AN ALGORITHMIC EDITOR FOR THE DX7

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OVERVIEW

A technique to enable simple and effective editing of sounds produced by FM Synthesis on a Yamaha DX7 Synthesiser is described. The software techniques enable complex modification of existing sounds via MIDI (the Musical Instrument Digital Interface) without any need for the user to understand Frequency Modulation (FM). The paper reviews the technical, aural, and psychoacoustic elements of the FM Synthesis technique used in the DX7, and goes on to describe the techniques used to analyse the parameters which produce a given sound. Some of the algorithms developed by the author for modifying the FM parameters using the analysed data are then described.

INTRODUCTION

The release of the Yamaha DX7 just over three years ago produced an explosion of activity which continues to grow to this date. The DX7 was the first synthesiser to effectively exploit digital technology to produce sound. It uses FM techniques originally described by Chowning [1] for all of the signal generation in the digital domain, with the only major analogue components being a DAC and the associated filtering. The range and quality of sounds produced by the DX7 has ensured its popularity and it has virtually acquired the status of a 'standard' within the sound recording industry. Unfortunately the DX7 has also gained a reputation for being very difficult to program, since the editing and creation of sounds is both difficult and time-consuming, requiring a consistent and structured approach. This is very different from the intuitive editing commonly used on more conventional analogue synthesisers, using VCOs, VCFs and simple Envelope Generators. Recently a few editor packages have been released which considerably simplify the user controls for FM synthesis. This paper will describe both the analysis of the FM parameters and the subsequent algorithms used to control these parameters in such an editor.

FM Synthesis using the DX7.

The conventional use of FM is for stereo radio transmission, where the instantaneous carrier frequency is at about 100 MHz and the modulator frequency is within the audio range. The paper by Chowning[1], which forms the basis of the Yamaha implementation of FM, describes the mathematics for carriers and modulators in the audio band, especially with respect to producing simulations of musical instruments. One unusual effect that often occurs in audio FM is that the side-bands of the resultant spectrum can have components which fall in the negative frequency domain. These frequencies 'reflect' around 0 Hz and combine with the spectral components in the positive frequency domain, either reinforcing or cancelling depending on the relative phases and thus destroying the symmetry of the sidebands about the carrier. Also, the use of fixed modulator frequencies and non-integer carrier / modulator frequency ratios can further complicate the spectra produced by audio FM. The result is to produce spectra with many frequency components from just a few sine waves.

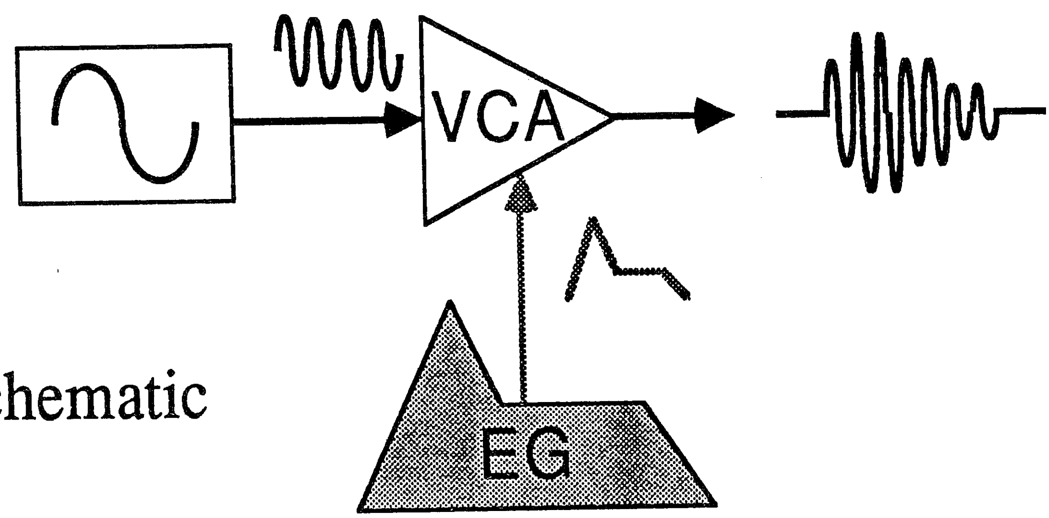
The major control mechanism of FM is superficially very simple, and is called the modulation index, that is: the ratio between the peak carrier deviation and the modulating frequency, usually given the symbol I. As the modulation index increases from zero, the spectrum evolves from a single carrier (at $I = 0$) , to a spectrum with a large number of components for an index of about

