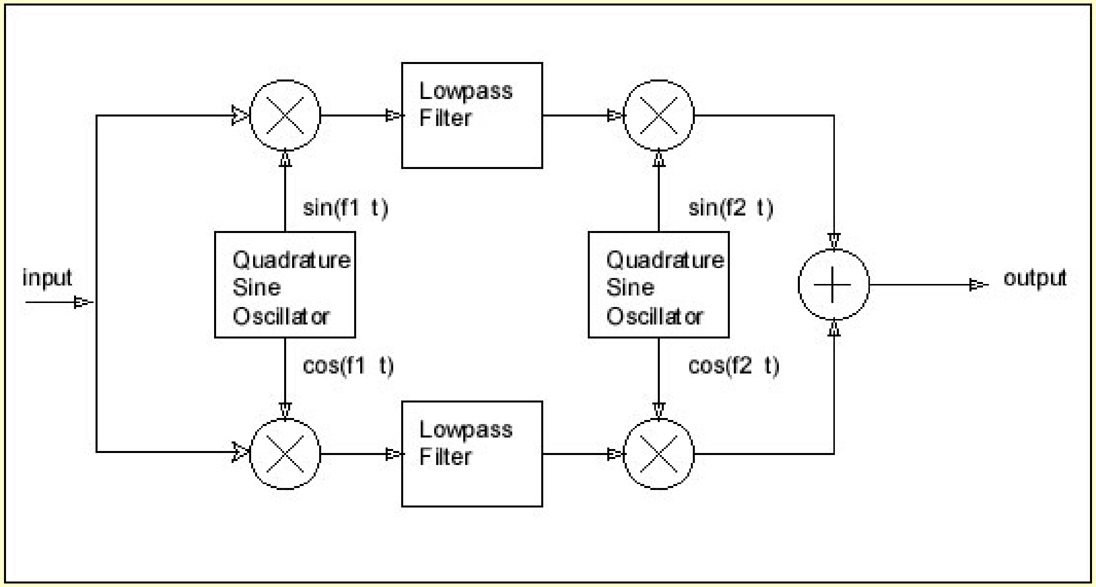


Figure 13.2. A block diagram illustrating the Weaver SSB frequency shifting technique.

A nice description of how this circuit works is given in the article "An Efficient, Precise Frequency Shifter", by Jens Groh, in the CSound Magazine, summer 2000 issue. Unlike the spectrum shifting method described earlier, this approach does not need any phase shifting filters, but it needs four ringmodulators instead of two, and two quadrature oscillators instead of one. In Tim Walters' patch the quadrature oscillators were implemented with two LFOs, where the phases of the LFOs are adjusted to be 90 degrees apart.

The amount of frequency shift is equal to the absolute value of the difference between the frequencies of the quadrature oscillators. Does it make a difference which oscillator has the higher frequency? It makes no difference from the point of view of the amount of frequency shift, but it does make a difference to the type of aliasing error that is added by the shifting process. Usually this aliasing error will be minimal when the amount of frequency shift is small, but can be significant if the shifting is large. The lowpass filters should have a cutoff frequency equal to (or less than) to the frequency of the first quadrature oscillator. Both filters must have the exact same characteristics. The filters will limit the allowable frequency range of the input. Thus it is better to use as high a modulation frequency as we can. The maximum allowable modulation frequency is one quarter of the sampling rate (or 24KHz). In Tim Walters' original patch we can only get up to 392 Hz. This greatly limits the range of frequency shifts that we can obtain, as well as limiting the frequency range of the inputs.

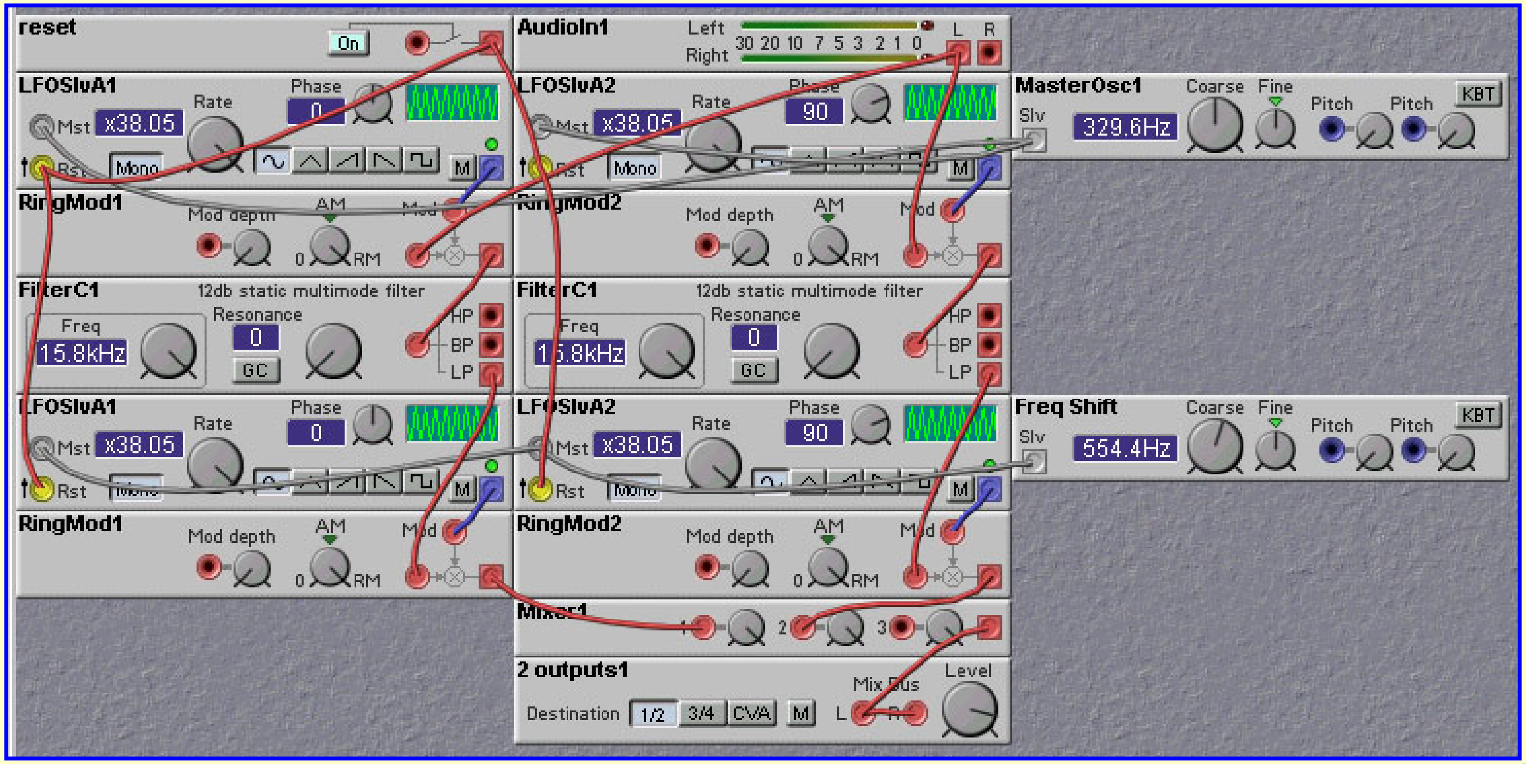


Figure 13.3. A Nord Modular frequency shifter patch based on the Weaver SSB technique (T. Walters).

A phase-lock-loop (PLL) circuit can be implemented that will give a higher frequency quadrature oscillator with the Nord Modular modules. This approach is used in the patch shown below.

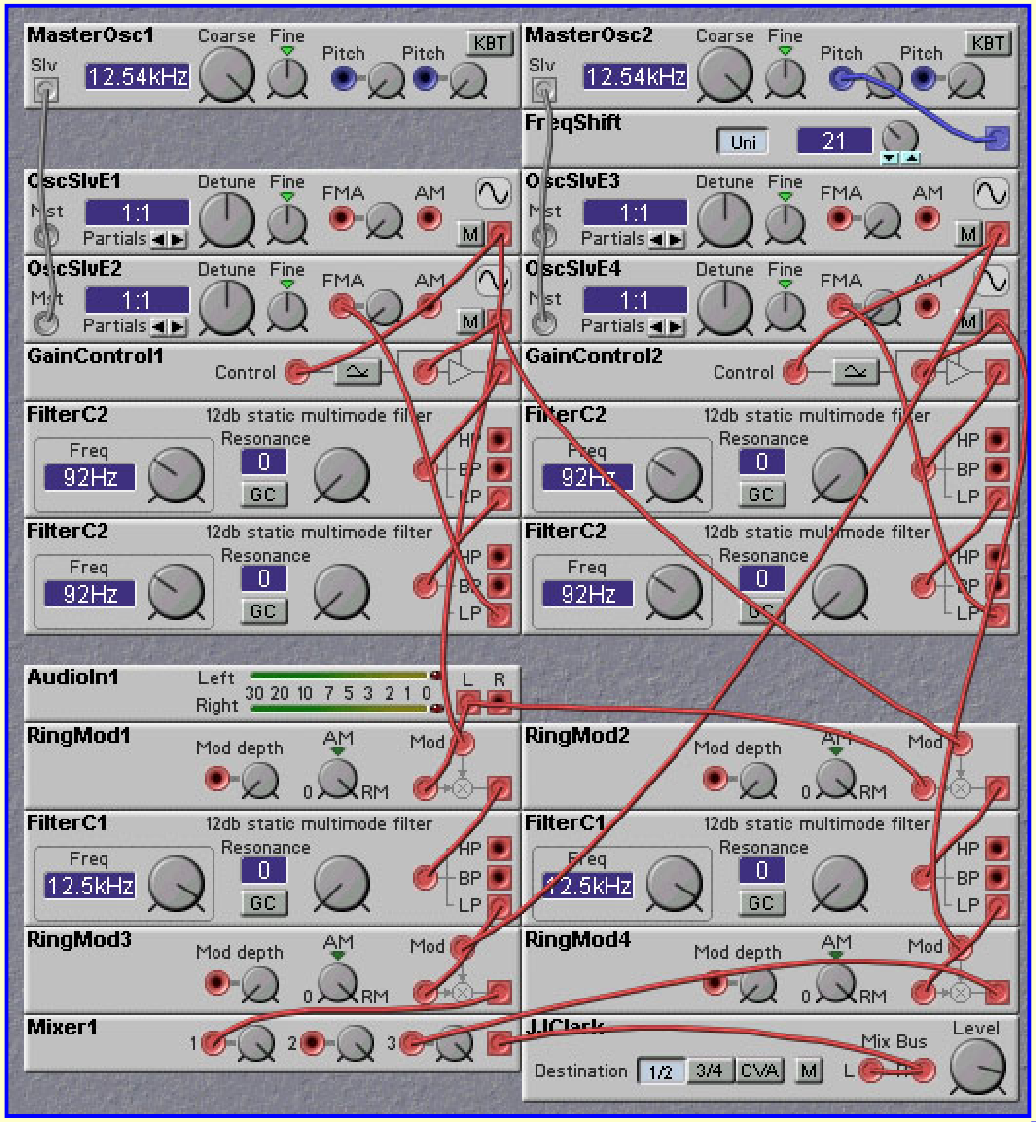


Figure 13.4. A Nord Modular frequency shifter patch based on the Weaver SSB technique using Phase-Locked-Loops to implement quadrature oscillators (J. Clark).

