FM Synthesis of Real Instruments (and any other type)

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1. An Introduction to FM

Back in the early 70's in an electronic audio lab at Stanford University John Chowning was experimenting with the university's analog synthesizers. He noticed something odd occurring while playing with vibrato on the instrument. When he turned the vibrato speed way up the sound stabilized and produced a set of harmonic tones around the main note. At specific settings the sound was harmonic and clean. He realized this was a way to produce very complex sounds using only two oscillators. Prior to this it took dozens of oscillators to do the same thing. Frequency Modulation Synthesis was born. He went deeply into the theory and developed the mathematics needed to produce the sounds digitally, and in 1973 he published his famous article on Frequency Modulation. In 1977 Yamaha licensed the technology and came out with the first programmable FM synthesizers in 1983, including the infamous DX7 which took the music industry by storm.

The DX7 sold over 200,000 units and is still considered the most iconic FM synthesizer in existence. In recent years it's been replicated on computers in VST's and is in the hands of millions of users. Despite over 30 years of use, the techniques used to program FM are still a mystery to most people. Yamaha surveyed DX7 owners and found that only about a hundred of those original DX7 users could program an instrument patch from scratch. Even the original Yamaha factory patches are a bit lacking. I own a DX11 which came out in 1987 and an FS1R from 1998, and only a hand full of the factory patches are very accurate to the real instruments they're named after. The user generated patch libraries aren't any better. Very few of the patches actually sound like the instruments they're meant to duplicate.

Searching the Internet for tutorials on how to program FM is also fruitless. There are plenty of primers explaining the basics of FM, but none go beyond that point to explain how to use FM to duplicate a real instrument. The only conclusion you can come to is that very few people understand how Frequency Modulation really works. Most users simply tweak an existing patch without ever really understanding what's really going on under the hood.

Is FM really so complicated that nobody can figure it out?

No, it's not! And we're going to prove it!

2. FM Basics

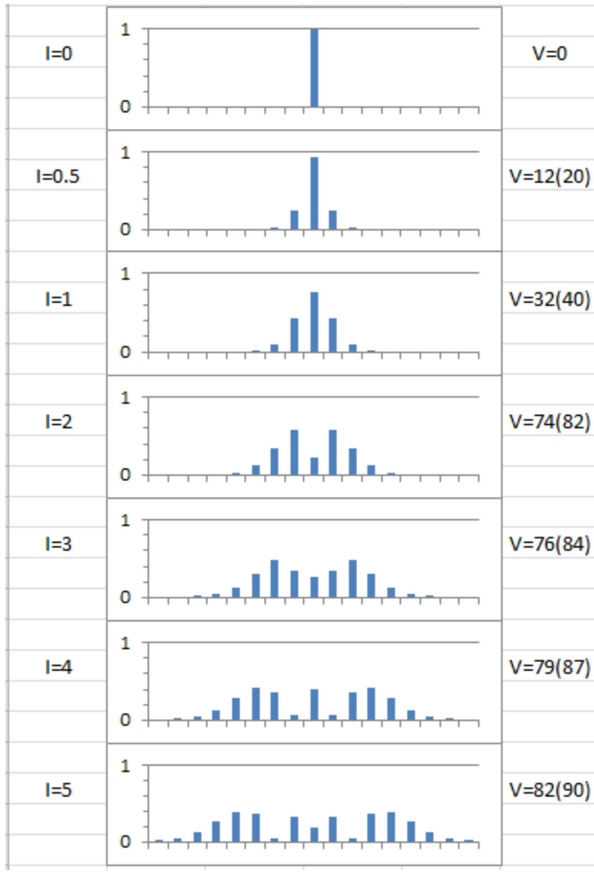
Frequency Modulation is really just vibrato on steroids. You start with a Carrier oscillator which determines the pitch of the note that is heard. The Carrier is altered by applying very high speed vibrato (modulation) using a Modulation oscillator. The result is that you hear multiple harmonics popping up on both sides of the main note. Increasing the amplitude of the vibrato increases the effect, creating more and more harmonics in the sound.

When you use modulation frequencies that are integer multiples of the Carrier frequency, you get melodic harmonics that work well together in duplicating the sounds of instruments like a flute or a violin. If you use a modulation frequency that is a fractional multiple of the Carrier you get inharmonic sounds which can be used to duplicate bells or to create dissonant sounds. Slightly inharmonic ratios are used to detune parts of the sound to replicate the timbre of a piano or a sitar. If you want to get fancy you can stack on more modulators to really complicate the harmonic structure, and you can add together multiple sets of carriers and modulators to further increase the complexity of the sound.

In FM the frequencies of the Carrier (Fc) and Modulator (Fm) are represented by the ratio of their frequency to the fundamental frequency of the note being played. In most of our examples at least one carrier will be set to a frequency ratio of 1.00, meaning the carrier is at the frequency of the key being pressed on the piano keyboard. In the graphs below, our modulator will be at a frequency ratio of 1.00 which spaces out the harmonics by multiples of the fundamental frequency. The first, second, third, and fourth harmonic frequencies (shown below) can be calculated by adding the frequency of the modulator to multiples of the carrier frequency.

F1 = Fc   The Fundamental Harmonic, #1
F2 = Fc + Fm   (and also F2 = Fc - Fm for the lower side band)
F3 = Fc + 2*Fm
F4 = Fc + 3*Fm

The graphs below show what happens to the output of the Carrier Operator as you increase the volume of the Modulator. The Carrier frequency in our example below is in the middle of the blue peaks at harmonic 9.00. The Modulator causes frequency spikes to form to the sides of the Carrier frequency as you increase the volume of the modulator. The spacing between the harmonics is equal to the Modulation frequency. The modulator is set to 1.00 below, spacing the spikes 1.00 apart. If this sounds confusing, just be patient and we'll clear all this up in a few more minutes.



Volume=DX11(DX7)

Figure 1. Frequency Modulation of a Carrier

On the plots in Figure 1 you can also see that as we increase the volume of the Modulator the center Carrier frequency spike gets shorter and shorter, along with the ones right next to it. So far the Operations we've discussed are done using sine waves. We're going to leave it there and pick up other wave shapes later to avoid too much complication.

The Index (I) on the plots above refers to the work of John Chowning, the inventor of Frequency Modulation Synthesis. He came up with the definition of a Modulation Index which is what the 'I=' terms refer to on the chart. On a DX synthesizer the Index (I) is adjusted using the <u>volume</u> of the modulator. The values for Volume on DX synths are in the right-most column of Figure 1, DX11(DX7). In the rest of our discussion we're mostly going to use the right half of the modulator/carrier plots. The left side will fall below 0 hz where it doesn't affect us much. Well, it does affect us somewhat, but let's just keep things simple for now and ignore the left half of the patterns.

This is the point where most tutorials stop. They give you a very brief intro, then drop the ball or lose you in a detailed explanation of the mathematics involved. We don't really want to know how to do the math, what we really want to know is how to USE the stuff.

3. Instrument Duplication

There are a couple of reasons why I chose real instruments as the target for this guide. The first is that there are lots of recordings out there of real instruments like the violin playing single notes, so there are lots of recordings for you to practice making instrument patches on. The second is that there are very few high quality FM instrument patches out there for these instruments. Why not fill in the gap? I do realize most people like FM because it allows you the freedom to create custom instruments. And, I know a lot of those people want strange new sounds, but not everyone wants to create a space-zombie pad with a trance beat to it. Some of us like to modify real instruments instead.

The method we're going to use to replicate a real instrument is not such a mystery after you've done it a few times, so that's what we're going to do. We're going to go step by step through a number of examples, and hopefully by the time we're done you'll know how to do it yourself. This technique applies to more than just real instruments and voices, you can replicate just about anything you can get a clean recording of.

In short, you start with a recording of the instrument (or sound) you want to duplicate playing a single note all by itself, then you load the sound sample into a spectrum analyzer like 'Audacity' (free on the internet) so you can get a good look at the instrument's harmonic structure. Using the harmonic spectrum plot, you formulate a plan of attack knowing the tools FM provides you with, selecting one of the algorithms on your synthesizer. This will get you in the ball park. You then listen to the results and make adjustments by ear to your instrument patch by comparing the synthesized sound with the recording over and over. You can also use a real-time spectrum analyzer like 'Visual Analyzer' (free on the internet) to assist in this, allowing you to compare the synthesized sound with the spectrum plot so you can see visually where to make changes in the settings. I've actually had more success at creating a better patch by listening as I make the adjustments, using a spectrum analyzer visually all by itself doesn't work out that well.

The process is basically the same for all FM synthesizers. Our examples are going to go over three families of synths, the DX11 family which includes DX21, DX27, DX11, TX81Z, FB01, YS100, YS200, Reface-DX, and all other four Operator VST synths, and the DX7 family which includes the DX1, DX5, DX9, TX802, TX816, SY77, TG77, SY99 and a growing number of six Operator VST synths. The third synth family we'll cover includes the eight Operator FS1R's and the new Montage FM-X engine. FM-X really expands on the capabilities of FM with some very useful harmonic waveforms. You Montage guys might want to skip ahead to that section, then come back to cover the basics of FM afterwards. I have a DX11 and an FS1R with FM-X. I don't have a DX7, but the FS1R can duplicate anything the DX7 can do so don't get too concerned if your synth is based on the DX7.