10 or 15 - higher indices tend to place the side-bands outside of the audio range, leaving only the carrier in the audible part of the frequency spectrum. Aurally, this evolution results in an increase in the 'sharpness' of the sound, progressing from a smooth-sounding sine wave to a timbre similar in tone to a sawtooth, having a mix of many harmonics. Although the changes in the spectral components are more complex the overall impression is very similar to increasing the frequency control on an analogue VCF. In practice, there can be up to three sets of carrier / modulator pairs or several modulators can drive a single carrier, depending on the arrangement of the carriers and modulators.

With the different arrangements of the available carriers and modulators (there are six available in a DX7), each with its own envelope generator (EG), the manipulation of the modulation indices in time becomes both powerful and complex, even though the basic control mechanism uses a maximum of just 5 separate indices. The use of a computer to make the necessary changes in the large number of parameters contributing to the modulation indices, is the basis of the technique described here.

The current software uses a model of FM based on the author's own experience of editing and programming sounds, and applies algorithms based on both the model and the sound to be edited. In this way the resultant changes to the sound retain some of the character of the original sound wherever possible, although after several changes to different options the edited sound can be considerably altered, as we shall see. This use of feedback from the model and the original sound also ensures that the editor moves sounds away from the original sound, since if you make a change and then attempt to undo it , the edited sound and not the original sound is used when the algorithm is applied, resulting in a different 'undone' sound than the original. In practice, most people tend to use an editor of this form to move away from the sound they started with, and so this divergence is actually useful. If you want to make slight, precise changes to a sound, then an editor which enables single parameter changes is more suitable, but is far more difficult to use.

<u>Analysis</u>
The conversion of the FM parameters which define the sound into a form suitable for processing by computer, and the derivation of the values used in the various algorithms, depends on the perception of the FM synthesis process itself. The basis of FM in a DX7 is the Operator, which can be thought of as the equivalent of a sine wave generator, voltage controlled amplifier (VCA) and envelope generator (EG), residing in a combination of hardware and software in a couple of proprietary VLSI chips.



Operator Schematic

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Each Operator can be used as either a carrier or a modulator depending on the arrangement and inter-connection of the Operators. Confusingly these arrangements are called Algorithms by the manufacturer of the DX7, and consist of 32 different combinations of the six available Operators.

Conventionally the Algorithms are illustrated with the carriers at the bottom, and with the modulators above, in what a Graph Theorist would call a Tree Network. Some of these networks have feedback loops within them, in which an Operator's output is connected to its input, either directly or via one or more additional Operators. The splitting up of the Algorithms into their component carriers and modulators is very important in the analysis, since the same parameter can have different effects on the resultant sound, depending on whether it is associated with a carrier or a modulator. The arrangements are printed on the top of the DX7 and can be readily converted into a look up table mapping the Algorithm to carriers and modulators. To analyse the arrangement of Operators within an Algorithm you first split the Operators into two sets: Operators used as carriers, and Operators used as modulators.

For example, an Algorithm such as number 32, with six Operators all used as carriers, is technically not using FM at all since the resulting sounds are merely the sum of six sine waves. Alternatively, Algorithm 1 has a 'stack' of three modulating Operators, connected in a series arrangement above a single Operator acting as a carrier, in parallel with a simple 'pair' consisting of a modulator and carrier. In the case of the stack of Operators the assignment of carrier and modulator designations is blurred, since the highest Operator modulates the Operator below it, but the resulting output then modulates the Operator below that and so on. This can produce spectra with large numbers of frequency components and very bright, harsh timbres.