The phase shift of 90 degrees needed for the quadrature oscillator is obtained by taking two sin oscillators running at the same nominal frequency and modulating the frequency (and hence the phase) of one of them. The modulation signal is derived by multiplying the outputs of the two sinusoidal oscillators and lowpass filtering the product. If the two oscillator waveforms are exactly 90 degrees apart, with the same frequency, then the time average of their product will be zero. If there was a phase difference of less than 90 degrees the product would be greater than zero, and the frequency of the modulated oscillator would increase slightly, causing the phase difference to increase. Similarly, if the phase difference was greater than 90 degrees, the product would be less than zero and the frequency of the modulated oscillator would temporarily decrease, causing the phase difference to decrease. In both situations the phase difference will be forced towards 90 degrees, where it will remain, barring any noise or other disturbances.

13.2 Pitch Shifting

A traditional pitchshifter works by filling a fixed length delayline with a fixed rate and reading it with a variable rate. The rate difference causes the pitchshift. The delayline in the Nord Modular has a fixed rate, but its length can be varied. This can also be made to give a pitchshift effect. To understand how this works, consider a long tunnel in which cars are driving. Suppose you are working for the highways ministry and have the job to count the number of trucks passing through the tunnel. So you sit at the far end of the tunnel and start counting. Each time you see a truck, you put a little blip on your chart at the time you saw it (figuring that you will add up all the blips later). Say you see one truck every 10 seconds. This makes you rather bored, wishing you could be off playing Wakeman-esque noodles on your Nord Modular. Suddenly, you are struck with a brilliant idea! You can speed things up by running through the tunnel against the traffic (being careful to dodge the trucks). This way you can count all the trucks without waiting for them to reach the end of the tunnel. You know there is a strict speed limit in the tunnel, so every truck is moving at 50 kilometres per hour. You decide to put on your sprinter's shoes and run down the tunnel at 25 kilometres per hour. Thus, it seems that the trucks are now moving past you at 75 kilometres per hour. So now you see one truck every 6.67 seconds. In other words, the apparent frequency of the trucks has increased (was 0.1 Hz, and is now 0.15 Hz). If you could run faster, the apparent frequency of the trucks would be even faster! You are sure to be finished before lunch at this rate! But as you near the entrance to the tunnel you realize that there is a catch. Once you reach the beginning, you will have to go back to the other end! So you head back down the tunnel at 25 kilometres per hour. You decide to keep counting the trucks, but you find that, since you are moving with the trucks, they have slowed down relative to you, and are only moving past you at 25 kilometres per hour now. So the apparent frequency of the trucks is now half of what it was originally, or 0.05 Hz, thereby undoing all the gains that you achieved in the first place. But then you remember that there is a teleportation unit at the entrance used by the maintenance people, that can whisk you instantaneously from the entrance to the exit. This will get you back to the exit without wasting anytime. So you reach the end of the tunnel and start running back and counting as before. But you notice that a lot of the trucks look familiar. In fact you have counted some of them before, since they haven't had time to exit the tunnel since you counted them. So you are still seeing trucks go by at 0.15 Hz, but you are double-counting half of them.

Now, as a technique for counting trucks, this probably won't win you any praise for your boss, but it may help your chances in the Olympics. However, for purposes of music generation and processing, this trick comes in handy. In place of the tunnel, take the Nord Modular delayline module. It is like a tunnel where audio samples enter and exit instead of trucks. The output of the delayline is taken at a variable location along the delayline. So the output takes the place of the runner in the above tunnel analogy. Continually reducing the delay time (the location of the delayline readout) corresponds to the runner moving back down the tunnel. Thus we expect that reducing the delay time in this manner will increase the frequency of the signal being readout of the delay line. Similarly, if we steadily increase the delay time, the frequency of the signal being readout will decrease, just as in the case of the runner running towards the exit of the tunnel.

In order to get a constant increase (or decrease) in frequency, we will have to set the delay time back to its maximum once reaching zero delay (just as the runner did in using the teleportation device). In this case the increase in frequency effectively comes from a double "counting" (or double reading-out) of the audio samples in the delay line. This sudden shift back to the end of the delay line will usually result in a big discontinuity in the signal being read out of the delay line (as the value of the signal at the end of the delay line will usually not be the same as that at the beginning). One way to get around this problem is to multiply the output of the delayline by a triangle waveform whose value goes to zero at the beginning and end of the delayline. In this way the output goes to zero around the discontinuity point. Doing this, however, results in an "amplitude modulation" of the output, which will certainly not be desired. So, what to do? Well, you could use two outputs, at different locations along the delay line. This would be analogous to having two runners in the tunnel. If you spaced the output locations half the total delayline length apart, then one will have a maximum modulation when the other is at zero. So, if you added the two output values together, the modulations would cancel and disappear.

A Nord Modular patch, constructed by *Kees van de Maarel*, implementing this idea is shown below. This patch uses a rounded square-wave rather than a triangle wave to do the amplitude modulation.

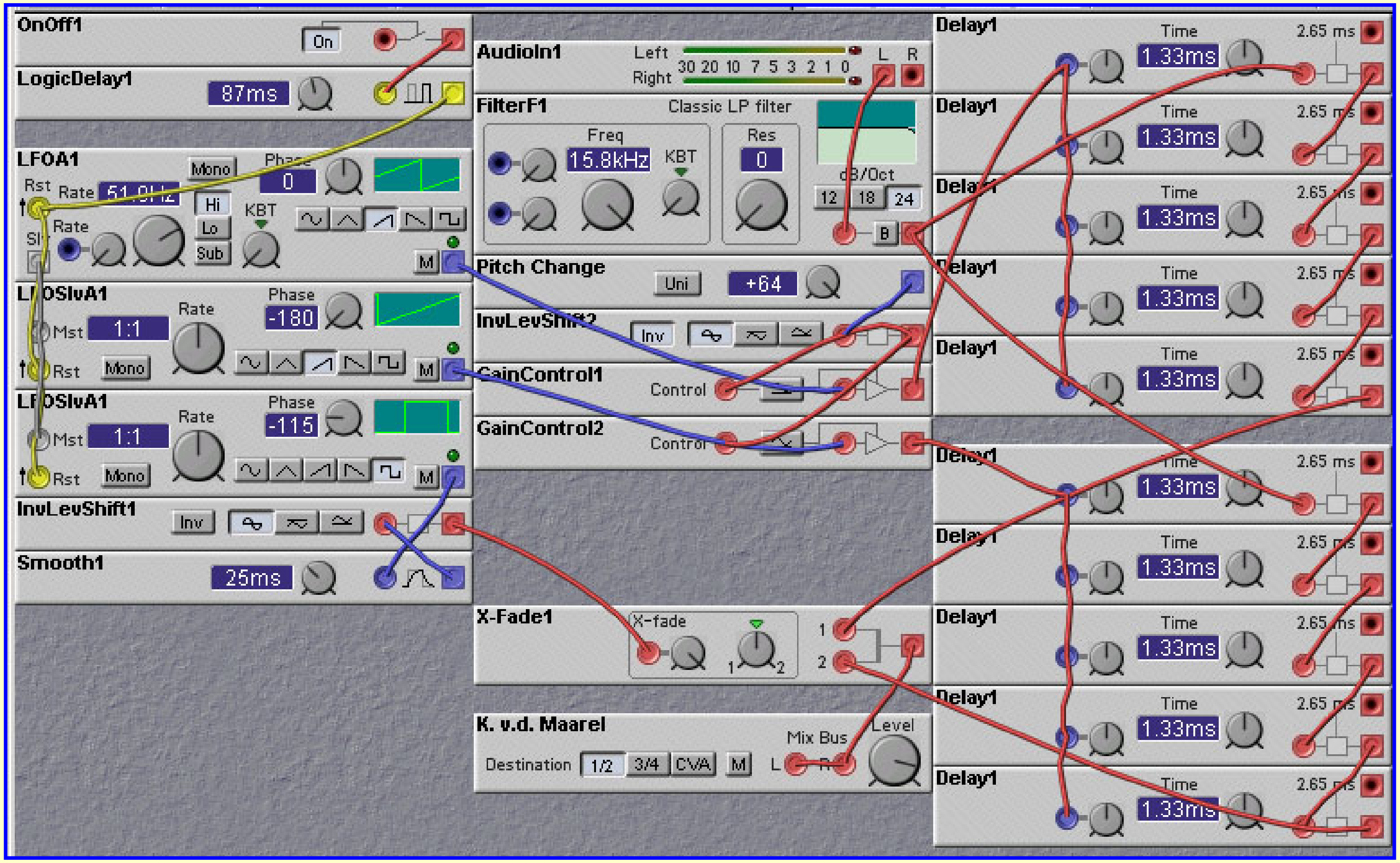


Figure 13.5. A delayline based pitch shifter patch (K. vd Maarel).

The length of the delayline influences the quality a bit and that's why you can find optimization parameters for different intervals on the pitchshifter effects in the better effects machines. These parameters in effect set the length of the delayline (think of the phase differences between the waveforms from the two reading points) and the number of reading points (you need at least two, but you can imagine that using more reading points and different windows can increase the quality).

A vocoder can also be used to perform pitch shifting. In this approach, a signals spectral shape is analyzed and use to reconstruct a new signal with a different pitch. Details on this approach can be found in the Speech Synthesis and Processing chapter.

[Prev (Distortion Effects)][Next (Spatialization Techniques)] [Table of Contents]

Chapter 14. Spatialization Effects

The human brain is capable of processing the auditory signals picked up by the ears and estimating the spatial location of sound sources. To do this, the brain makes use of many cues present in the auditory signals.

The two most important localization cues are the *interaural time difference*, or ITD, and the *interaural intensity difference* or IID. The IID is what most synthesizer users are familiar with, and arises from the fact that, due to the shadowing of the sound wave by the head, a sound coming from a source located to one side of the head will have a higher intensity, or be louder, at the ear nearest the sound source. One can therefore create the illusion of a sound source emanating from one side of the head merely by adjusting the relative level of the sound that are fed to two separated speakers or headphones. This is the basis of the commonly used *pan* control, such as the Nord Modular pan module.

The interaural time difference is just as important, if not more so, in permitting the brain to perceive spatial location. This time difference arises from the difference in distance between a sound source and the two ears. Since the sound travels at a constant velocity (under usual listening conditions) this distance difference translates into a time difference. A sound wave travelling from a sound source located on your left side will reach your left ear before it reaches your right ear. As the sound source moves towards the front of your head, the interaural time difference will drop and become zero when the sound source is centered between your ears. The maximum interaural time difference depends on the width of your head and on the speed of sound in your listening environment. If we take the the speed of sound to be 330 meters/second, and the distance between your ears to be 15 cm, then the maximum interaural time difference will be about 0.45 msec. Fortunately, the Nord Modular Delay module can easily provide this range of time delays, so that it is straightforward to implement the ITD spatialization cues with the Nord Modular.

The following patch, courtesy of *Anig Browl*, illustrates basic sound source localization using both IID and ITD cues. The important elements of this patch are the Pan module (which provides the IID cue) and the Delay modules (which provides the ITD cue). The patch also includes a filter which models the reduction in high frequency content of a signal when it is located behind the head.