Tools you need:

Audacity: https://sourceforge.net/projects/audacity/

Visual Analyzer: http://www.sillanumsoft.org/

I use two different spectrum analyzer programs, at times so I can have them both open at the same time. I use Audacity to open a recording and display a spectrogram to work from, then Visual Analyzer to watch the notes as I play them on the synthesizer. You can compare the live spectrum plot to the static one and make adjustments while listening to the sound until you get it close. Audacity has a tendency to average out the higher frequency parts of the sound, so be aware of that failing.

You can also create static spectrograms using Visual Analyzer which are much more detailed than Audacity. On the main interface, look at the mini control panel and find the "Main" tab, then look for the "Hold" checkbox. Just below it is a setting for averaging the plot over several samples. I like to use 2 or 3 averaged samples to sharpen the peaks in the plot. Check the "Hold" box then play a note on your input keyboard to trigger the spectrogram. Screen-capture the plot so you can use it later. To clear the plot, uncheck the "Hold" box. The problem here is figuring out how to route the sound you want to analyze into Visual Analyzer, it only likes live input channels. It's easy to route in a synthesizer in your studio, but it takes more thinking (possibly another computer or loop-back device) to play recordings and route them in your main computer for analysis.

4. Spectrum Analysis

The method we're going to use to replicate an instrument is based on replicating the harmonic structure. We start the process by obtaining several recordings of the instrument in question. There are numerous places on the internet where you can find recordings of real instruments playing single notes. If you are duplicating another type of instrument or sound, you just need to find a good clean recording to work from. I found some nice collections of instrument recordings on the following websites:

http://www.philharmonia.co.uk/explore/make_music

http://theremin.music.uiowa.edu/MIS.html

http://www.compositiontoday.com/sound_bank/default.asp

Another source of high quality recordings is in the synthesizer next to you. The samples in my Motif-XF are superb, and it's simple to pipe the sound into my computer into the spectrum analyzer.

Select a sample where the instrument plays one or more clear notes all by itself. Select a note to model somewhere in the middle of the voice range of the instrument. Instruments with a resonating body like a Violin or Bassoon are harder to model, so it's best to start in the middle of the voice range. We'll talk about this in more detail in the section on Formants. If the recording is muddled on the spectrum analyzer and doesn't show clear-cut frequency spikes, select a different recording to work from.

After attempting this several times, you'll realize that it doesn't always work out using the very first recorded sound sample or strategy. There can be dramatic differences between sound samples… and even between different notes in the same sound sample. Sometimes it takes several attempts before you capture the essence of the instrument. Having more than one sound sample to work from is essential if you want to succeed at replicating a specific instrument. You can also cheat and start from the patches I put together in the Appendix.

The next step is to spectrum analyze the sound. Open the recording up in a sound editing program like 'Audacity', highlight one note in the sound sample, and open it up in the spectrum analyzer (menu item Analyze/PlotSpectrum). You may need to increase the resolution of the sample in order to see the frequency spikes clearly. I use sample Size: 2048. Stretch the window out horizontally as far as possible so you can see the individual peaks on the left side of the plot easier.

As you move the cursor across the plot a vertical line will snap from peak to peak. In the numbers below the plot you can read off the numerical value for each frequency spike. On the chart below I've used a green arrow to show the cursor line and circled the frequency of the peak it's pointing to.

On the Clarinet plot below I've labeled the peaks 1, 2, 3, 4, etc. according to which harmonic it represents. Peak number four will be 4 times the frequency of peak number one, and so on. We'll be using these ratios when we program the synthesizer in a few minutes. On a 'Linear Frequency' plot like the one below all of the harmonics will be equally spaced, the same distance apart as peak 1 is from the left edge of the plot (half of a grid square). I picked a Clarinet here on purpose so you could see an example where one of the harmonics is so low (#2) that you almost can't see it. The sound of the Clarinet comes primarily from the loudest frequency spikes, 1, 3, 5, and number 7. Pay attention to the even spacing!

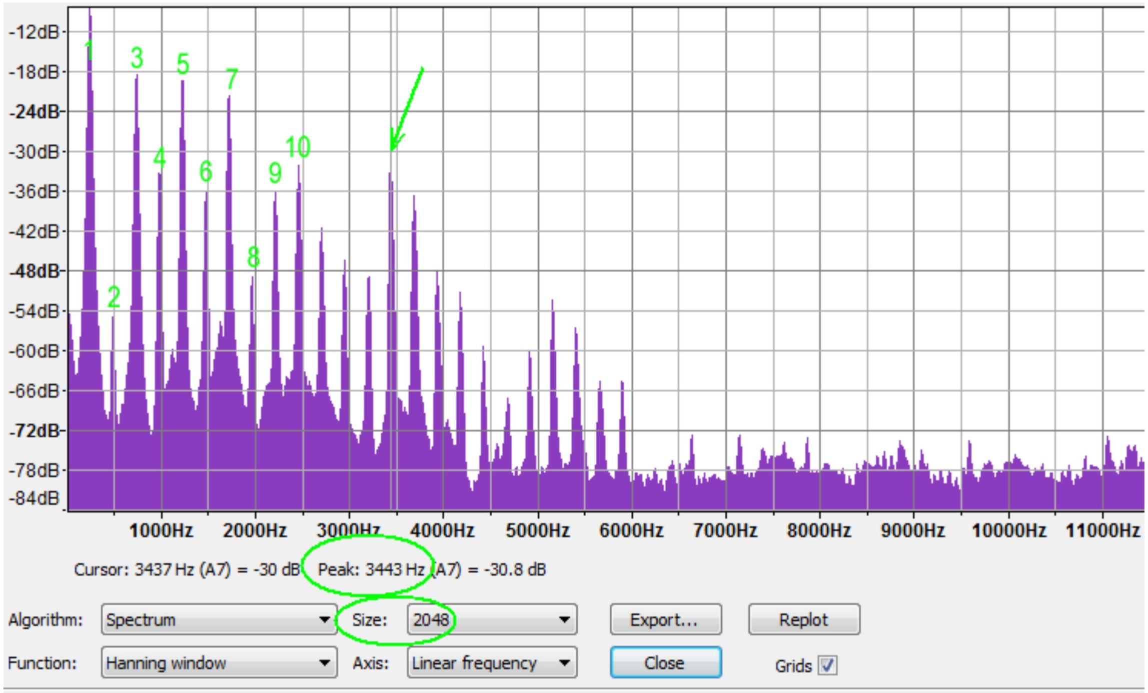


Figure 2. Clarinet Harmonic Spectrum

On the plot above, the scale on the left side is in decibels (db) which is logarithmic. The volume scaling in our instruments is also logarithmic, so it's reasonable to assume we can scale the volumes of each harmonic off the chart above. Place the tallest peak #1 at a setting of 99 (the maximum volume). Peak #3 is at about 80% of full scale so scale it to a volume of 80, and so on. The drop in volume between peak #1 to peak #2 is about 10 db, which means harmonic #2 sounds about half the volume of harmonic #1 to your ears. Peak #10 is down about 25 db from peak #1 so it sounds less than 1/4 as loud as peak #1. Played in the background along with harmonics 1, 3, 5 and 7 you can just hear #10. Peaks with an amplitude lower than about -70 db can be ignored since they can't really be heard anyway.

To get the sound exactly right we would need to duplicate harmonics 4, 6, 9, and 10 as well. We haven't got enough capability to do this on a 4-operator synth, but we should be able to do it on a 6 or 8-operator synth. A 4-operator synth like the DX11 can get 90% of the way there, which actually sounds pretty good.

5. Single-Stack Operators

Selecting an Algorithm is the next step in the process. Below are pictured the algorithms for the DX11 family 4-operator synths. You will find similar algorithms in the algorithm charts for the DX7 and FM-X engines, there are a whopping 32 and 88 of them to choose from. We're going to start from the simpler chart for the DX11 family, which has an abbreviated set of 8 algorithms. The additional algorithms for the 6-OP and 8-OP synths are just expansions on these. If you are using a different synth like the Reface-DX, find a similar algorithm for your synth and use that one instead.

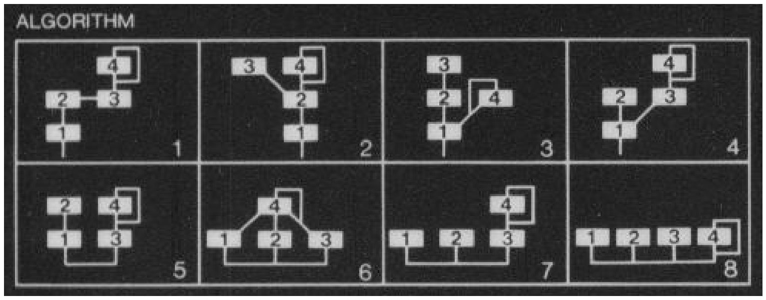


Figure 3. Yamaha 4-Operator DX Synthesizer Algorithms

The algorithms represent pictorially how the Operators are arranged in the synthesizer. In algorithm #1 above, Operator 1 is the Carrier. The carrier is the Operator you hear, the one that plays the note you press on the keyboard. It has a stack of three Modulators on top of it, Operators 2, 3, and 4. Modulators alter the carrier creating additional harmonics in the sound. Operator number four has a line going around it which represents a feedback loop. The feedback loop creates additional harmonics for just Operator number four, essentially creating a set of harmonics that mimic a triangle wave. It brightens the sound produced by the stack, increasing the number of harmonics the algorithm produces. With feedback turned off, Operator 4 produces a simple sine wave like all the others. Feedback is useful in modeling Brass instruments.

