Adapting DX7 Patches for Additive Synthesis

The following is an approach I have created for translating Yamaha DX7 patches into information I can use for programming my Kawai K5. This should also work for any synthesizer that uses additive synthesis. This approach only works well with certain kinds of patches such as bells, pianos, and plucked sounds. It doesn't seem to work very well with brass and string sounds (at least for the K5). That still leaves quite a few patches that should work. You should also know that while I have had quite a bit of experience creating sounds with FM synthesis using several software programs (some of which imitates the DX7), I don't own a Dx7. So, I haven't been able to do any A/B comparisons. I have, however, used this technique with very good results.

For the sake of brevity, I will assume you know something about FM synthesis (and additive synthesis). If you don't, there are several web pages out there that explain the process. Just type FM synthesis into your web browser and you are sure to come across some of them. You will also need to know something about the DX7's architecture. I will try to help you out there.

Let's begin by "translating" an electric piano patch. I am using this example not because I think the world needs any more e. piano sounds but because this patch is relatively simple to use and because it is probably the sound the DX7 is most famous for. Here is the all the information we need from the e. piano patch:

Algorithm: 5   Feedback: 6

	Envelope:	R1	R2	R3	R4	L1	L2	L3	L4	Level
Op: 6	ratio 1.00	95	29	20	50	99	95	0	0	79
Op: 5	ratio 1.00	95	20	20	50	99	95	0	0	98
Op: 4	ratio 1.00	95	29	20	50	99	95	0	0	89
Op: 3	ratio 1.00	95	20	20	50	96	99	0	0	99
Op: 2	ratio 11.00	95	35	35	78	99	75	0	0	58
Op: 1	ratio 1.00	96	25	25	67	99	75	0	0	99
Pitch: 3		99	99	99	99	50	50	50	50	

First, let's look at the algorithm this patch uses (the DX7 has 32 algorithms):

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Algorithm 5:       M2    M4    M6_|
                    |     |     |
                   C1____C3____C5
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Each "M" is a modulator and each "C" is a carrier. The number besides each carrier and modulator represents it's operator number. All DX7 algorithms have 6 operators. The lines going around operator 6 is my attempt to show that this operator uses feedback. When hearing a DX7 patch, you are only listening to the carriers.

Now, let's look at the ratios between each of the carriers and it's modulator.

C1 and M2 have a ratio of 1:11 (the original patch has the ratio at 1:14 but I changed this to 1:11 because it sounded better to me). This generates a bell like tone. C3 and M4 have a ratio of 1:1. This produces the body of the sound. C5 and M6 also have a ratio of 1:1. I assume this is to add detuning to the sound.

This is where the work really begins. Beginning with C1 and M2, we will calculate the partials these two operators generate. In order to do this, we will use the following formulas:

C, C+M, C+2M, C+3M, C+4M, etc.
C, C-M, C-2m, C-3m, C-4m, etc.

Since the ratio between C1 and M2 is 1:11, the results are as follows:

1,12,23,34,45,56
1,-10,-21,-32,-43,-54 (ignore the negative signs)

In other words, for C1 and M2 we have partials 1,10,12,21,23,32,34,43,45,54,56. You could continue to calculate higher partials but since the K5 only goes up to 63 (in twin mode), I stopped at 56.

Using the same formulas to calculate the partials for C3 and M4 (1:1), we get the following numbers:

1,2,3,4,5,etc.
1,0,-1,-2,-3,-4,-5,etc. (ignore the zero and the negative signs)

Thus, we have the partials 1,2,3,4,5,6, etc.

Since C5 and M6 have the same ratio as C3 and M4, we will ignore them.

What do we do with all these numbers? Well, before programming your synth, you could take a look at the level settings (0-99) of each operator. If a carrier has a high level setting then you know the partials it generates will have high amplitudes (it also depends on how much it is being modulated). If a modulator has a high level setting, it generally means the sound it generates through the carrier will be bright (the feedback setting should be taken into account as well). Both of these factors should be considered when determining how high to set the amplitude of the individual partials. To get a feel for how the level of the partials change over time, you should look at the envelope settings. If you are familiar with the envelopes on the K5, you should have no trouble interpreting the DX7's envelopes. Finally, look at the pitch envelope to see how it affects the sound.

Unfortunately, I don't know how to calculate the amplitudes of the partials based on how much a carrier is being modulated (some of you may know how to do this). Generally speaking, the partials that are generated tend to decrease in amplitude the higher you go in frequency. Just think of a saw wave and you will know what I am talking about. Since we know what an e. piano is supposed to sound like, it isn't hard to make an educated guess as to how interpret the numbers.

With all that said, we really already have what we are looking for;

namely, the partials generated by this patch.

Starting with the body of the sound, we will use the results of C3 and M4 (partials 1,2,3,4,5, etc.). Excluding the bell sound, the rest of the sound is not very bright. We really only need partials 1 through 6.

Set the level of the first six partials as follows:

1: 99 (max)
2: 88
3: 81
4: 72
5: 57
6: 37

These settings are for the K5; however, you shouldn't have any trouble adapting them for any synth that uses additive synthesis.

What we have sounds like a highly filtered saw wave.

For the envelopes, I have partials 1 and 2 assigned to envelope 1. I programmed it to have a simple decay typical of a piano with a short release. Partials 3 through 6 are assigned to envelope 2. This is set like envelope 1 only with faster decay times.

Now, on to the bell part of the sound.

As you may recall, the partials generated by C1 and M2 were 1,10,12,21,23,32,34,43,45,54,56. I set them to the following levels:

1: We have already set this partial.
10: 34
12: 75
21: 33
23: 74
32: 32
34: 72
43: 31
45: 71
54: 30
56: 70

Again, I set the levels of each partial based more on experimentation than on any formula.

The "bell" like sound decays much faster than the body of the sound, so I set the envelopes to have a faster decay rate.

I assigned envelope 3 to partial 10 and 12, the rest I assigned to envelope 4.

At this point, the rest is window dressing. You could have the filter set up to respond to how hard you play a key, detune the two sources to create a chorus effect, rate scale the overall VCA so higher notes decay more quickly, etc.

Many patches use algorithms more complex than algorithm 5. They require greater calculations and a bit more finesse to make work. If there is enough interest in what I have written so far, I may give more examples.

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