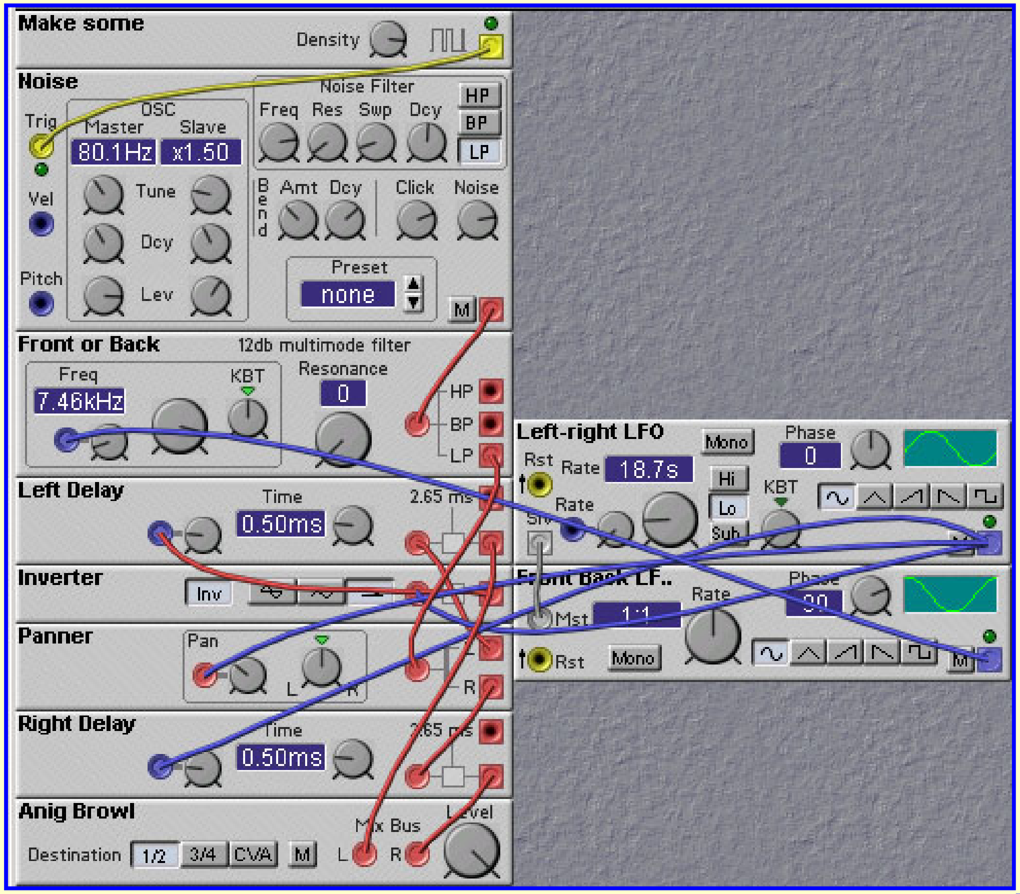


Figure 14.1. A patch illustrating the use of IIT and ITD cues in spatial localization of a sound (A. Browl).

The spatialization model used in the above patch is very simple. In actual practice the IID and ITD values depend on the frequency of the sound source. For example, the ITD is largest for low frequencies and rolls off at around 1KHz to a value about 1/2 its low frequency level at around 5KHz. IID is more important at high frequencies. This is because the low frequencies are not attenuated very much by the head, while the high frequencies begin to get attenuated by the head shadow around 2KHz. This head attenuation actually begins to be reduced around 5KHz, due to conduction of sound along the skin surface. Thus, there is a peak in the response in the far ear centered around 5KHz, after which the response falls away once again. Based on these details, we can make a more realistic IID and ITD based spatialization patch, as shown in the following patch.

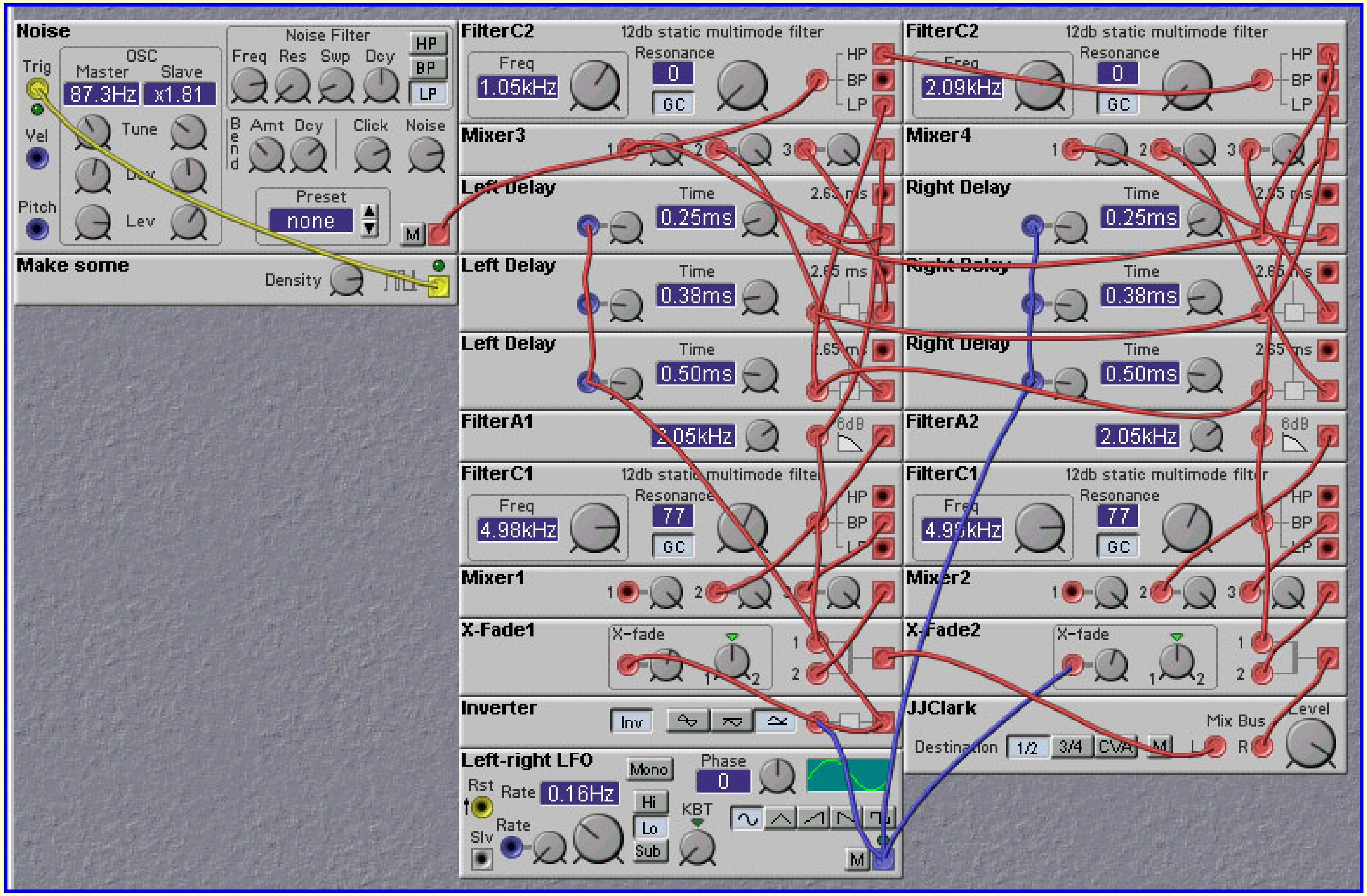


Figure 14.2. A patch implementing frequency dependent IIT and ITD cues in spatial localization of a sound (J. Clark).

The structure of the ear itself is constructed so as to aid in the localization of sound sources. The outer ear structure, which is refered to as the *pinna*, filters the sound in ways which depend on the direction that the sound impinges on the ear. All of the folds, which give each ear its own distinctive shape, reflect and absorb sound waves. The reflected waves interfere or support each other in a wavelength dependant manner, creating the filtering effect. The brain can use the pinna-induced spectral changes to extract additional localization informations. Note that, unlike the IID and ITD cues, which the pinna related cues can be used to provide information about the vertical localization of sound sources. That is, one can get information about whether sounds are coming from sources above or below the head. When people move their heads they get slight changes in the various cues that permit localization. Many animals (and some people!) can even move their ears, which can produce changes in the way in which the pinnae interact with the incoming sound. These slight changes themselves provide extra information that the brain can use to improve localization. These subtle, but important, motion effects are usually not reproduced in most synthesizer listening situations, due to the lack of sensing of head motion. This does not mean that such a thing is not possible. There are many devices on the market that can be used to sense head position. You can even try building your own! These devices could be modified or enhanced so as to render their outputs in the form of MIDI controller messages, which can then be used to adjust the spatialization parameters (IID, ITD) in a Nord Modular patch.

In addition to the filtering action of the pinna structures, the ear canal itself acts as a resonant filter, with a peak at around 5KHz. The frequency and position dependent characteristics of the pinnae and ear canal are usually summarized in the form of what is called the *Head-Related Transfer Function* or HRTF for short. These transfer functions describe the frequency response of each ear for sound sources at various locations. The transfer functions are obtained by doing acoustical measurements with microphones embedded in the ear canal of individual listeners, or microphones implanted in models cast from the ears of listeners. Clearly, doing these measurements for every possible listener of your music is impractical, so designers of spatialization systems have relied on using generic models obtained from a small set of test listeners. The results obtained with these generic models are not as convincingly realistic as with individually tailored models, but are still an improvement over the more basic approaches that do not model the influence of the pinna at all. From the point of view of implementing HRTF models with the Nord Modular, the news is not good. The HRTFs, even the generic ones, are quite complicated, with many little peaks and valleys in the response curve. It is these little variations that give the accurate spatialization, so approximating the HRTF by simpler filters does not yield realistic results. Another difficulty is that HRTF data is hard to come by - it is difficult to make your own measurements (but feel free to try!) and companies that have their own generic models tend to guard them jealously and don't give out the details. We will discuss some alternatives, however, in the next section.

Virtual Speaker Simulation in Headphones

As most readers will know, listening to music through loudspeakers is a much different experience than listening to it through headphones. Part of the reason for this is that people usually don't play their loudspeakers loud enough to hear all the details which they pick up on headphones. More importantly, however, it is difficult to provide convincing spatialization with headphones using the simplistic IID spatialization techniques found on many recordings. Consider a sound panned hard left. In loudspeakers, the sound is perceived as coming from the left, and outside the body. In headphones, the same sound would be perceived as being located right at the left ear! Thus, a lot of research has been done in developing *virtual* speakers where headphones give the same aural impression as speakers located in some position (usually 30 degrees off center).

The first system that was developed to improve headphone spatialization was the *crossfeed network* developed by Benjamin Bauer in the 1960s. This system simulated the crosstalk that is present in normal hearing by mixing in a time-delayed (0.4 msec) and lowpass filtered (to 5KHz) version of one audio channel to the opposite channel. Since this system did not accurately model the frequency dependent pinna effects, it did not provide a completely natural sound localization, but did produce a sound that was spatialized to some extent and solved the problem of hard panned sounds being perceived as "inside" the headphone. A Nord Modular patch that implements this crossfeed system is shown below:

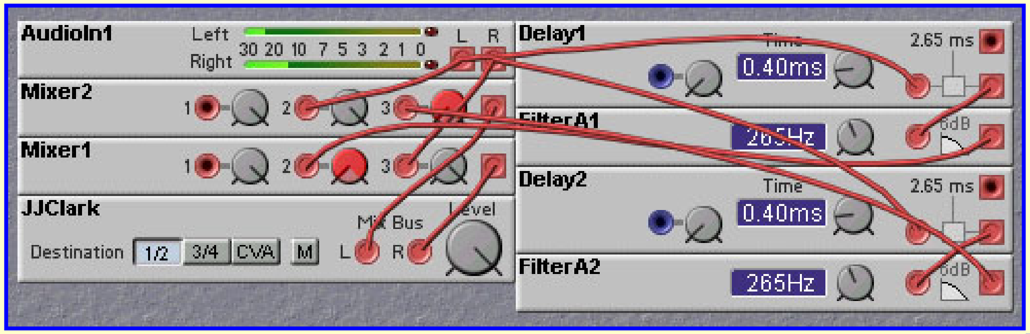


Figure 14.3. A headphone spatialization patch based on the Bauer crossfeed network (J. Clark).