In algorithm #2 Operators 3 and 4 are both modulating Operator 2, which modulates Operator 1 as the carrier.

In algorithm #8 we have four carriers and no modulators at all.

We're going to start with the easiest algorithm to understand, number 8. Algorithm #8 simply adds the output of four Operators to form the sound you hear. We can set up a Clarinet by setting our Operators up to mimic the four loudest harmonics in the sound. (Use Algorithm #32 on the DX7 and #1 on the Montage.)

Operator	**Freq**	**Vol**	
Op1	1.00	99	Four individual sine waves
OP2	3.00	80	
Op3	5.00	78	
Op4	7.00	76	

Now go in and adjust the volumes of the four Operators a bit by ear, and it should sound close to a Clarinet. Adjusting the volume of each individual harmonic is a bit difficult to do by ear, which is the downside of using algorithm #8. It's a good thing we can estimate the volumes off of the spectrum chart. This method is called Additive Synthesis, which doesn't really use any of the advantages of FM. It would take eight sine wave Operators used like this to match what FM can do with just two or three. Algorithm #8 sounds a bit thin, so we'll try another algorithm to improve on it, but first let's set up the envelope generators. It's easier to get the harmonics right when the envelopes are set up so the shape of the sound is closer to the sound of the actual instrument we're working on.

6. ADSR Envelopes - Clarinet

The next step is to set up the way the synthesizer plays each note. ADSR stands for Attack-Decay-Sustain-Release. The Attack rate sets how fast the synth gets from silence up to full volume for the note. A fast attack rate of 31 on the DX11 (99 on a DX7) means it hits the note hard. A medium attack rate 15 (55) means it raises the volume of the note slower and eases into the note smoothly. A slow attack rate of 2 (16) means it's going to take several seconds to get the note fully going. I usually start with a medium 15 (55) and bump it up or down depending upon what the instrument is like. A trumpet hits the note harder and faster, a Clarinet takes a bit longer so we'll try 13 (49).

The conversion chart to convert patches from 4-OP to 6-OP synths is included later in this book for reference. From now on I'll abbreviate the numbers giving the DX11 value first, then the DX7 value in parenthesis. DX11 (DX7)

If you have an FS1R or a Montage, the envelope rates are expressed as a time and are inverted to the DX7.
Use 99 – DX7Value = MontageValue.

The levels don't need to be converted on the Montage, they are exactly the same as the DX7.

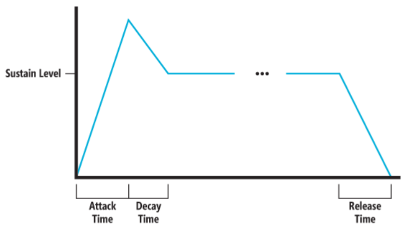


Figure 4. An ADSR Envelope

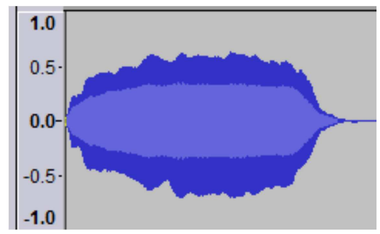


Figure 5. Clarinet Note Envelope

Set the envelopes up on all of the Operators the same to start with. Otherwise, you won't be able to tell what you've got, the fastest, hardest one will overpower the others. The envelopes on the modulators need to match the carriers for now. For most instruments the harmonics turn on as the note begins, matching the increase in volume as the note begins to sound. Setting the Modulator envelopes to match the volume envelope on the Carrier does this quite nicely. You can make adjustments to the modulator envelopes later, after we get the shape of the notes closer to the actual instrument.

In reality, a Clarinet has a more rounded envelope (seen in the Figure 5 to the right above) which we approximate with a rectangular shaped envelope. We set the Attack to medium 13 (49) the Decay rate to 15 (doesn't matter right now) the Sustain rate to 0 (0 and flat) and the Release to 8 (58). The next button over on the DX11 sets the Sustain Level, which we set to the max of 15 (99). This gives us an envelope like the one below. It isn't exactly like the real thing above, but it's close enough. If you have a Montage you will find FM-X patches similar to these in Appendix C with the envelope values laid out in the chart.

Attack	**Decay**	**Sustain**	**Release**	**Sustain Level**	
13(49)	15(55)	0(0)	8(58)	15(99)	DX11(DX7)

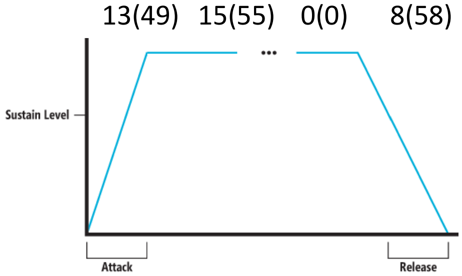


Figure 6. Clarinet ADSR Settings

If we wanted to punch the start of the note like you might on a trumpet, one technique would be to lower the Sustain Level a bit to 12 (46) putting the sharp peak back onto the start of the envelope, and set the Decay rate to 15 (55) causing the note to drop smoothly to the sustain level, as shown in the ADSR Figure 7 below. If you set this up on the modulator only, you get kind of a fast 'wow' sound like a Trumpet on the first part of the note. Modulators, remember, modify the timbre of the sound, not the volume.

Note: Setting the Release rate lower than 8 (58) can be used to simulate reverb on the older DX instruments.

As another enhancement, I like to set the sustain rate to a slow 2 or 3 (16 to 19) on one modulator and lower the sustain level a little, then the instrument note will soften and change timbre slightly as it plays a long note, enhancing the sound. This works great on an expressive instrument like a stringed instrument or a saxophone.

Attack	Decay	Sustain	Release	Sustain Level
13(49)	15(55)	2(16)	8(58)	12(46)

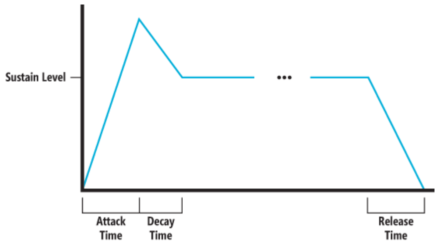
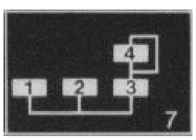


Figure 7. Saxophone ADSR Settings

7. Double-Stack Algorithms - Clarinet

Example 1……………………………………………………………………………………….

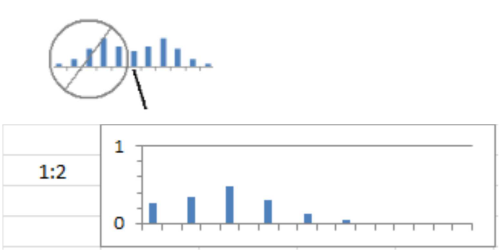
To improve the sound of our Clarinet and go over the types of double-stacked FM algorithms, we'll start out using Algorithm #7 which uses one FM carrier/modulator pair and two single Operators.



(Use #31 on the DX7)

On the Clarinet spectrum plot in Figure 2 the strongest peak is at harmonic #1, the fundamental harmonic. On a DX synthesizer we use ratios instead of frequencies, so we will set Operator 3 to a ratio of 1.00. 1.00 means the Operator frequency is set equal to the fundamental harmonic frequency of the synthesizer, the one you hear when you press down on one of the keys. The spacing between the frequency spikes on this plot is <u>twice</u> the fundamental frequency (i.e. the spacing between peaks 1, 3, 5, and 7 are all 2) so we will set Op4 to a ratio of 2.00. Listen to how it sounds and raise and lower the volume of Op4 until it sounds right to your ear.

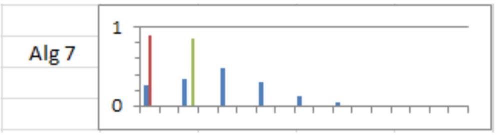
Operator	Freq	Vol	
Op3(5)	1.00	99	Sets the center frequency of the pattern to harmonic 1, all the way to the left
Op4(6)	2.00	69(77)	Sets the spacing of the spikes to 2, the volume sets the pattern shape



The volume of Op4 is set to about I=2.5 here from Figure 1

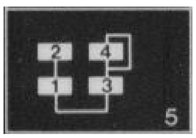
We will now back fill the slots where the lower peaks occur in the middle of the FM plot (the left two on the mini chart for 1:2 just above). We should be somewhere between I=2 and I=3 on the FM graphs above, so we need to boost the amplitude of harmonics 1 and 3. Set Op1 to a frequency of 1.00 and set Op2 to 3.00, then adjust their volumes. The graph for single Operators like Op1 and Op2 look like the graph above where I=0, a single spike. The sound should now be much closer to the mellow Clarinet recording we started from. See the plot below.