Having located the carriers, the Algorithm can be further analysed by considering the outputs of each carrier as a separate part of the sound. Each Operator has individual parameters for frequency and output level of the sine wave, as well as those associated with an 4-rate / 4-level envelope generator. In addition, there are further parameters which act on all the Operators, giving a total of 145 parameters needed to specify a single sound. An extra 23 parameters determine the performance characteristics of the DX7 as a whole. The Output Level parameter determines the size of the waveform leaving an Operator and therefore the effect it has on any subsequent Operators to which it is connected. For simple pairs of carriers and modulators, the Output Level of the modulator is related to the modulation index, and the Output Level of the carrier acts like a volume control. Thus the relative levels of the various sections or parts of an Algorithm can be determined by monitoring just the carrier Output Levels. In the same way the Envelope Generator for a carrier determines the envelope of that section of the whole sound.

For the modulators the situation is more complex. As the Output Level is increased, the modulation index increases, resulting in an increase in the number of harmonics in the sound. Thus the modulator's EG acts as a modulation index control, varying the spectrum in time. The modulation index is also related to the frequency of the modulator Operator, and in conjunction with the carrier frequency, provides a coarse control over the timbre. As a rough guide, the ratio between the carrier and modulator frequencies in a simple pair arrangement, determines the number and spacing of the harmonics. For a 1:1 ratio (that is carrier : modulator) each harmonic of the fundamental will be present for some value of the index, but for a 1:3 ratio some harmonics will not occur, since the resulting spectrum has been stretched laterally. It is also possible to use Operators to produce fixed frequencies, and this can have the effect of fixing harmonic groups in the resultant spectrum, simulating the effects of a resonant structure in a real instrument.

Algorithms

Further analysis is best described with reference to the Algorithms used to alter the FM parameters. In the following sections I will use a form of computer notation known as 'pseudo code' - that is a

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representation of the form of the algorithm , but not expressed in any specific computer language.

As described above, the major control over the harmonic content of a sound is the modulation index. In order to manipulate this it is necessary to determine the carriers and modulators and alter the relevant Operator Output Levels, ie. changing one parameter in up to five Operators. This is both tedious and time consuming. For more complicated changes to sounds a large number of parameters may need to be changed, often many times before the desired result is obtained. Let us examine the commands needed to change the harmonic content or brightness of a sound - a very useful overall tone control.

```
brightness:
    find modulators : if no modulators then end
    input a control delta value
    for each modulator
        note down existing output level
        increment or decrement it by the  control delta value
    end
```

This is the basis of the all the algorithms described here. Although simple it is very effective on most sounds, and can be improved in many ways, for example, an Operator with a fixed carrier or modulator frequency should generally not have its Output Level changed for brightness control, since this affects the spread and balance of the harmonic group associated with it and this changes the timbre or tone colour rather than just the perceived brightness. Many other improvements are possible.

A similarly simple but very useful control is one over the release of notes, ie. what happens once the keys are released. Normally this means finding the carriers and changing the EG Rate 4, which is the parameter determining the time for an EG output to reach its final level. Algorithmically, this can be described as:

```
release:
    find carriers
    input control delta value
    for each carrier
        note down the existing EG Rate 4
        increment or decrement it by the control delta value
    end
```

In a similar way, algorithms can be composed to alter Attack by altering the EG Rate 1 of both the carriers and modulators, or Sustain by altering the relevant parameter. For changes such as these, the parameters do not interact and the algorithms are correspondingly simple. For changes where more than one parameter is altered, or several parameters interact, the algorithms are more complex. For example, one of the main methods of expression on a keyboard instrument is via the velocity sensing or touch sensitivity function, which provides a measure of how 'hard' the key is pressed. The way that the sound changes tone with changes in key velocity is a very significant part of the 'signature' of a sound. The piano being a notable example of an instrument which sounds very false when it merely gets louder instead of brighter when you play the keys harder. In practice, the Velocity Sensitivity of Operators needs to increase as you climb up through the modulators within a stack:

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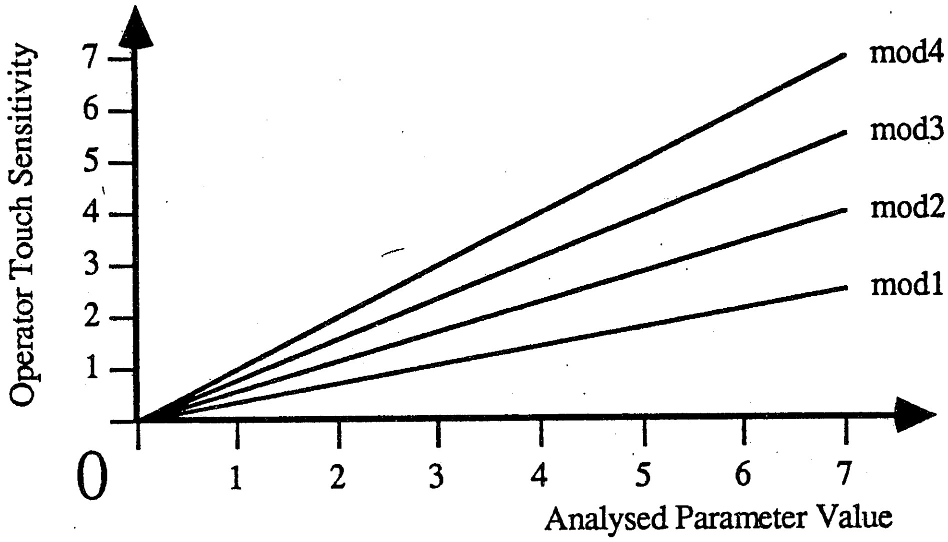
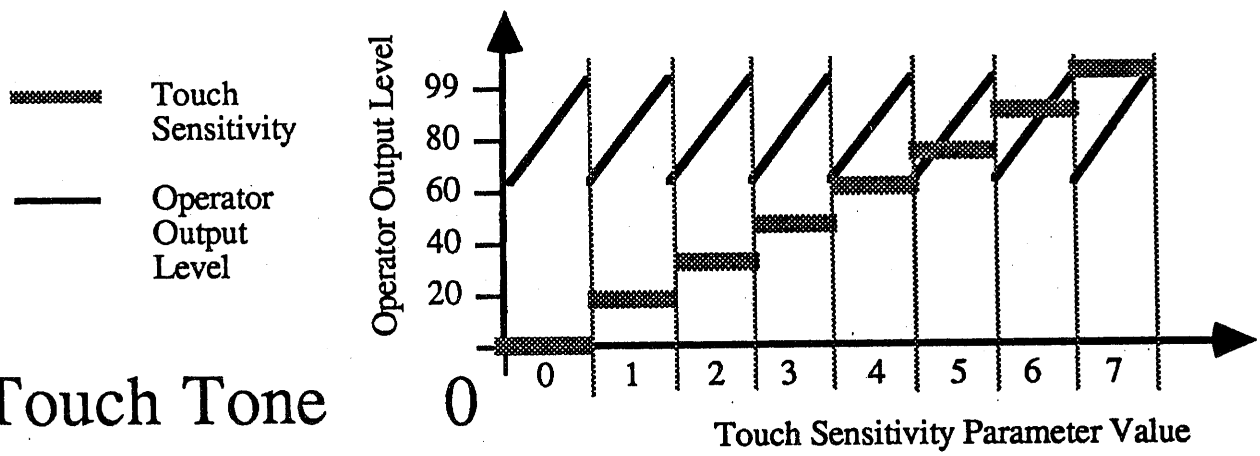
```
touch tone:
    find modulators : if no modulators then end
    input a control delta value
    for each modulator
        assign a weight depending on the vertical position within a stack
        note down the existing velocity sensitivity
        increment or decrement it by a function of the control value scaled by the weight
        note down the existing output level
        increment or decrement it by a function of the control value scaled by its weight
    end
```



Touch Tone

The control value is scaled by a function which relates the velocity sensitivity to the output level - that used by the author varies the output level linearly from 50 to 99 for each velocity sensitivity increment, and this seems to work for most sounds. The 'analysed parameter value' in the graphs shown here is the original value of the parameter before the edit has been made, but scaled and presented to the user as the 'visible' parameter which is being altered.