To improve upon the Bauer crossfeed system, we must take account of the HRTF of the ear structures. One of the advantages of merely trying to recreate the sound of a single speaker in a given spatial location, rather than the more general problem of placing a sound in an arbitrary location in space, is that only one HRTF need be used. This can be extended to the problem of surround sound by recreating the sound of four or five speakers. This would require four or five fixed HRTFs, which is still easier than computing suitable HRTFs on the fly for localizing sound sources at arbitrary positions. A simple approximation to a generic HRTF was implemented, in 1996, by Douglas Brungart (US Patent 5,751,817). In his system the HRTF was modeled on the HRTF measured in a dummy head 7 feet from a sound source 30 degrees off center. Brungart approximated the rather complex HRTF that was measured with a simple filter that had a resonant peak (with Q of 5) at 5KHz. This filter reasonably models the ear canal resonance but does not do a good job with the pinna effects. A patch with implements this simple HRTF filter is shown below:

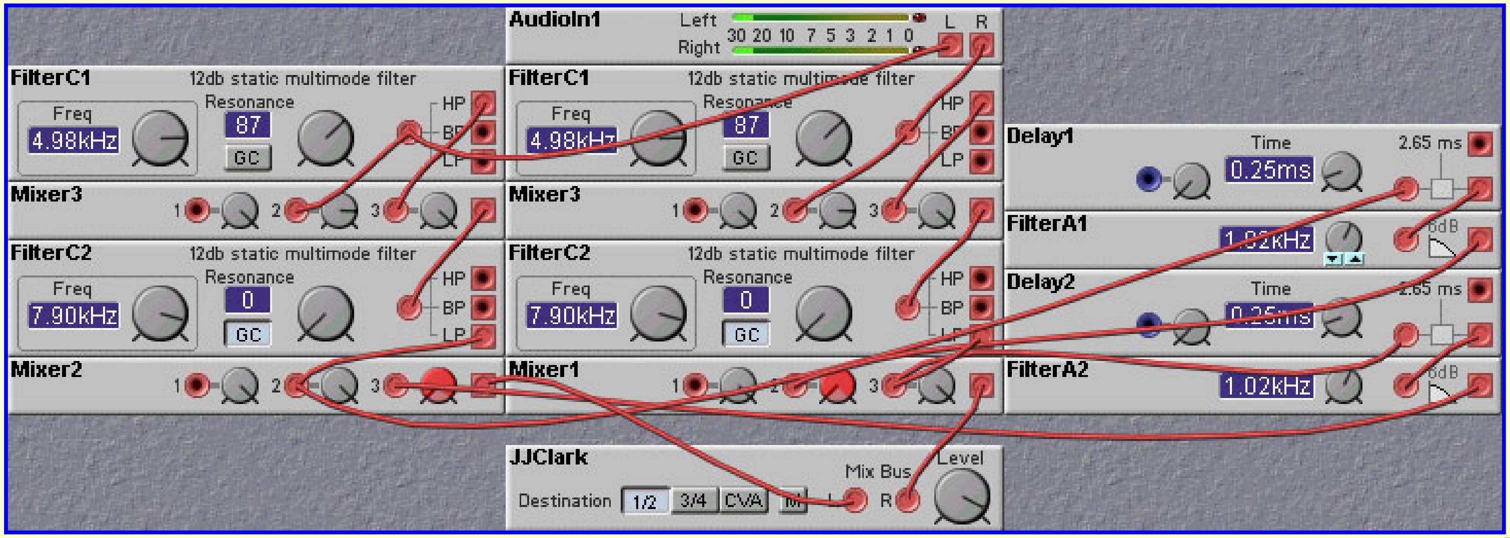


Figure 14.4. A headphone spatialization patch using the Brungart approximate HRTF filter (J. Clark).

Spatial Enhancers and Ambience Recovery

As previously noted, actual HRTFs are very complex and vary from person to person. So they are computationally expensive to implement and usually impractical to acquire. Alternative methods have therefore been sought. One of these alternatives is to perform *ambience recovery*. In ambience recovery the stereo sound field is widened by emphasizing the directional information contained in each channel. This directional information is primarily contained within the channel difference (L-R) signal. Because of this, we can increase the spaciousness of the sound simply by amplifying the L-R signal. The following patch illustrates implementation of this simple ambience recovery technique on the Nord Modular. It is similar to the Koss "Phase" brand of headphones, based on a design by Jacob Turner (US Patent 3924072) in 1974. The amount of crossfeed is set by Knob 1. An extreme clockwise setting removes most of the sound, leaving only the ambient information. An intermediate setting is best.

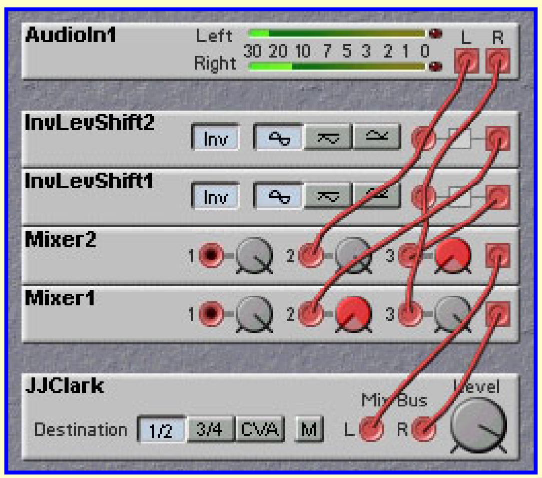


Figure 14.5. An ambience recovery patch (J. Clark).

Reverberation and Spatialization

An important cue for spatialization is the presence of reflected sounds. In an enclosed environment the sound that reaches your ears from a sound source comes not only via the direct path from the source to your ear, but also via indirect pathways corresponding to reflections of the source sound waves off of objects in the environment. Since the precise timing and frequency response characteristics of these secondary (and tertiary...) sounds depend on the location of the sound source relative to the reflecting objects, and on the location of these objects to the listener, the reflections give addition cues as to localization. Modeling the reverberation effects is not just a simple matter of passing the source signal through a reverb unit and adding the result to the direct signal. The reflections will come from different directions and will thus have different time delays, amplitude, and frequency dependent variations. These position dependent effects are not modeled with standard reverb units. In any event, modeling the reflected components is beyond the current capabilities of the Nord Modular, as it lacks delay modules with the required delay times.

Spectral Diffusion

A popular sound processing technique is the "chorus" effect, whereby a single sound, usually rather plain and anemic, is phase- or pitch-shifted by a small time-varying amount and added to its original form. This gives the effect of having multiple independent sources of that sound being added together. The result is a much fuller and richer sound. We can obtain a similar effect with respect to spatialization. A mono source can be spread over the perceived space by causing various components of the sound to be positioned at different points in space. For example, one can apply a bank of narrow bandpass filters to the sound and place the outputs of each filter at different spatial locations (through simple panning, or through more complex localization schemes). This is demonstrated in the following Nord Modular patch, designed by *Keith Crosley*, in which a set of bandpass filter modules is used to decomposed the source signal, and the filter outputs are either panned hard left or hard right.

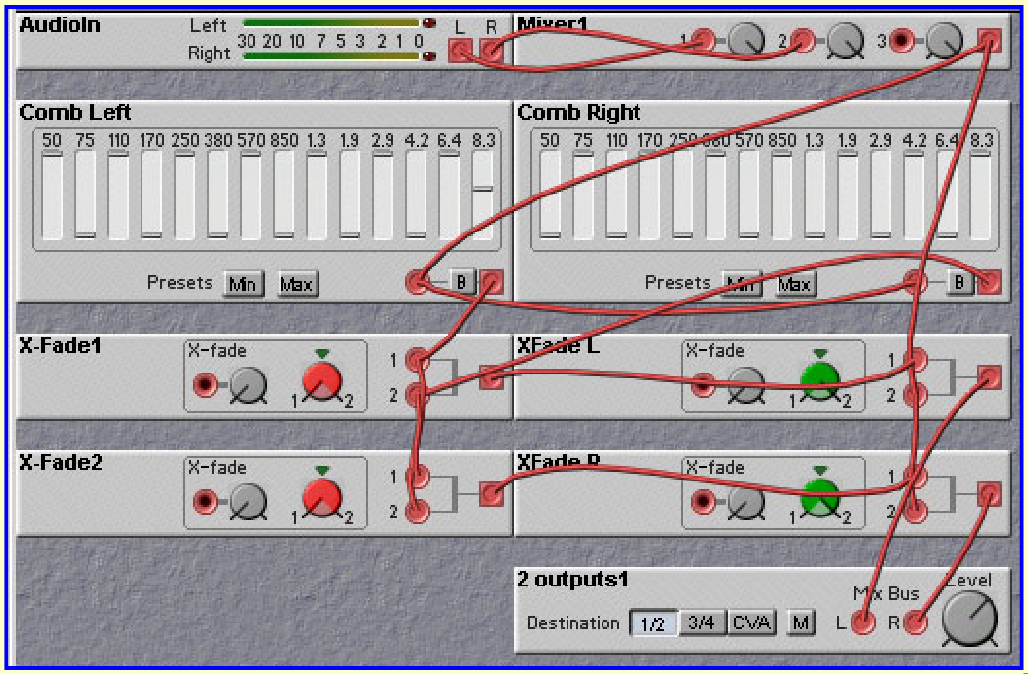


Figure 14.6. A patch that provides a wide stereo field through spectral diffusion (K. Crosley).

If you toggle between the dry mono sound and the spatialized sound you can immediately notice that this spatialization technique results in a much richer and fuller sound, similar in some ways to the chorus effect. Indeed, one can improve upon this approach by adding in time-delay cues to the panning, and applying chorusing to the individual components. This is done in the next patch. The resulting sound is even richer. Note the usage of the poly area. We can make use of this to increase the polyphony since the spatialization can (and should) be shared amongst the different voices.