Operator	Freq	Vol	
Op1(3)	1.00	80	Two individual sine waves, the red and green spikes below
Op2(4)	3.00	80	



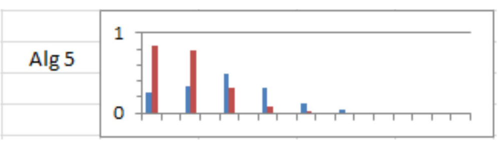
Example 2……………………………………………………………………………………

Now let's switch to Algorithm #5 and fill in the center frequencies a different way, leaving Op3 and Op4 set the way they are. Instead of filling the gap with two individual sine waves, we can fill in the gap using another FM modulator/carrier pair. We can set up Op1 to the fundamental frequency 1.00, then modulate it to create the side bands and pick up harmonic #3 using a lower volume on Op2. Remember the peaks are 2.00 apart, so we lower Op2 to a frequency <u>spacing</u> of 2.00.



(#5)

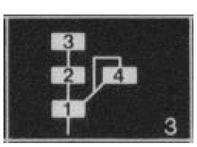
Operator	Freq	Vol	
Op1(3)	1.00	99	The center of the FM pattern, the first red peak on the left
Op2(4)	2.00	60(68)	The spacing of the red spikes to the right of #1, about I=1.5 in fig 1.



Example 3……………………………………………………………………………………

The bottom carrier Operators in both of our FM pairs are at the same frequency, 1.00, and the same volume. We could use one less Operator if we combined our two modulators Op2 and Op4 onto one carrier. Looking at the other algorithms we have to choose from, there are two other algorithms that do this for us, algorithms 3 and 4. We can save algorithm 5 for another time.

We'll pick Algorithm #3 and set the output volume for the top Operator Op3 to 0.0 since we don't need it. The result sounds exactly the same as our previous set up. We now have two modulators, one set to about I=2.5, the other set to about I=1.5, and we're using only one carrier now.



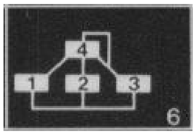
(#8)

Operator	Freq	Vol	
Op1(3)	1.00	99	Carrier

Op2(5)	2.00	69(77)	Modulator #1
Op3(6)	1.00	0	Not Used
Op4(4)	2.00	60(68)	Modulator #2 (lower volume/Index than Op2)

Example 4………………………………………………………………………………………….

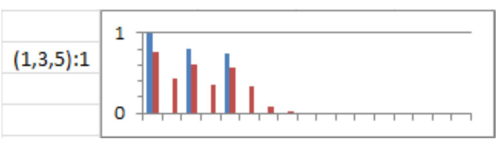
In addition to the three double-stacked algorithms we've tried, there is also another variant we can use. Algorithm #6 provides an entirely different way to combine our modulators. It uses three carriers and modulates all three using the same modulator.



(#22)

Looking at the Clarinet spectrum chart again, all of the even and odd harmonics are separated by the same spacing of 1.00. In a lot of cases all of the double-stacked algorithms tend to use modulators set to 1.00, hence algorithm 6. Using Algorithm #6, we could construct the Clarinet patch by targeting harmonics 1, 3, and 5 and pick up the even harmonics 2, 4, 6, and 8 in between by modulating the three carriers, which is what we've done below. You don't have the Option of different modulator volumes, but that isn't always needed.

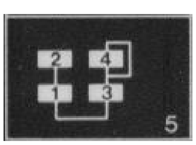
Operator	**Freq**	**Vol**	
Op1(3)	1.00	99	Carrier #1 in blue, far left
OP2(4)	3.00	80	Carrier #2 in blue, middle
Op3(5)	5.00	75	Carrier #3 in blue, right side
Op4(6)	1.00	65(73)	Sets the spacing of all the added harmonics in red



The sound is a bit lacking without the 7th harmonic, but I think you get the idea. We can fix it a little by using the Feedback feature of Op4, which we haven't used yet. Raise the Feedback up to 5 (out of a maximum of 7) to brighten up the sound a little. Like we mentioned before, feeding back a sine wave Operator back onto itself forms a triangle wave which is brighter in timbre than the original sine wave. It adds more harmonics when used as a modulator this way.

Example 5………………………………………………………………………………………….

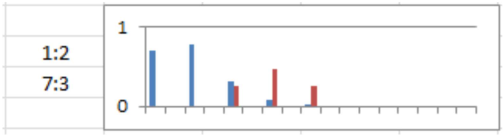
Let's do one more double-stack algorithm example, let's go back to Algorithm #5 and try something a bit more creative. We can set up one FM pair to target harmonics 1, 3, and 5, and set up the second FM pair to target harmonics 5, 7, and 9, with the second carrier centered on harmonic #7. (Initially I tried 4,7,10 off the plot but it didn't sound right.)



(#5)

Operator	**Freq**	**Vol**	
Op1(3)	1.00	99	Carrier #1
Op2(4)	2.00	69(77)	Creates harmonics 1, 3, 5 in Blue
Op3(5)	7.00	60	Carrier #2

Op4(6)   2.00   75(83)   Creates harmonics 5, 7, 9 in Red



This combination sounds really close to the original Clarinet recording, it's probably the best one in the bunch. You can also start to see that you can get the same type of sound out of several different algorithms. The algorithm you use has little to do with the sound that results, it's how you use the algorithm that matters. There are actually only a small handful of algorithms that get used all the time, the rest are mostly ignored. The number of algorithms doesn't really matter all that much, most of the algorithms are there simply because the Yamaha design engineers wanted every possible combination covered.

8. Spacing Harmonics Over 2 Apart

Up until now we've limited the harmonic spacing to just 1 or 2 when modulating. It gets more complicated when you go to 3 or higher… FM doesn't behave quite like you expect it to, half the time it adds in a few extra harmonics in a repeating pattern, and half the time it doesn't. This is somewhat confusing, so skip over this section if you'd rather.

If you use a large frequency carrier and a smaller modulator (like a 5:1 combination) FM works like you expect it to. But, when you have a small carrier and a larger modulator (like a 1:5) it adds in a few extra harmonics. The extras are predictable in location for combinations with a 1 in them (the most common combination you will use). The extra harmonics get added in between each of the ones you expect to see, they lag the expected ones by two. So, if you expect the 6th harmonic, you also get the 4th (see below). In the 1:5 plot below we expect to see the harmonics 1, 6, and 11, but we also get the 4th and 9th harmonics as extras. The extras are the result of harmonics in the pattern reflecting across the 0.0 axis, but it's a bit difficult explain why that occurs.

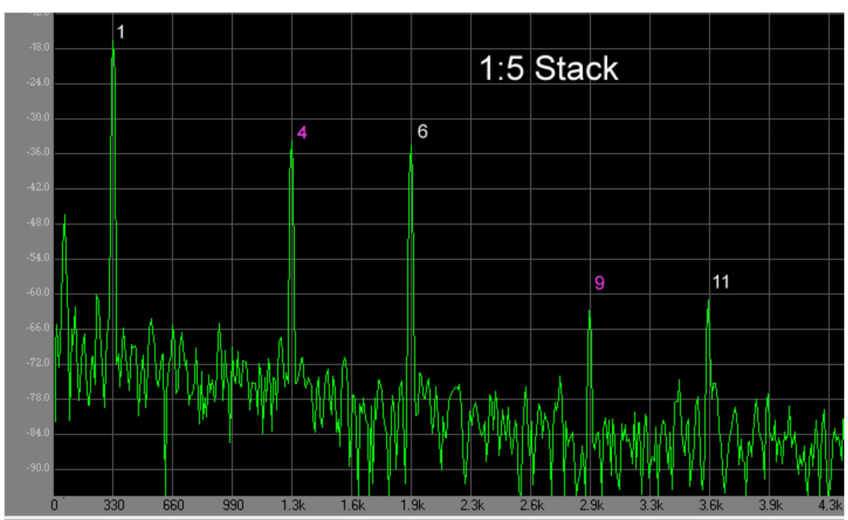


Figure 8. Extra Harmonics 4 & 9

Using other ratios of harmonics with numbers greater than 1 as the lowest one, results in unpredictable extra harmonics. There are charts out there that show the locations of the extra harmonics, but every combination seems to be different. Just be aware of this. We'll try to ignore the extra harmonics for now in our discussions, just don't be surprised if you see a few more harmonics than you bargained for on a live spectrum analyzer plot.

9. Triple-Stack Algorithms - Saxophone

Now that you've seen some of the basic concepts, let's try something a bit more complicated and try out a few algorithms with a stack of three Operators. I picked the Saxophone because every single recording I loaded of a Saxophone was different. The Saxophone is very expressive, with a lot of variations between individual instruments. I've made some number notations on the plot below to show the harmonics used to construct the patch. This is just one way to approach a Saxophone. You can probably see a better way to model this from the plot below, but I picked this method for demonstration purposes even though it doesn't sound all that great. We'll switch to a better method in a minute.

Example 1……………………………………………………………………………………….