Touch Tone

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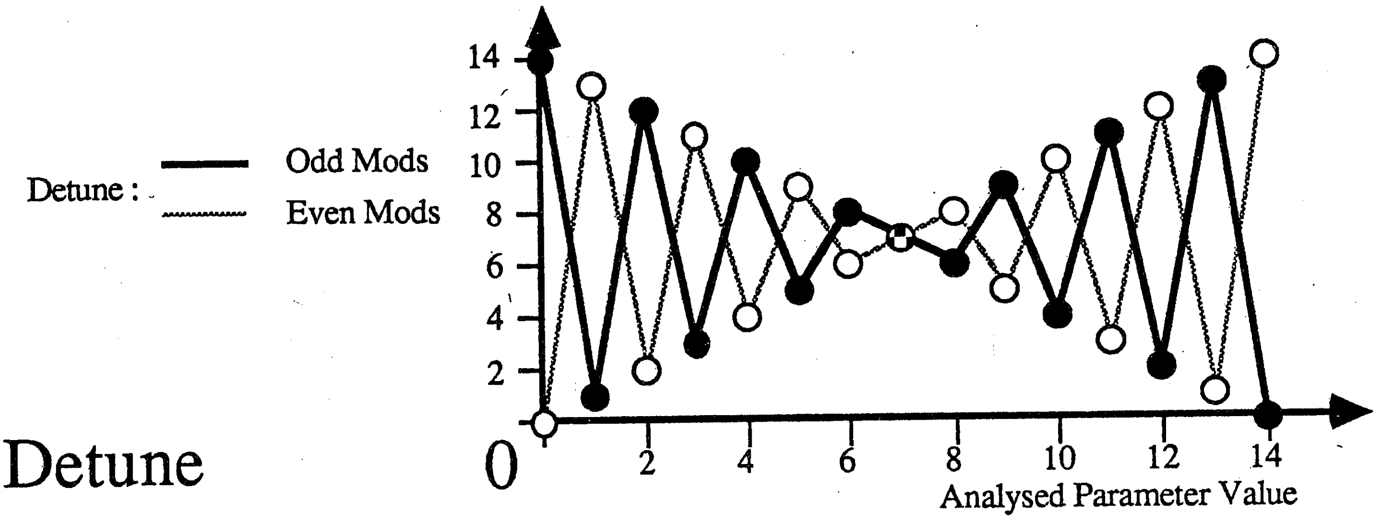
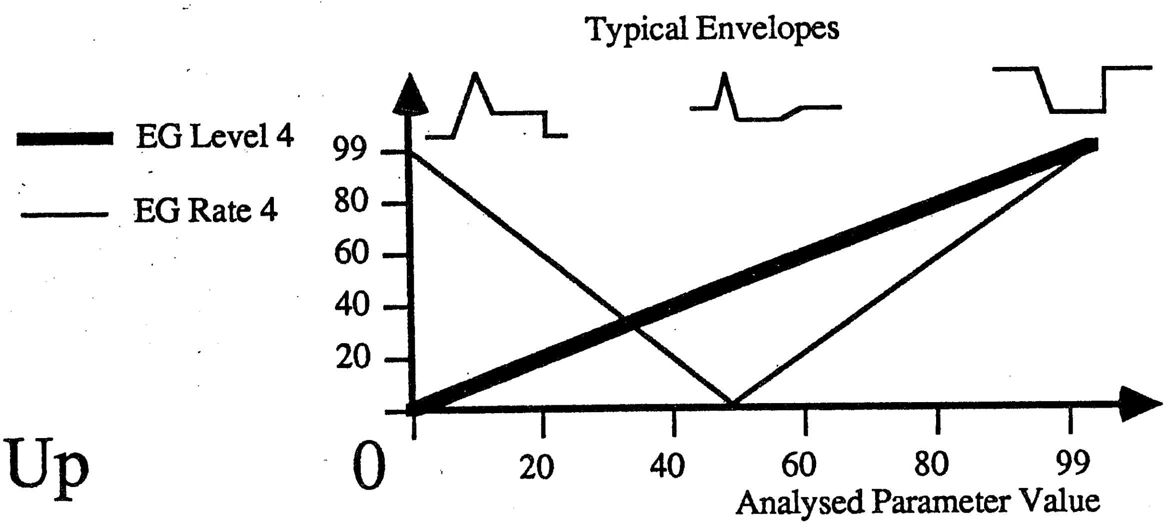
Alternative strategies could take into account the presence of fixed frequencies, which usually need less change with velocity, or could monitor the Operator with feedback, in order to prevent the changes in output level from causing the generation of undesired noise. The point at which the Operator produces noise varies with feedback level, output level and frequency, as well as the number of Operators within the feedback loop.