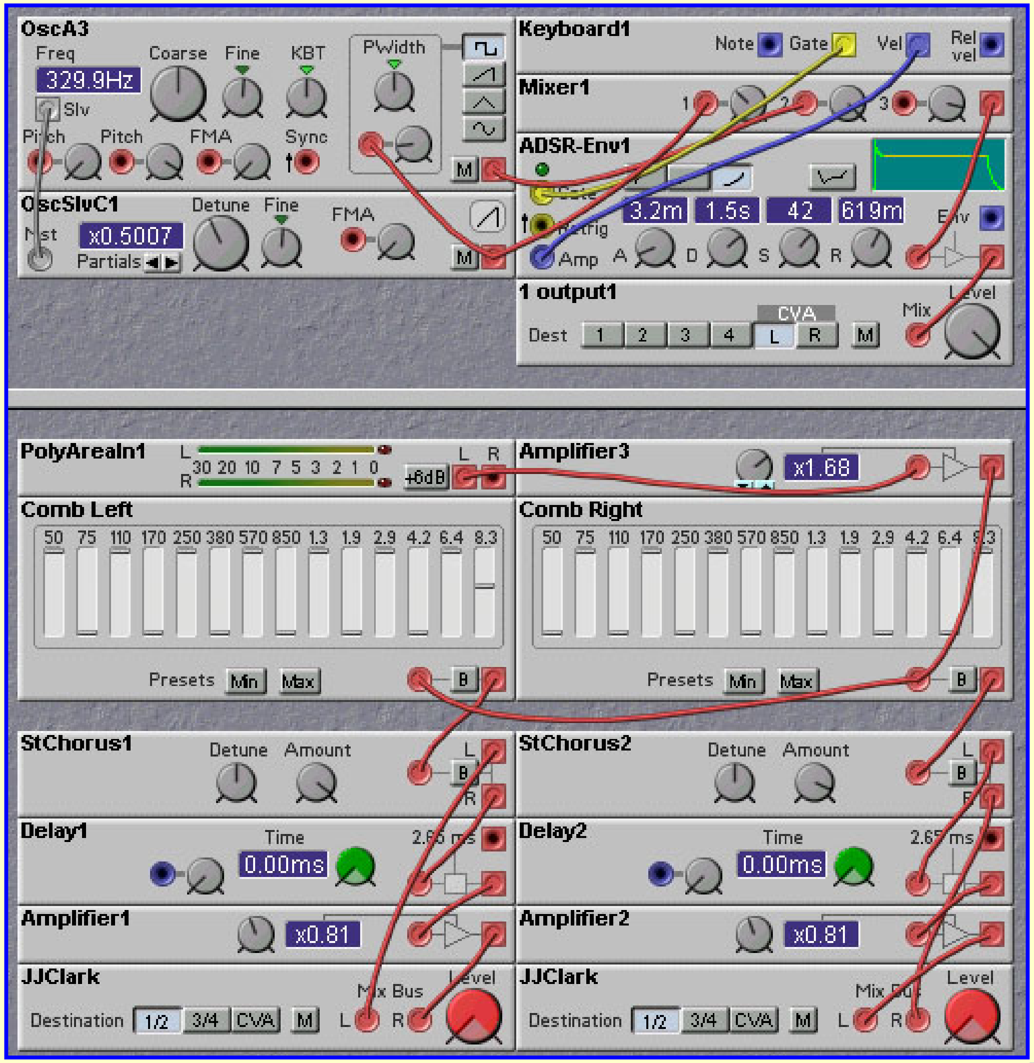


Figure 14.7. Spectral diffusion with time delay cues (J. Clark).

[Prev (Algorithmic Composition)][Next (Emulating Classic Synths)] [Table of Contents]

Chapter 15. Emulating Classic Synths

Every synth that has ever been made has at one time or another be forced to try and make a sound like that produced by some other synth. No matter what the synth, or what the manufacturers sonic intentions, somebody is always trying to get it to sound like a Minimoog! Usually the results are disappointing, but with very flexible synths such as the Nord Modular such expectations are very high. There are those that say emulating other synths on the Nord Modular is a fools errand, and people are better off getting the synth they really want, and use the Nord Modular for the unique sounds that only it can produce. Don't listen to them! Who can afford all of the synths whose sounds you want, anyway? In this chapter we provide a taste of the many and varied efforts that people have made to provide emulations of various popular (and some mythical) synths.

15.1 General Guidelines

by Sam Streeper

You don't really have much control over control ranges (knob ranges, LFO and osc ranges) so don't worry about those, let the tuner find something that works and then who cares if it's accurate. As far as the filters, your choices are rather limited (ie do you want the D, E or F filter) and then since I almost never choose the F one, it's just a matter of picking whether you like the sound of the D or E one (and maybe if you need the features of the E one...) so you really can't get too far away from the nord character. Fortunately these filters are very good.

I think the most important thing really is just getting the routings right, if the signal flows through all the right things and you have all the right modulations available, you're going to be about as close as you can get and you ought to have a usable and fun synth in any case. Mixers are easily the most useful module, since they can gather all the modulation sources and set their levels prior to modulating pitch or whatever.

A few things that might not be totally obvious. For example, the Octave Cat is a 2 VCO synth with 1 LFO and 2 envelope generators (ADSR and AD). However, to emulate it, I had to build a 7 oscillator synth with 2 LFO's and 2 ADSRs, just so all the suitable signals were always present in the same way. But a synth like this is one heck of an instrument, especially if you break from some of the limitations of an accurate emulation (tune the oscs separately, add modulations etc).

It is useful to use master and slave oscs, and you should note a couple things that might not be obvious. A master will propagate its pitch modulations to its slaves, but not its FM modulations, so you have to FM modulate the slaves the same way with similar signal routings as the master. Also, when a master osc is sync'd it doesn't sync the slaves, so you have to do that routing too, though not all slaves have a sync input.

Some of the old monosynths would have made superior polysynths, which makes them fun to emulate. You can do "what-if" exercises, such as "what-if" my ARP Odyssey had a vocal tract filter.

Finally, the one thing that you can't easily copy is the instability and signal overshoots of VCO's, it's helpful to remember that the nord oscs are perfect in shape and stability, so you might want to take steps to compensate, like mixing in noise, or modulating with a bit of noise or adding FM from an unpredictable source. AM can be useful here too.

15.2 Yamaha DX7

by Wout Blommers

Wout Blommers has undertaken an extensive project to convert a vast library of DX7 sounds to the Nord Modular. In so doing, he wrote a nice description of FM synthesis and an explanation of how each of the DX7 "algorithms can be mapped to the Nord Modular. His writeups are available in the chapter [DX7 to NordModular](#).

The results of the conversion project, some 9400 different Yamaha DX7 patches, can be found at http://nordsynth.zevv.nl/010_NordModular/011_Patches/FM_for_Xmas/.

An example of one of these converted patches is shown in the figure below:

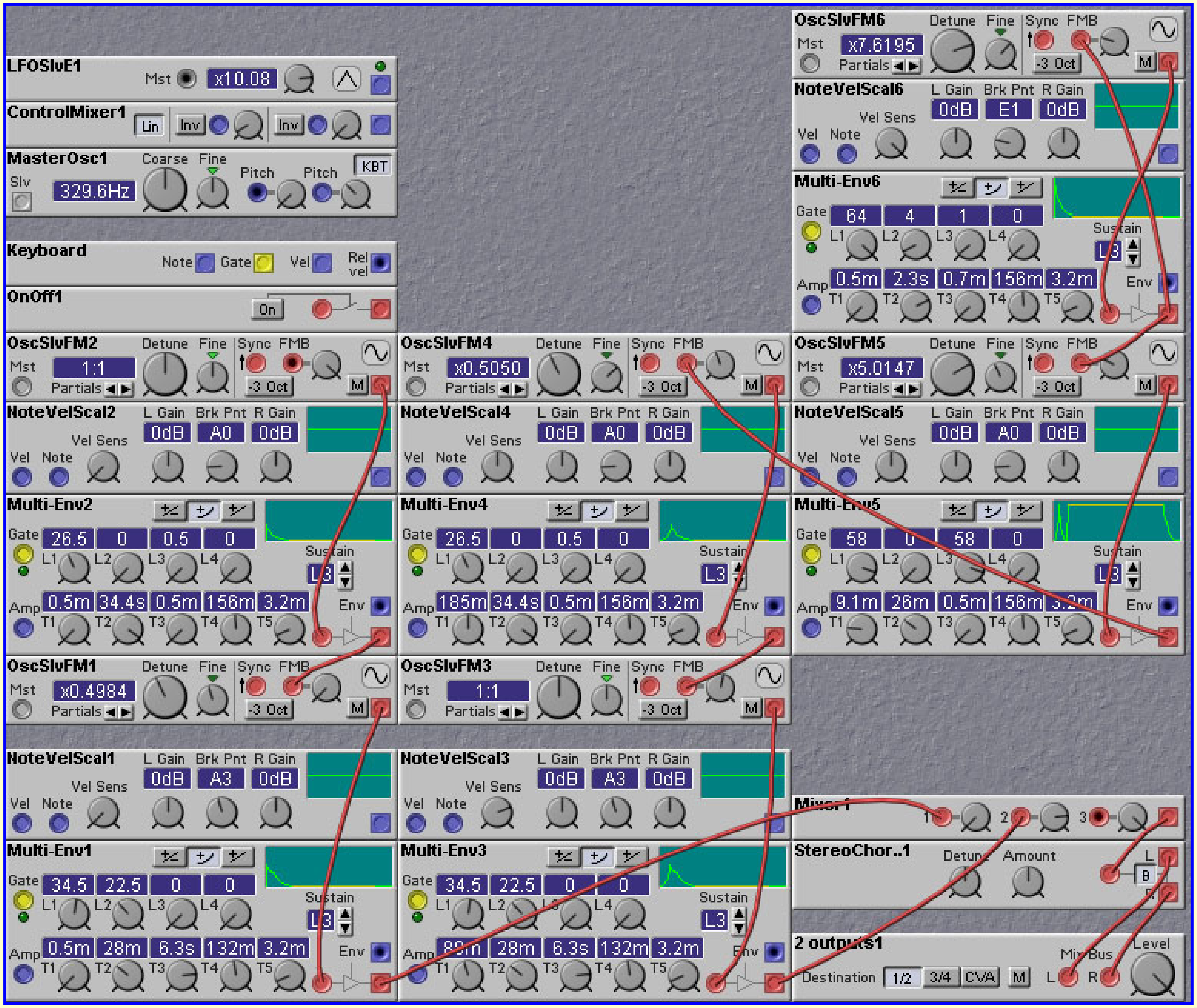


Figure 15.1. A Yamaha DX7 patch converted to the Nord Modular (W. Blommers).

15.3 Octave Cat

by Sam Streeper

This patch mimics the architecture of the old Octave Cat analogue synth, my favorite monosynth. As much as possible, I also tried to duplicate the sound and feel of that synth.

Editor Conventions
I named all the interesting modules in this synth. If there are no symbols preceding the name, either I am indifferent about editing the parameters in this module in the editor, or the values are important enough that I exported their control to an external controller. If I put a '+' symbol before the name, that indicates a module that has values that I want to encourage myself to tweak in the editor, like significant or interesting modulations or routings that I didn't export. In the case of the switches a '+' usually indicates that I intend to try different routings, but the mix levels in the switch may be specifically set and I generally won't mess with those. If a module name is preceded by a '!', that indicates a structural element or logic setup that is carefully set (a necessary routing or a test) that I generally don't intend to mess with. In other words, these tweaking these modules is more likely to break the patch than to provide interesting modulations. I don't think I have any such discouraged modules in this Moctave Cat. In this document if I refer to the Cat I mean the Octave Cat, not the Moctave Cat.

Transient Generator section
The ADSR amounts are controlled by the sliders on my QS8 keyboard (my master controller), which are set up as controllers 12, 13, 91, and 93 respectively. The Attack/Release (AR) sliders are not exported. They can be set in the "+AR" module, which looks suspiciously like an ADSR envelope. The Cat's "ADSR repeat" function is not exported. It can be set in the editor between off and gated in the "+ADSR repeat" module. The additional non-gated auto-repeat functionality of the Cat can be achieved by decoupling the envelopes' gates from the voice gate and holding it high. The Cat's sample and hold switch is not exported, but it can be set in the editor with the "+S&H source" module. It samples either VCO1 or noise, and is triggered by the LFO.