Looking at the frequency spectrum chart, there are a series of peaks in the harmonics that are all about three apart (ignore the blue circles for now) peaks 2, 5, 8, and 11. We're going to model these and the harmonics in between using a three-stack algorithm, Algorithm #3. First, we pick a base harmonic (peak #2) to center our pattern on, setting Op1 to 2.00. The spacing between the green numbered harmonic peaks is 3, so we set Op2 to 3.00. Now we're going to fan the harmonics out around each of the peaks we've created using our third Operator Op3. We'll set the spacing of the fan to 1.00 in Op3 and set the volume on Op3 so it creates just one to two smaller peaks on either side of the center one. You can see this much more clearly in the mini plots below.

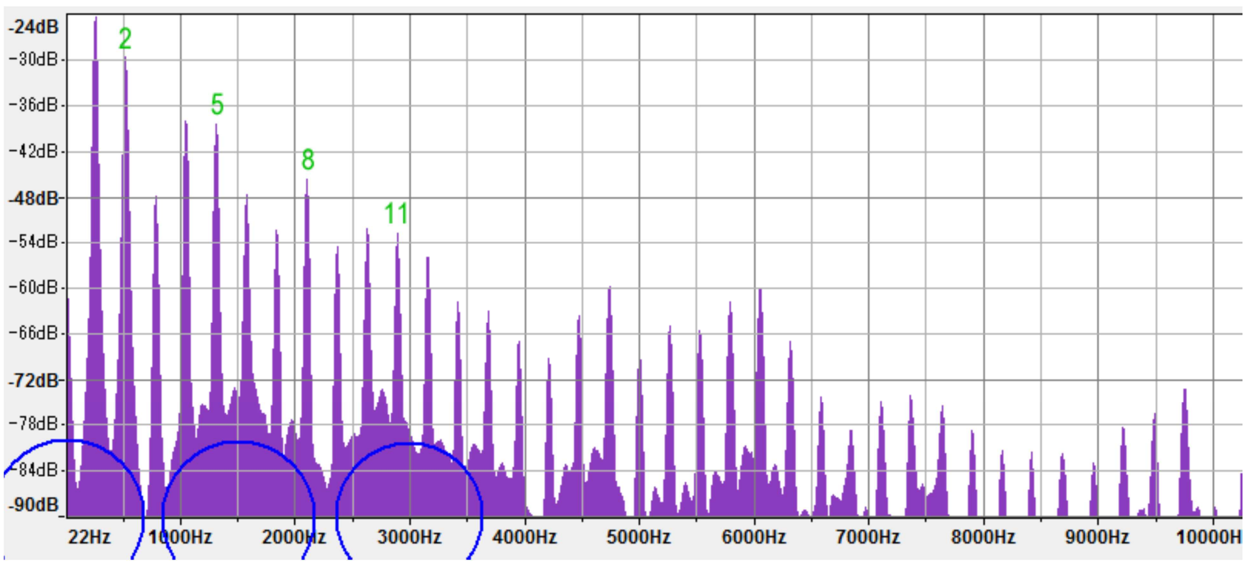
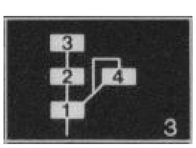


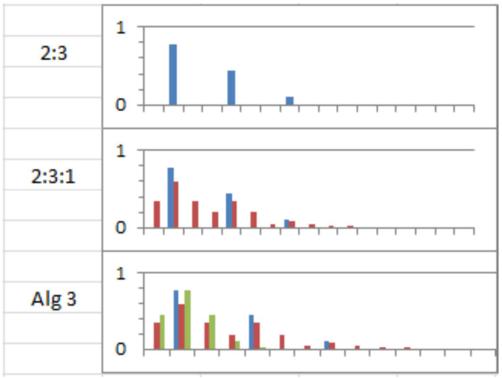
Figure 9. The Saxophone Harmonic Spectrum

To help pick up the fundamental harmonic #1 we'll also use Op4. Using it to modulate Op1, it will add more volume to harmonics 1 and 2 if we use a low volume for Op4. We'll go over this one step at a time now.



(#8)

Operator	Freq	Vol	
Op1(3)	2.00	99	Main Carrier, the left-most blue peak below
OP2(5)	3.00	70(78)	Creates harmonics 2, 5, and 8 in blue
Op3(6)	1.00	65(73)	Adds the harmonics in between in red
Op4(4)	1.00	50(58)	Helps pick up a little more of the fundamental, shown in green



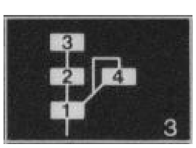
Set up the frequencies in the table above, turn off Op3 and Op4, then listen to the result while varying the volume of Op2. These are the harmonics shown above in blue. Now turn on Op3 and adjust it's volume, it adds in all the harmonics in between the others (1, 3,4, 6,7, 9, 10, 12,13) all the red ones. Adjust the volumes of Op2 and Op3 until you're satisfied, then go on to Op4. When you get done it should sound similar to one type of Saxophone. Not close enough, though. We need try something else. In hind sight, we should almost always choose 1.00 for the frequency of the Carrier, Op1, it just sounds better overall and is easier to fix if it doesn't.

Example 2…………………………………………………………………………………………

Ok. So let's try a totally different method of using the same 3 stack algorithm, but this time we'll use a big number to modulate Op2 with Op3 instead of a small one. Using a large number for Op3 will take the base pattern we create with the first two Operators, copying the whole thing over to the right. If you look at the saxophone plot again you will notice at the bottom of the chart (marked with blue circles) are rounded clusters of harmonics centered around 1, 6 and 11. We're going to model these clusters. For future reference, these clusters are formants which the FS1R can target.

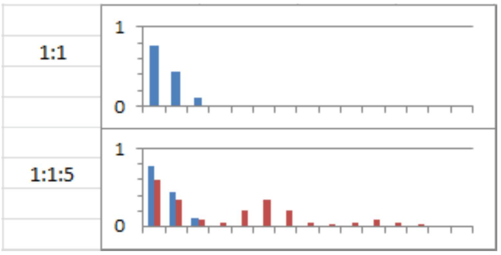
First we create a fan of harmonics around the fundamental #1, then we're going to copy it over to the location of harmonic 6 and also to 11 (both are spaced by 5's to the right). First, set Op1 to 1.00 and Op2 to 1.00 creating the basic harmonic pattern around #1, shown in blue below. Then set Op3 to 5.00 and listen to what it gives you, while adjusting the volume.

It sounds good, but we can fill in some of the lower harmonics using Op4. For some reason it sounded better to me when Op4 was set to 2.00 than 1.00, it needed more of harmonic 3 than it did of harmonic 2. Remember, the modulator sets the spacing in between the added harmonics.

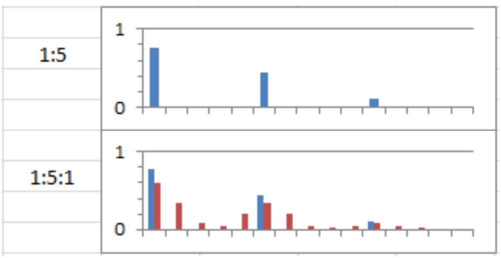


(#8)

Operator	**Freq**	**Vol**	
Op1(3)	1.00	99	Main Carrier, the left-most blue spike
OP2(5)	1.00	65(73)	Creates harmonics 1,2,3 in blue
Op3(6)	5.00	60(68)	Copies the base pattern 5 and 10 units to the right in red
Op4(4)	2.00	72(80)	Helps pick up more of harmonics 1 & 3 (not shown)



Some of you may be wondering if the order of Operations makes a difference, does a 1:1:5 sound the same as a 1:5:1? Can you swap settings for Op2 and Op3 and get the same sound? Actually you can, and after listening to both, 1:5:1 sounds a bit brighter and more resonant than 1:1:5 because of the different order of Operations. Remember the extra harmonics we mentioned in section 8? However, with some adjustments to the volumes they will sound about the same. Compare the plots above and below.



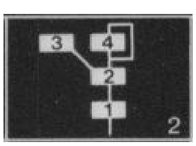
Same end result, different order of Operations

Looking at the results on the spectrum analyzer reveals an interesting story. Below an Index of 1.0 everything behaves fairly close to our model. However, if you crank the modulation volumes way up to an Index of 3.0 or greater, the order of Operations makes a huge difference in the results. The 1:1:5 construct is filled in near the center and creates a denser, closer-in fan of harmonics. The 1:5:1 construct creates a wider fan, with an empty spot in near the center by harmonic 1 and less dense patterns outwards. In FM-X you can bypass Figure 1 altogether if you want to, using the new waveforms.

There are many other ways to set up a Saxophone patch we could try. For example, we could go back to double-stack Algorithm #5 and set up two pairs of FM carrier/modulators. Use one FM pair to model harmonics 1, 4, and 7, and the second FM pair to model harmonics 2, 5, and 8. I'll leave you to try that one out on your own.

Example 3……………………………………………………………………………………….

What about Algorithm #2? How do you suppose having two modulators on Op2 could be used in this algorithm?