As the complexity of the algorithms increases, so do the functions required to provide the needed conversions between a single control value as seen by the user, and the actual values to enter as parameter values.



Detune

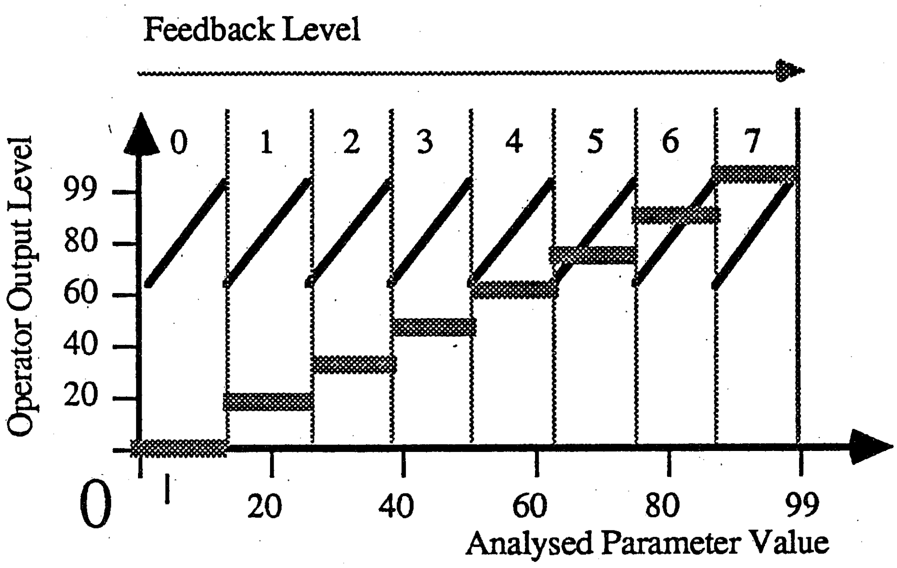
The detune algorithm uses two mutually exclusive bipartite networks to generate a series of values for the detune parameter which increase in their difference as you move away from the centre value of 7. Odd numbered modulators (weighted as above) receive one value, while even numbers receive the other. This alternating arrangement is chosen because it follows the usual form of many 'published' sounds and also seems to work very effectively in most cases, but is obviously capable of expansion and refinement.



Key Up

The 'key up' algorithm uses another technique to process the control value. Two parameters need to be controlled by varying just one user control. The EG Level 4 parameter sets the start and end level for the Envelope Generator, and EG Rate 4 sets the time for the envelope to change from the sustain level (EG Level 3) to the end level: The chosen function allows you to choose EG Level 3 and Rate 4 by providing a restricted set of possibilities, chosen to give useful combinations of level and rate. Typical envelope shapes are shown for three of the portions of the graph:

Feedback



For control over feedback two functions are again controlled, but this time by a discontinuous function. The Output Level of the Operator with Feedback is selectable from 50 to 99 for each of the feedback values of 0 to 7. The modular arithmetic necessary to achieve functions like these is relatively simple to implement on most computers. The algorithms are much the same as before, but with either look-up tables or approximation formulae used to generate the required functions:

```
overdrive:
    find operator with feedback selected
    if no operator with feedback then end
    input control delta value
    note existing values of output level and feedback
    calculate the new values for the parameters based on the control delta value
    change the parameters accordingly
    end
```

CONCLUSION

This paper has described some of the techniques for analysing the FM parameters which define a sound, and then using this information to derive algorithms to produce changes in specific parameters of the sound. In effect, this provides a mapping between two completely different domains: the FM domain contains a large number of difficult to use parameters, whereas the Control domain has a small number of simple commands which produce understandable changes in the sound produced by the FM Synthesiser. The increased complexity of methods of sound production can be greatly simplified by the use of techniques similar to the ones described here, and this will increasingly be necessary in order to effectively use the synthesis systems of the future. I hope that other researchers will continue the work started here, and hopefully increase and improve the algorithms available for this sort of editor software.

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