Lower Left section
The duties of the pitch slider are relegated to the pitch controller. The Cat's octave shift button is not implemented as I have a big keyboard. The Cat's glide (portamento) slider is not implemented in the Moctave Cat; it should be there but I haven't figured out the right way to do it yet. The LFO frequency is exported on knob 1. The Cat's LFO supplies a square wave (this is also used to trigger the ADSR repeat if enabled) and a triangle. In the Moctave, I substituted a sine wave for the triangle, since it gives a better sounding pulse width modulation and smoother vibrato and otherwise doesn't matter on my Cat patches. I think the original Cat designers would have used a sine wave here if they could have generated it inexpensively.

VCO1 section
Coarse tuning is on knob 4, fine tuning is not exported but can done in the editor by tuning the VCO1 master osc and the 3 slaves. The slaves are not phased accurately to the real Cat which means you can't really do that goofy Cat note-sequencing thing accurately, but I've always thought that that's more interesting than useful. As long as the phases are wrong, I slightly detuned the slaves because I like that sound better. The Cat's VCO1 keyboard control switch isn't fully implemented. If you bump the Moctave polyphony down to 1 voice, then the Moctave emulates the Cat's mono mode, and if you turn off keyboard tracking in the "VCO1 pulse" module, it's like turning this switch off. I don't thing the Nord can really emulate the Cat's weird poly mode, since that would require a bit of logic processing on the uncooked keyboard status and the Nord really only forwards processed keys. The VCO1 modulation routing switches are not exported, but they can be set in the editor. The first switch selects LFO sine (my default), LFO square, or sample&hold as the modulator and is labeled "+VCO1 mod1". The second switch selects ADSR, the AR envelope, or VCO2 (default) as the modulator, and this one is labeled "+VCO1 mod2". The Cat has a single knob to select the amount of modulation for each of the mod sources, and these knobs are in the "VCO1 modamt" module and exported as knobs 2 and 5. The VCO1 mod amount module also mixes in a bit of weather modulation since my VCO1 is not entirely stable. I did not have enough free modules to accurately model the effects of weather on my Cat though! The Cat allows you to either set the amount of VCO1 pulse width modulation or to dial in the width of the pulse. This functionality is not exported but is found on the "+PulseMod switch" module. Unlike most of my switches, I do need to set the knobs on this module corresponding to the selected switch. Finally, VCO1 supplies 4 waveforms: a suboctave square wave (1 octave down), a sawtooth, a triangle, and a pulse. You set the mix level for each of these waveforms on knobs 3, 6, 9, and 12. These mix levels determine both the

sound of VCO1 and the degree of FM modulation of each VCO1 waveform on VCO2 if you have dialed in FM on VCO2. (That's knob 11 and the default selection of switch 3 on the "VCO2 mod2" module for those of you who have read ahead...) Similarly, the levels of the VCO2 waveforms will affect the amount and sound of FM in this VCO1 section if you left the default routing alone and dialed up knob 5. By the way, if the knob assignments aren't obvious (and I don't expect that they should be yet) there's a simple organization to them that I will explain near the end ofthis document.

VCO2 section
VCO2 coarse tuning is exported on knob 7. VCO1 can be synced to VCO2 with the switch in the "+Sync VCO1" module, though this doesn't really sound like the Cat's sync, and I haven't looked into why. On the Cat, you can get a nice percussive sound by using AR as the main envelope and using a quick ADSR envelope with 0 sustain to modulate the synced pitch of VCO1. I haven't tried this yet on the Moctave Cat, though. This section largely mirrors the VCO1 section, so "+VCO2 mod1" selects LFO sine, LFO square, or S&H as a VCO2 modulator, and "+VCO2 mod2" selects ADSR, AR, or VCO1 as a modulator, and the mod levels are set in "VCO2 modamt" and exported as knobs 8 and 11. VCO2 supplies 3 waveforms: suboctave square, square, and saw, and their mix levels are exported as knobs 10, 13, and 16.

Filter section
I use the Nord's filter D for the Cat filter, since this one is closest to my Cat's sound. By the way, my Cat is the original model with discreet components, later Cats like the SRM have completely different filters. The Nord's D filter is the one that comes closest to the Cat's high end sizzle, though its frequency range doesn't go nearly high enough; The Cat filter has a magical sound as it emerges from super-audible frequencies that no digital filter that I have heard can match. Also the D filter has the closest behaviour in that magical hi-Q zone near oscillation, but again a digital filter and midi controller really can't do justice to an analogue filter and a pot. Finally, at hi-Q, the D filter thins the bass frequencies the least and again, this behaviour is closer to the Cat sound than the other Nord filters. There is a bummer to filter D, though; Its modulator input is control rate rather than audio rate, which cruds up the filter if you attempt to FM it, and filter FM is yet another key to the Cat sound. I don't know what to do about this yet. Note that I left an E filter in there as well; If you dial that one in with the "+Filter picker" module you can get a decent Minimoog-like bass if you dial the other parameters in right. The Cat switch for VCO1 audio out is not exported but can be set by turning off knob 1 in the "+VCF Input Mix" module; this turns off its audio without disabling its FM abilities. Filter keyboard tracking is not exported but can be set in the "VCF" module. Like the VCOs, there are two switches in the filter section for mod routings, and they are not exported. "+Filter Mod1" selects between LFO sine(default), LFO square, and S&H to modulate the filter cutoff frequency, and "+Filter Mod2" selects between ADSR, AR, and VCO1 (default) to do the same. On the Cat, you can get some great nasty but still musically useful sounds by FM'ing the filter frequency. A good modification for the Cat would be to use an envelope to just FM the filter frequency on the attack. The filter modulation amounts are set in the "Filter Mods" module, exported as knobs 14 and 17. This module also includes a non-Cat mod routing: the third mixer sets the amount that ADSR modulates the cutoff frequency and this level is assigned to the main modulation controller (my mod wheel). This allows me to use both FM and ADSR on the filter, which I always wanted on the Cat. Finally, filter frequency and resonance (Q) are exported on knobs15 and 18.

VCA section
Still reading? Well we're almost done since the Cat is almost out of knobs, and the Nord is completely out of knobs... The switch for amplitude level is not exported, but it can be set in the "+AEnv choose" module. The selections are ADSR, AR, and bypass, and by default I use the ADSR envelope to determine the output amplitude. On the Cat, you often use AR as the amp envelope when you are using ADSR for effects like a filter sweep or transient pitch weirdness. Bypass mode makes for a great drone. And for the final knob, I don't export the Cat's noise control but you can set it with knob 3 in "+VCF Input Mix". A worthwhile modification to the Cat would be to route the noise through an envelope, but that's not how the Cat does it.

Knob organization
There's a method to the knob organization that makes it reasonably easy to remember, especially if you're used to the real Cat. The first vertical block of 6 knobs is loosely dedicated to VCO1, the second block to VCO2, and the third block to the filter. The middle row is dedicated to modulators, so by default knobs 2 and 5 modulate VCO1 by the LFO and FM respectively, and knobs 8 and 11 do the same thing for VCO2, and knobs 14 and 17 do the same thing for the filter. The 4 knobs in the lower left select the output levels for the 4 VCO1 waveforms, and the upper right 3 knobs do the same thing for the 3 VCO2 waveforms. That leaves us with 3 knobs in the upper left which are LFO speed, VCO1 tuning and VCO2 tuning, and 2 knobs in the lower right which are filter cutoff and resonance, just where they are on the Cat.

Interesting modifications
A lot of the Cat's nasty sound comes from all those rough edges in the waveforms. You can smooth the Cat a good deal by changing the suboscillators to sine waves. This will lose a bunch of high frequencies from the sound; it can sound very good but it won't sound like a Cat. Many of the weirder Cat sounds are had when you do bi-directional FM, where VCO1 modulates VCO2 and vice-versa. You can route a signal for FM into either an oscillators pitch input or its FM input. These inputs have very different ranges and somewhat different responses, so I let you pick whichever sound works better for you; pick the ones you like in each of the two "+FM type"modules.