(#14)

Looking at the layout of algorithm 2, the additional Operator Op3 is added on top in parallel to Op4. The intent of this layout is to give you the ability to widen the frequency fan out. If you use a 1:5:1 like we did before, you can now modulate the top harmonic fans using two different modulator <u>volumes</u>. Set Op3 and Op4 both to 1.00, then set Op3 to an Index of about 3.0 creating a wide fan, and Op4 to a lower Index of 0.8 creating a narrow fan emphasizing the peaks in nearer the center. This is essentially what we did up on the Clarinet the first time we used algorithm 5, remember?

Another Option would be to use different spacings on the two modulators, maybe set one to 1.00 and the other to 3.00. The algorithms are stacks of building blocks to construct a set of harmonics with. Be creative and you'll find you can model just about any instrument using these basic building blocks.

Another 3-Stack Trick

I recently ran across another trick you can use on a stack of three Operators. If the top-most Operator uses a fixed frequency rather than a ratio (such as 7.98 hz at a loud volume) you get an explosion of high frequency harmonics. This is used in the 'UprightPiano' patch on the FS1R. It adds in a burst of harmonics to model the plural nature of slightly out-of-tune piano strings, which was then filtered back down into submission so it would sound like a piano.

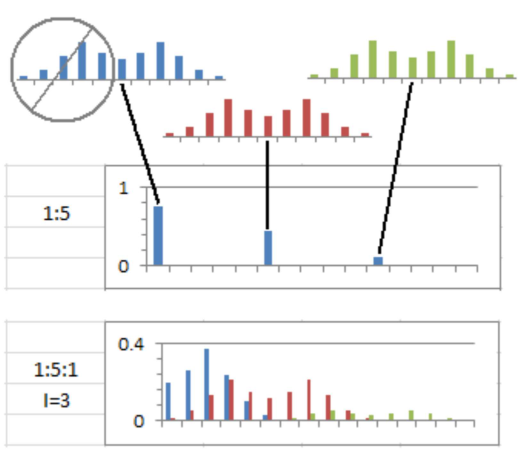
I've also seen fixed frequency Operators used to model the mechanical thunk in a piano when the hammer hits the strings. Yamaha used a fixed Carrier frequency of 1.0 hz modulated by a fixed frequency modulator at 94.06 hz. It sounded exactly like the action of a mechanical piano in the background behind the main piano harmonics.

10. Fan Overlap -Interference Patterns

Stacking three or more Operators introduces a new difficulty into the mix. If the added harmonic fans are wide enough, the fans can overlap each other. In the zones where they overlap you get an interference pattern, just like the way ripples in a pond intermingle. The pattern is somewhat random, you get a different pattern in the overlap every time you play a new note. This changes the timbre of the notes, making a slightly different sound for each note. If this is something you want, then great! You've achieved your goal. If it's not, you need to correct this by reducing the volume of the correct moderator. This type of rolling dissonance is used in creating Piano patches by Yamaha on the FS1R.

Let's take a look at an example for clarity, a 1:5:1 pattern of harmonics. We'll start with a 1:5 pair, pictured below, which creates three base harmonics, 1, 6, and 11 in blue. In our 1:5 pattern you can see below that we have four empty slots in between each harmonic. If our added fan only had two peaks on each side, the added fan would fit without interfering with the one next to it. Instead, we're going to add a wide fan using a third Operator at an Index of 3.0. Can you see how the added fans are too wide to fit without overlapping each other? (See below.)

The areas where the fans overlap will result in interference, causing a random interference pattern to form. The interference pattern causes the peaks in these areas to form at random heights. Random changes will occur in the harmonics each time you press a new key to play a new note, sounding slightly different each time. You can correct this by reducing the Index value to 1.0 or below on the third Operator by lowering the volume, narrowing the fans added on by the third Operator in the stack.



What if you were working on a 1:1:5 stack of Operators, which Operator do you think needs the Index value adjusted this time? You adjust the middle 1.0 Operator this time. As a rule, you adjust the modulator with the smallest frequency ratio, the one that adds on the closest-spaced fan of harmonics to the stack.

11. ADSR Envelopes - Saxophone

We're going to start with the same basic envelope shape as we did before, the Clarinet and Saxophone are both wind instruments after all. After messing around with the envelopes I finally settled on a slightly different approach for shaping the Saxophone notes.

I used Op2 (a modulator) to shape the timbre of the notes when they are held for longer periods of time, using a Decay of 4 and Sustain Level of 12. See below. This alters the timbre of the sound slowly when a note is held on to. The slow rate of 4 lowers the level slowly from 15 down to the lower sustain level of 12. I also used sustain rates of 2 on all of the Operators to reduce the volume slightly over time.

Operator	Attack	Decay	Sustain	Release	Sustain Level
Op1(3)	13(49)	31(99)	2(16)	8(34)	15(55)
Op2(5)	13	**4(22)**	2	8	**12(46)**
Op3(6)	13	31	2	8	15
Op4(4)	13	31	2	8	15

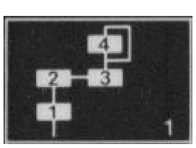
I've found it's very useful to have two different envelopes per voice, one for the Carriers and another for the Modulators. One set of envelopes controls the shape of the notes, the other controls the timbre. Take a look at the instrument setup charts at the end of this book and you'll see a large number of examples.

It can also be effective to use a different envelope for each set of modulators for multi-stack algorithms. On multi-stack algorithms, it is commonly set up with one stack modeling the instrument attack, and the second stack modeling the sustain portion of the sound. You can use one set of envelopes for the note attack, and a second set for the sustain level when modeling an instrument.

I've also seen an Operator stack split used on an FS1R to model different instrument registers. One stack was used to model a piano over almost all of the range, and the second stack was used to enhance just the bottom two octaves. Use your imagination. I'm sure you'll come up with some very interesting patches.

12. Four-Stack Algorithms

Algorithm #1 on our chart has all four Operators stacked, modulators modulating modulators modulating modulators. A number of the DX7 algorithms use this same construct. What goal would you have in stacking four (or more) Operators, possibly a wider set of harmonics? A lot of people seem to like using the 1:1:1:1 stack for some reason in the patch database I have for the DX11. I played around with this in a spreadsheet and figured out that a stack of 1:1:1:1 doesn't get you anywhere fast. After the first 1:1 you have to mix in some larger numbers if you want to expand the harmonics by more than one or two. A better way to create a wide fan of harmonics would be to use a stack like 1:1:3:6 or 1:1:4:10.

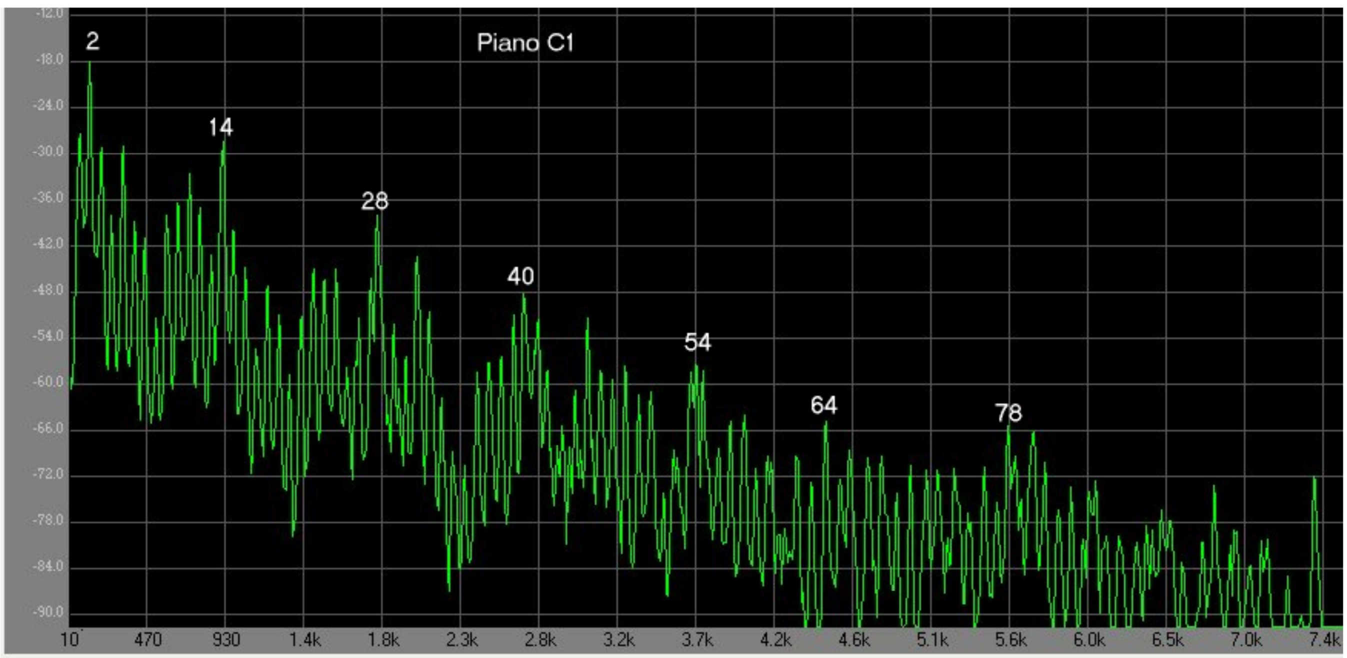


(#1)

In evaluating all of the four-stack instrument patches I could find, I did find people using stacks of four Operators. A lot of them could have been accomplished with a simpler three stack if they adjusted the modulator volumes a bit. A few of the others mixed in different wave forms into the modulators, like w1:w2:w4:w1. Using different wave forms changed the sounds produced. You just have to use a little more creative thinking when using additional stacked modulators.