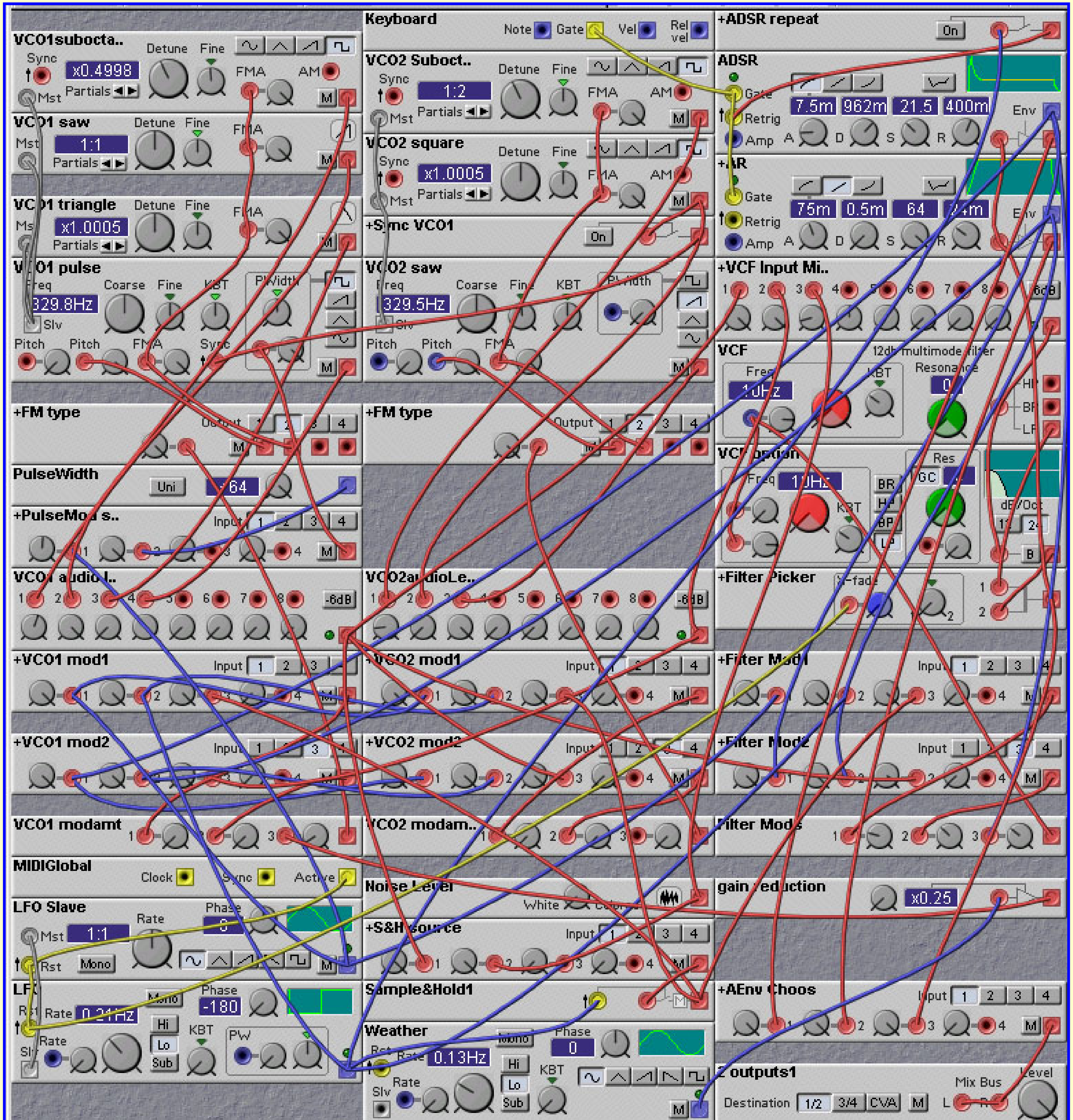

VCO1subocta..
Sync
x0.4998
Detune
Fine
FMA
AM
Mst
Partials
VCO1 saw
1:1
VCO1 triangle
x1.0005
VCO1 pulse
Freq
329.8Hz
Coarse
KBT
PWidth
Slv
Pitch
Sync
+FM type
Output
PulseWidth
Uni
64
+PulseMod s..
Input
VCO1 audio l..
-6dB
+VCO1 mod1
+VCO1 mod2
VCO1 modamt
MIDIGlobal
Clock
Sync
Active
LFO Slave
Rate
Phase
Rst
Mono
LFO
0.21Hz
-180
Hi
Lo
Sub
PW
Keyboard
Note
Gate
Vel
Rel vel
VCO2 Suboct..
1:2
VCO2 square
x1.0005
+Sync VCO1
On
VCO2 saw
329.5Hz
VCO2audioLe..
+VCO2 mod1
+VCO2 mod2
VCO2 modam..
Noise Level
White
+S&H source
Sample&Hold1
Weather
0.13Hz
0
+ADSR repeat
ADSR
Gate
Retrig
Amp
7.5m
962m
21.5
400m
Env
+AR
75m
0.5m
64
+VCF Input Mi..
6dB
VCF
12db multimode filter
Resonance
HP
BP
LP
1.0Hz
Res
GC
dB/Oct
12
24
BR
+Filter Picker
X-fade
+Filter Mod1
+Filter Mod2
Filter Mods
gain reduction
x0.25
+AEnv Choos
2 outputs1
Destination
1/2
3/4
CVA
Mix Bus
Level

Figure 15.2. A Nord Modular emulation of the Octave Cat (S. Streeper).

15.4 Arp Odyssey

by Dave Peck

The first synth I ever bought was an Arp Odyssey, in 1973. Although the Minimoog was more popular, and although many feel the Mini had a better sounding filter, I felt the Odyssey's versatility more than made up for any sonic shortcomings. The Odyssey had pulse width modulation, oscillator hard sync, a ring modulator, and a versatile sample/hold section, none of which were included in the Minimoog. Also, the sample/hold section had a dedicated input mixer, and the output of this mixer could be used for audio-range modulation of the filter, also not possible on the Minimoog.

I sold my original white Odyssey to finance the purchase of bigger synths, and then bought another early black face Odyssey several years later during a sudden attack of gear nostalgia. I've used it from time to time, but maintenance is an issue, as it is with all vintage analog synths. Thus, it seemed like a good idea to replicate the Odyssey in the Nord Modular.

The Odyssey used an ingenious system of patch switches to quickly select a variety of patching configurations. This was a good compromise between versatility and speed, without the hassles of patch cords or the limitations of a completely hard-wired synth. My goal was to replicate the Odyssey as completely and accurately as possible, including the method used to set up a patch. This meant devising a basic patch architecture that included patch switches instead of requiring the user to draw patch cords.

On the Virtual Odyssey, the patch switches are represented by the switches in the 4-to-1 switch modules, and the names for these modules show the available switch selections. For example, the module labeled "NG/RING MOD" selects between the noise generator and the ring modulator as the input to the first knob in the audio mixer.

Thus, by simply selecting switch positions and turning virtual knobs, you can replicate any sound that was possible on the real Arp Odyssey, without drawing any patch cords.

In fact, you can make sounds that were not possible on the real Odyssey, because some parameters on the NM have a greater range than the equivalent parameter in the real Odyssey. For example, the LFO can run slower and the envelope release times can be longer than the same parameters on the real Odyssey.

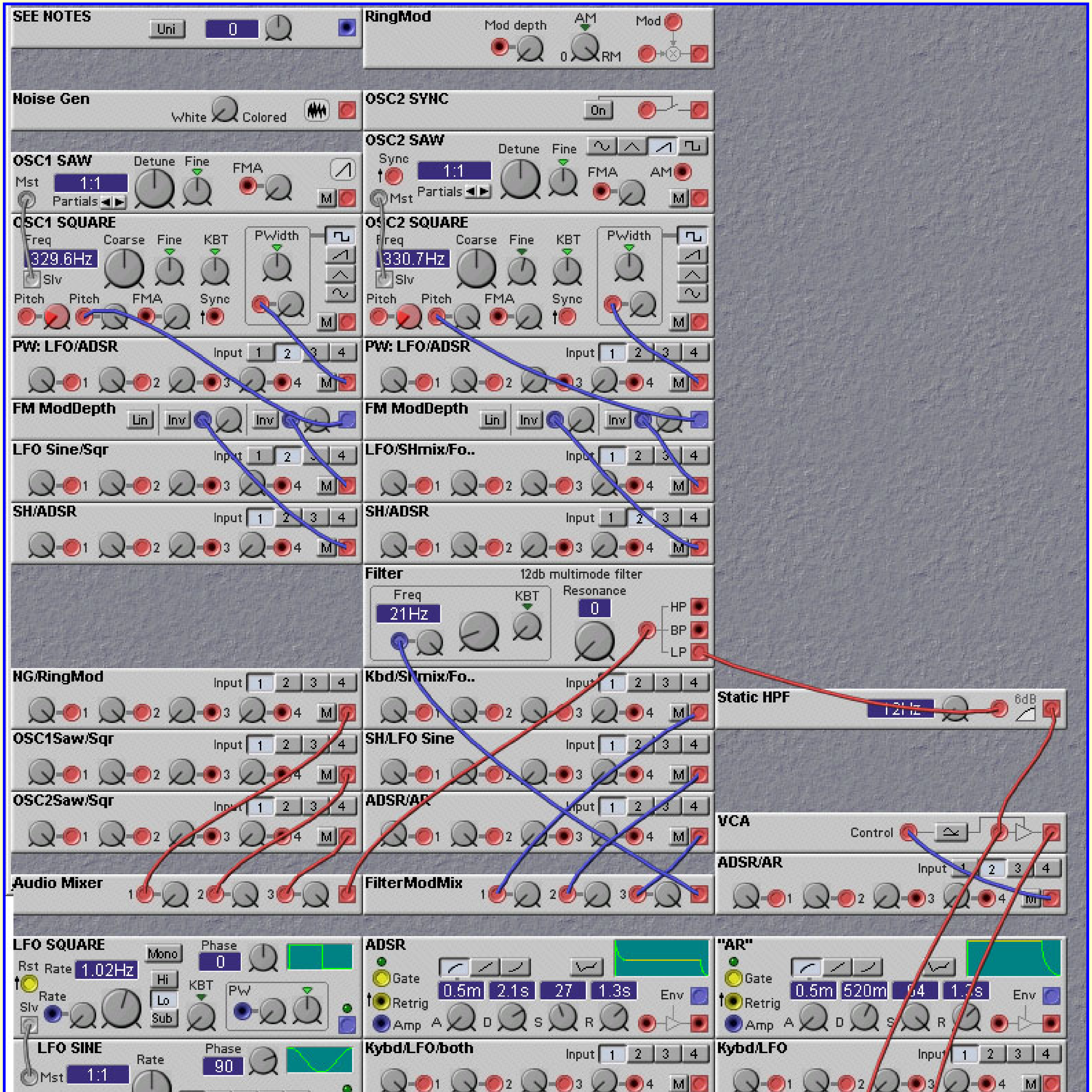
There are some modules which are used to process the Nord's signals to more closely resemble the signals on the Odyssey. The waveforms in the Odyssey's audio oscillators and LFO were positive-going only, unlike the bipolar waveforms in the Nord Modular. Thus, a collection of Level Shifter/Inverter modules were used to alter the Nord's waveforms. Although an audio oscillator sounds the same whether it is bipolar or unipolar, it makes a difference when you use these waveforms as modulation sources.

Since the Odyssey had more sliders than the Nord has knobs (I think the Odyssey has 32 sliders), I opted not to assign any parameters to the Nord's front panel. I'll leave it up to the user to assign them as needed when setting up a patch.

The real Odyssey had one major feature that I was not able to replicate on the Nord. The Odyssey had a unique "duophonic" keyboard. When you pressed two keys, OSC1 played the low note and OSC2 played the high note. This was useful for producing distortion effects when using the ring modulator, among other things. Note that this is not the same as simply setting the Nord to have two voices. If you get ambitious, please feel free to try to figure out how to achieve the duophonic effect.

One minor difference between the Nord and the Odyssey is the way the filter responds when resonance is set to maximum. On the Nord, you must send some amount of audio into the filter to trigger filter self-oscillation. This was not necessary on the real Odyssey.

The one feature I added, which is not true to the original Odyssey voice architecture, is the additional Vibrato LFO, which is controlled by the mod wheel of a midi controller keyboard (via the red morph group). The Odyssey had really lame performance controls, and I just had to do something about it. I hope the purists are not too offended :-)



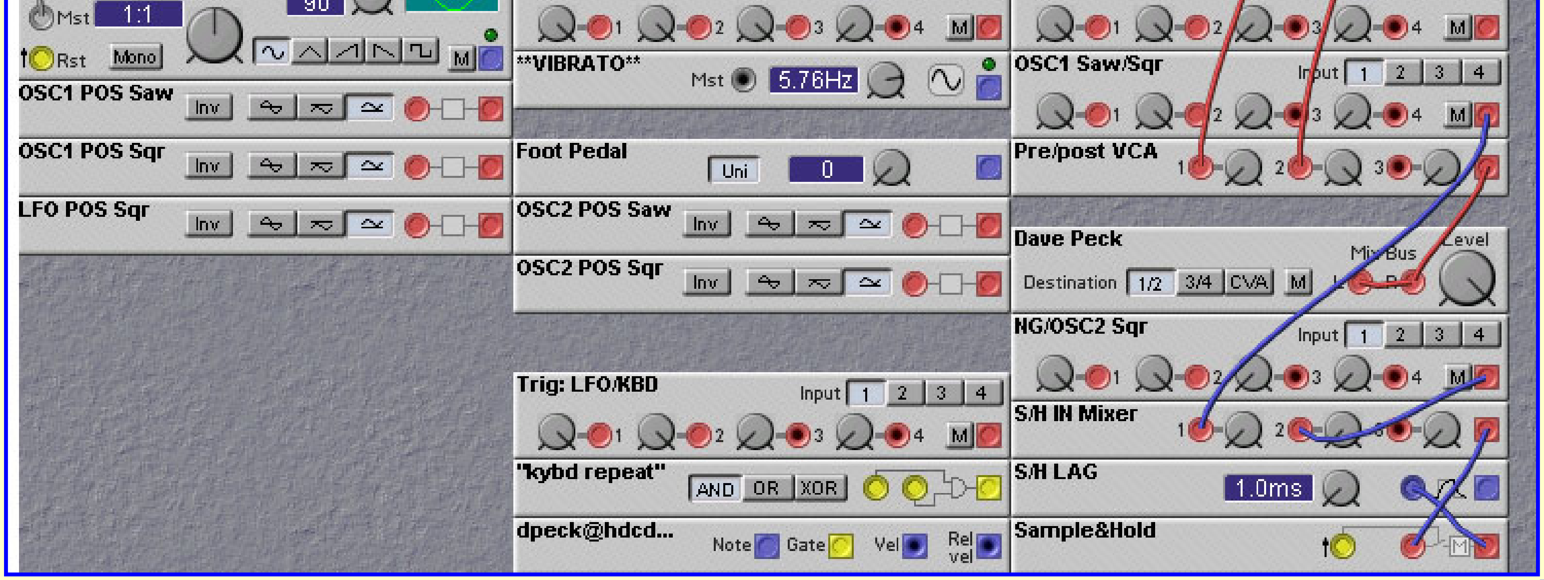


Figure 15.3. A Nord Modular emulation of the Arp Odyssey (D. Peck).