Stacking four Operators usually gets too complicated to explain or follow easily. The fourth modulator expands the number of harmonics exponentially, creating a real mess to try and use. For myself, I'll be sticking to simpler two and three stack algorithms most of the time. Dr. Chowning mentioned in one of his interviews that using too many stacked Operators would eventually get you into trouble since the increased complexity of the harmonic structure would start to sound more like high-pitched noise than anything useful. I've found that having several sets of two and three stack Operators are more useful and can model just about anything, especially with additional waveforms like on the DX11, FS1R, and FM-X on the Montage.

There are a number of four, five and six stack algorithms on the FS1R and in FM-X. I found one factory four-stack algorithm used on the FS1R where the top-most modulator was a formant, used to introduce a small amount of fuzzy dissonance into a piano patch to model the out-of-tune nature of piano strings. Growl in a Saxophone also looks like this on the spectrum analyzer. I have also found a great use though, for multiple stacked operators for complex harmonics in the Piano and Saxophone. These instruments have ripples on ripples on ripples in the harmonics, lots of strings equals lots of sub-resonances (see below).



Looking at the Piano spectrum above you can see there are dominant harmonic groups about 14 harmonics apart, with more ripples in between. These can be modeled using a complex stack like a 1:14:5:1. Note that this combination will leave a blank spot between the 1st and 14th harmonic when the Modulator volumes are turned up high, you have to fill in the missing harmonics with a second stack of Operators. The FS1R with 16 Operators per voice and multiple waveforms has the advantage here, while the Montage at 8 Operators and multiple waveforms is a close second. You can also achieve this second stack by layering two patches into a performance, playing them simultaneously. Some 6-Operator VST's have this feature, giving you 12, 18, or even 24 Operators to work with.

Another detail to pay attention to on the Piano is that below middle C (C3) the second harmonic is dominant (see above). Above C3 the first harmonic is dominant. On the FS1R the factory Piano patch layers two patches into a performance to achieve a high-quality Piano. One patch adds in the deeper resonance required from C0 up to C3, and the second patch is active from C0 all the way up to C6. It's the best FM acoustic Piano patch I've found.

13. Complex Waveforms – DX11

On the DX11 and TX81Z you can increase the capability of your tools by adding in a set of eight different waveforms. The additional waveforms are essentially a way to more efficiently use the four Operators you have to work with. The waveforms duplicate specific combinations of FM pairs, which can be used to replace similar combinations in your constructs. Most of them duplicate 1:1 pairs of Operators. The different waveforms available are tabulated below, and are actually arranged in order of increasing brightness. I was able to create a much more convincing 4-OP Bassoon patch using the 1:2 W2 waveform, look at the examples in spreadsheet at the end of this book. You'll see several patches that use these different waveforms for the DX11. The DX7 doesn't give you this Option.

Wave	**Description**	**Carr:Mod**	**Amplitude**	
W1	Sine Wave			
W2	Odd Partials 1,3,5,7	1:2	I=0.5	V=12
W3	Even Partials 1,2,4,6	1:1	I=0.5	V=12
W4	Partials 1,2,3	1:1	I=1.0	V=32
W5	Partials 1,2,3,5	1:1	I=1.5	V=65
W6	Partials 1,2,3,5	1:1	I=1.7	V=75
W7	Partials 1,3,4,5	1:1	I=3.7	V=85
W8	Partials 1,3,4,5	1:1	I=4.0	V=87

The V2 (in Japan) became the DX11

Complex Waveforms - SY77, TG77, and SY99

These synths also have a set of more complex waveforms, a set of 16 which are very similar in nature to the ones on the DX11. The waveforms duplicate specific combinations of FM pairs, 8 that are duplicates of the DX11 waveforms and 8 new ones with similar characteristics. The graphs in the manual are pretty hard to read, and since I don't have access to these synths I'll leave that up to you guys to dissect.

14. Troubleshooting Patches

Once you have a patch initially set up, it's time to make adjustments. There are a few basic methods of adjusting a patch to match the recording you are working from. Below are a few of the most common adjustments.

A. The patch sounds terrible, it's not even close to the recording even after adjusting the modulators: You have something wrong with the frequencies you've chosen for the patch. Go back to the original spectrum chart and check the frequencies you've entered for each of the Operators. You can check it on a spectrum analyzer too.
B. The patch sounds like it's higher in frequency than the recording: Adjust the volumes of the carrier Operators to make the lower frequency carriers louder. Or vice versa.
C. The patch sounds like it is thinner in frequency content than the recording: Adjust the volume of the different modulator Operators to increase their volume, adding more harmonics into the sound.
D. The patch sounds too smooth and mellow compared to the recording: Add some variation into the patch by detuning the Operators. Detune the Operators by +- 1 to 3 units randomly until you get sufficient variation in the sound. The Yamaha Piano patches are detuned this way.
E. The vowel sound of the patch is wrong, it has an 'mmm' sound rather than the 'ah' sound you want: This is caused by formants in the actual instrument and can be difficult to fix if you don't have an FS1R. Part of the problem may be that you targeted a low note in the recording when a more middle-register note would work better. You might also have a problem with the modulator volumes. Sometimes you can change the vowel sound of the patch by adjusting the volumes of the modulators. With the FS1R you can change the formant filter frequencies using Figure 25 and select the vowel sound you want.

After doing this dozens times I've noticed a couple of things in adjusting patches. First, you really need to get the amplitudes of the first 3 to 5 harmonics correct relative to each other for the patch to sound right. These harmonics shape most of the tone of the sound and have to be correctly balanced in order for the patch to sound right. For the higher harmonics, this doesn't seem to be as critical. The second thing I've noticed is that matching the spectrum chart doesn't quite give you the sound you want, it always seems to be duller than the recording. This may be the result of the spectrum analyzer not working quite right. I always end up boosting the amplitudes of the higher frequency components to fix it. I recommend doing this right from the start, if you can hear it.

15. Adjusting Note Registers

We also have to compensate for changes in an instrument's timbre as it plays higher notes, which was one of the main points John Chowning made in his original paper back in 1973. The DX FM synths allow you to adjust this with the Keyboard Scaling settings. I've found that setting the Level Scaling for modulators helps imitate a real instrument closer. On the Saxophone I've set the Operators on the DX11 to the following, 0:39:0:60 for algorithm 5 (two sets of carrier/modulator pairs). Using these settings, the two modulators lose intensity as you play up the keyboard lessening the harmonics generated. In most real instruments higher notes have fewer harmonics. It should work the same way on the DX7.

Brass instruments are the Opposite, they tend to get brassier as the notes get higher and also as the volume increases. You can model both of these characteristics on your keyboard. Leave the keyboard scaling set to 0:0:0:0 on the DX11 to keep the notes brassy as you play up the keyboard. To adjust the volume effect, use the Key Velocity Sensitivity setting on your modulators. Raise the setting for the modulators up to a 3 or 6 on a DX11 on your brass instrument patches to make them brassier as you hit the keys harder.

Another trick I like is to set up the LFO for vibrato with a long delay, so the vibrato doesn't start until the next measure of the piece. That gives you an instrument with no vibrato until you hold the note for an entire measure. I like to set up flute patches this way.

16. Balancing Voices

Once you get all of your patches set up, you need to test them out with a piece of music. You need to listen to your patches side by side with others so you can check their volume levels and balance them. When I set up my first string ensemble piece to test my string patches, the Bass Viol was way too loud. It overpowered all of the other instruments and I had to lower the carrier volumes so it would blend well with the other patches when set to the same volume in a midi file. I also found my Flute patch disappeared in a wind ensemble. I had to increase the Flute's volume and shrillness by increasing the volume of the modulators so it could be heard along with all the other wind instruments. I've noticed using the factory patches on several DX keyboards that very few of the patches are balanced with one another. It's not as important here because these keyboards are mainly used as solo instruments. The Yamaha MU-series tone generators are much better at balancing voices, since one of their original design goals was to play back multi-track midi files. I've got an MU-2000 to play with.

Another anomaly that occurred had to do with the Bass Viol patch on the DX11. I started with a slower Attack on the notes (an 11 or 12) to allow the instrument to resonate and come up to speed as the note progressed, just like the real instrument did in the recordings. However, when played in an ensemble the slow attack sounded like the Bass Viol was coming in an 1/8 note late on all of the faster notes, which was very disconcerting. It reminded me of practicing in the university orchestra when the conductor would stop and criticize the Basses for coming in late. Playing by themselves it sounded fine. Playing with the rest of the orchestra they sounded like they we coming in late. In hindsight, it was the note attack. They weren't bowing hard enough on the fast notes, the same problem I was having with the Bass Viol patch. I shortened the attack timing in the ADSR envelope to fix the problem.