Knobs in 4-to-1 switch modules should stay at MAX. Switches in 4-to-1 modules act as the patch switches across the bottom of the panel on the real Odyssey. Module names in the 4-to-1 modules indicate switch selections. Example - the switch module below the VCA is labeled "ADSR/AR". Button 1 selects ADSR, button 2 selects AR to control the VCA. GREEN cables - audio signals routed to patch switches (HIDE). PURPLE cables - control signals routed to patch switches (HIDE). RED cables - hardwired audio signals BLUE cables - hardwired control signals YELLOW cables - gate & trigger signals (HIDE).

Some example patches illustrating the use of this template are:
OdyXWarmXPWM.pch
OdyXTrillXDrone.pch
OdyXSquareLead.pch
OdyXFilterXPatterns.pch
OdyXFilterFX.pch

15.5 Casio CZ Series

by Rob Hordijk

Just looked up Phase distortion in Curtis Roads's excellent computer music tutorial - and there is this bit on the subject: "Phase distortion synthesis is a term invented by the casio corporation to describe a simple modulation technique developed for several of its digital synthesizers. PD synthesis uses a sine wave table lookup oscillator in which the rate of scanning through the oscillator varies over the cycle. The scanning interval speeds from 0 to PI and then slows down from PI to 2PI. The overall frequency is constant, according to the pitch of the note, but the output wave form is no longer a sine....." I think Casio used two (digitally simple and DSP cheap) tablescan oscillators with both the same basic frequency and synced to each other. Then when adding a bit of the output to the tableindex of the other the phasedistortion occurs. This would mean a sineoscillator's output to an FMB input on a SlaveFM oscillator, setting both to the same frequency and syncing the SlaveFM to the sineoscillator. But I remember vaguely from the frontpanel of a CZ-100 (it was called that way, wasn't it? only knew one guy that had one of those Casio's in the eighties) that it could be done with other waveforms than a sine as well. Anyway, I like to do it as in the attached patch. Using the multiwave slave gives a more harsh sync-like sound, more like the CZ-100 as I remember. The sinevalue here is not added to the 'tableindex' as with the SlaveFM (FMB), but lineair to the frequencyvalue (FMA), so not proper I suppose. But what's proper anyway? Additionally the AM input on the slave oscillator is used as a ringmodulator, suppressing small glitches that occur at the beginning and end of a waveperiod on the NM's oscillators when using sync, due to waveform interpolation (beginning) and FMA modulation (end).

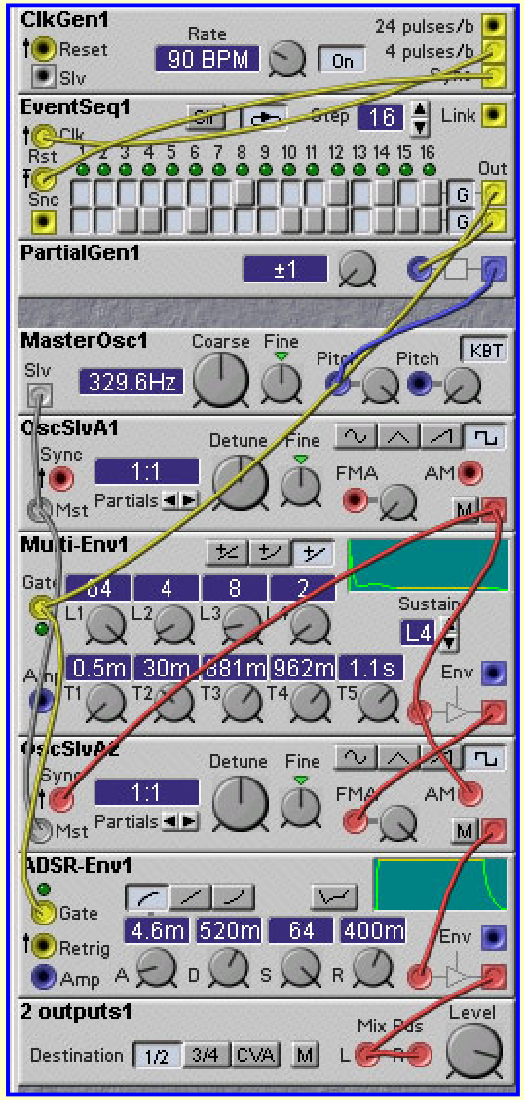


Figure 15.4. A Nord Modular emulation of the Casio Phase-Distortion synthesis technique (R. Hordijk).

15.6 Hallsey Mark 1 and 2

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by Wout Blommers

In the late sixties and early seventies my musical friends and I were really POOR! Most of the gear was fake... The bassplayer could paint the Gibson lettering so well, he's now a typographical designer. In those days we build our own Synths, yes sir! For the Hallsey Mark 1 we used the oscillator of our Farfisa organ. The Synth sound was made by distortion and it sounded like distortion...

The first try-out we did was during a gig and the sound was so terrible, the organization of the party threw us out! After that failure we build the Hallsey Mark 2. (The name Hallsey was chosen 'cause our bassman made MARSHALL on two plates, MARS did fell off the fake amplifier and we added 'sey') The Mark 2 was a real modular synth, only its oscillators wouldn't tune in line... (It had sample&hold, mixers, LFO's and all that stuff) The Mark 2 was stolen during a gig and some days later our leadsinger found the poor thing at the side of a road, 'cause the thieves couldn't make any sense out of it, me think.

Mark 3, of course also from the famous 'Hallsey factory', was, so I believe, one of the first Synths which used AD and DA converters. Two of our gang worked at the university, where such electronic equipment was developed for Philips. (No, they didn't steal them: it was just an university, you know, where the professors also carry uranium 238 in their pockets) I think only the Mark 1 is interesting to emulate here ;-) Played in the lower regions, it was a wonderful beatbox, but we didn't knew then... (Press the low C and hold it a while) Those Shapers were really there. The LFO's in a way too. And the distortion remains!

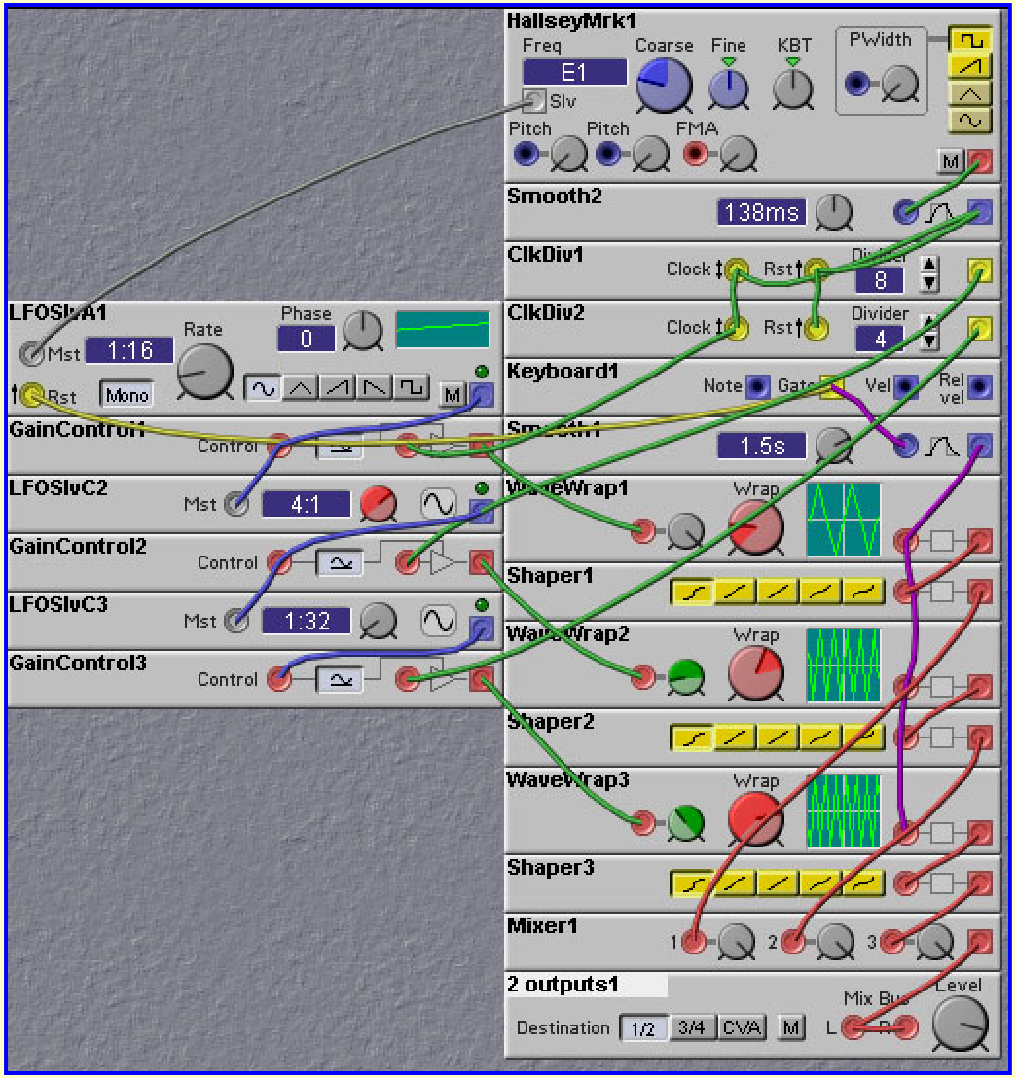

Figure 15.5. A Nord Modular emulation of the famous Hallsey Mark 1 (W. Blommers).

15.7 Other Emulations

Emulating favorite synthesizers seems to be a popular pastime with Nord Modular owners. Here is a list of some vintage synthesizer emulation patches.

ARP 2600, by *Roland Kuit* (this is just a basic template - Roland has created many more patches using this template).
ARP Omni, by *Andy "Buckshot"*
ARP Omni, by *Rob Hordijk*
ARP Axxe, by *Kevin Thomas*
ARP Axxe, by *Mr. Marko*
ARP Solina, by *Bergmuller*
Roland Juno, by *Roland Kuit*
Roland Juno-6, by *Bergmuller*
Roland Juno-2, by *Kendall Jackman*
Korg Electribe EA-1, by *Douglas R. Kraul*
Korg Mono/Poly, by *Rob Hordijk*
Korg MS-20, by *Wan Kemper*
Korg MS-2000, by *Rob Hordijk*
Korg Polysix, by *Bergmuller*
Korg Poly61, by *Keyboardslover*
Korg Poly800, by *Klaus Ludwig/Peter Gorges*
Moog Minimoog, by *Bill Barton* (there are a number of other patches available based on this template)
Moog Rogue, by *James Maier*
Roland Jupiter-8, by *Bergmuller*
Roland JX-8P, by *Roland Kuit*
Roland SH-101, by *Rob Hordijk*
Roland SH-101, by *Bergmuller*
Sequential Circuits Prophet-600, by *Bergmuller*
Sequential Circuits Sixtrak, by *Rikard Latvala*
SID, by *Andreas Wickman*
SID, by *Rob Hordijk*
Yamaha CS-40M, by *Bergmuller*

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