17. Performance Voices

After attempting to match a number of real instruments on a 4-operator synth you may find you still can't get it quite right. Something is missing in the sound you just can't seem to compensate for. We have one more trick up our sleeve that might help. (This works on the 6-OP and 8-OP synths as well.) On the DX11 and the other multi-timbral synths you can set up a Performance where the synth will play more than one patch at a time. You can double the number of Operators you have available, turning your 4-OP synthesizer into an 8 Operator synth, by splitting a voice up between two different patches and playing them together using a custom Performance. You set the midi input for both Performance voices to the same channel so they both play from the keyboard at the same time when you play the custom Performance. The DX11 can actually play up to 8 different single note patches at the same time.

Editing the two halves of the voice can be tricky. It's a bit difficult to anticipate what you're going to get since you have to edit each half separately, but you can really increase the capability of your instrument this way. On the DX11 layering two voices reduces the number of notes you can play down to 4 at a time, but it allows you to emulate any of the DX7 patches which all use 6 Operators to make up a voice. Basically this increases your synthesizer up to an 8 Operator FM synth. For a good solo patch you really only need two note capability, so you could even increase this up to 16 Operators if you want to. I don't recommend going higher than this, though. The DX11 has a rather chopped note transition when you play a second note before releasing the first one if you play all eight voices at the same time. Experiment and see what you come up with.

It's a bit more complicated to set up, but if there is a DX7 patch you just can't live without on your 4-Operator synth, this is one method of getting at it without buying another synthesizer.

18. 4-OP to 6-OP Patch Conversion Chart

You can use the following chart to convert backwards, from a 6-OP synth to a 4-OP. It's a bit difficult, but it can be done. You just have to test each portion of the 6-OP patch to determine which part is the most important. I've found by previewing DX7 patches on the FS1R that many of the patches have one column of 3 or 4 Operators that are doing most of the work. The other Operators are redundant, turned off, or don't contribute much the sound. I found this conversion chart online and added a few more details into it. If you are working on an FS1R or a Montage you can import DX7 patches directly, there is no need to convert them manually.

Converting 4-OP Patches to 6-OP Synths:

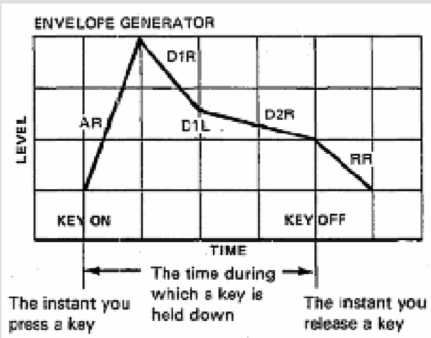
```
All comparisons were done by ear, between a DX100 and a DX7S.

ALGORITHM: Use the obvious mappings. Two cautions: pay attention when copying
parameters, since you will be mapping Operators 1/2/3/4 to 3/4/5/6.
Also, watch out where the feedback loop is.

DX100   DX7     Operators____
1       1       1234 → 3456
2       14      1234 → 3456
3       8       1234 →    3564  (This one is different…)
4       7       1234 → 3456
5       5       1234 → 3456
6       22      1234 → 3456
7       31      1234 → 3456
8       32      1234 → 3456

FEEDBACK LEVEL: Same, although the 4-OP has slightly more "bite" at the same
level. That is, at FBL=7 the 4-OP has more "bite" than the 6-OP does at 7, but the 4-OP
at FBL=6 has less of a "bite" than the 6-OP at 7. You may want to try adding 1 to the
original FBL, and see how it sounds.

FREQ RATIO: Same. All of the 4-OP ratios can be achieved by a 6-OP synth. Of
course, the 6-OP synth should be set for Frequency(Ratio).

DETUNE: Same. Well, almost. 1,2,3 on the 4-OP really come out as 1.5, 2.5, 3.5 on the 6-
OP, but it doesn't have those so you'll have to make do with 1,2,3.  Incidentally, I
don't know about the old DX7, but the DX7S is never in tune on all keys with the DX100.
The variations are all less than 0.2 Hz, but quite audible when you beat sine waves
together. Wasn't this what made the Minimoog sound so good? :-)

OSC. WAVE: If you don't have a DX11 or TX81Z and this isn't set to 1, sorry. I'd
guess that as long as no more than two Operators use a Wave other than 1, you
could come pretty close by using additional modulators.

ENVELOPE GENERATOR: The 4-OP synth gives you control over four rates (range 0
to 31 on the first three, 0 to 15 on the last) and one level (range 0 to 15).
The 6-OP gives you control over four rates and four levels, all range 0 to 99.

4-OP                     6-OP
  AR   corresponds to  R1
 D1R   corresponds to  R2
 D2R   corresponds to  R3, which defaults to 0 on 4-OP and 10 on 6-OP
  RR   corresponds to  R4
                        L1  must be set to  99
 D1L   corresponds to  L2  use the table below
                        L3  is set to L2-R3
```

L4 must be set to 0

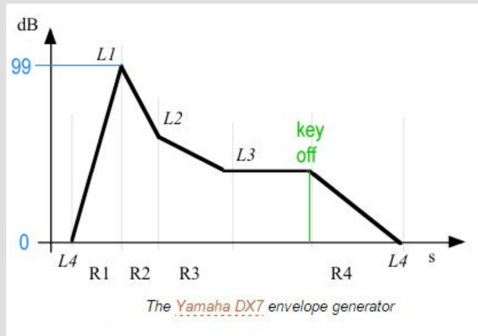
To convert AR, D1R, and D2R to R1, R2, and R3, multiply by 3 and add 10.
To convert RR to R4, multiply by 6 and add 10.
No, this doesn't quite work out as you approach maximum; you'll have to fudge it a bit.
A value of zero should be set to zero, since this means an infinite time.

The relationship between D1L an L2 is audibly non-linear. Refer to the table below:

D1L	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
L2	99	96	92	87	82	78	74	70	66	63	60	56	53	50	47	0



DX100



DX7

OUTPUT LEVEL: Range 0 to 99 on both, but slightly offset. The 6-OP level is equal to the 4-OP level plus 8. Yes, this means that 4-OP levels greater than 91 cannot be reproduced by the 6-OP synth. On a carrier, this is easy to deal with; on a modulator, it is not. Sorry about that. Fortunately, most of the patches I wanted to convert used a modulator Output Level less than 90.
Of course, the Output Level of the unused Operators should be set to 0.

KEYBOARD RATE SCALING: The 4-OP has a bit of rate scaling even when KRS=0. And on the DX100 (at least), rate scaling has discontinuities in it, which makes it a tough call to get *exactly* the same. But this is close:

4-OP KRS	0	1	2	3
6-OP KRS	1	2	5	7

KEYBOARD LEVEL SCALING: Start by setting the 6-OP level scaling parameters as follows: Break Point = A1; L-Curve = -LIN; R-Curve = -EXP; L-Depth = 0. Then set the R-Depth to 90% of the 4-OP KLS. 99 becomes 90, 50 becomes 45, etc.
This is almost perfect, but not quite; I'd like to hear from anyone who has refined it. I tried moving the break point and fiddling with the left curve; it didn't sound any better, and was harder to program.

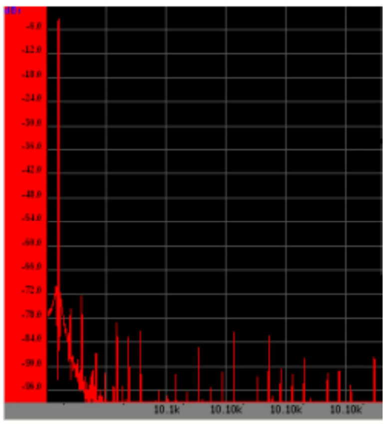
There you have it. Following these rules, I converted a variety of the factory presets from a DX100 to a DX7S; the two were virtually indistinguishable, except that the DX7S sounded much cleaner.

The LFO, AMS, PWS, AMD, PMD, velocity curve, and pitch envelope are left as an exercise for the reader. :-) Seriously, I find most of the factory settings for these pretty wretched, and they are easy to reset by ear.

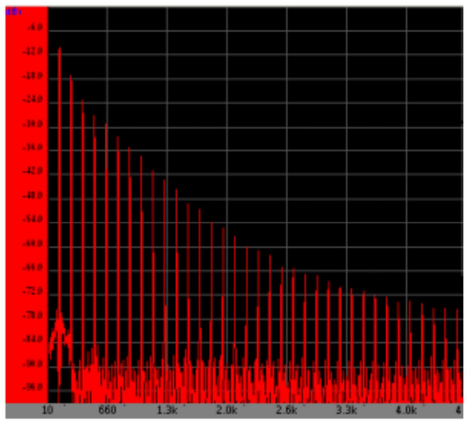
19. Montage FM-X and the FS1R

One of the biggest improvements in the capabilities of FM instruments is the additional of some really broad harmonic waveforms. These synthesizers have a set of waveforms with an almost infinite number of harmonics in them. Basically, you can use <u>one</u> Operator all by itself to model an entire stack of DX7 sine wave Operators.

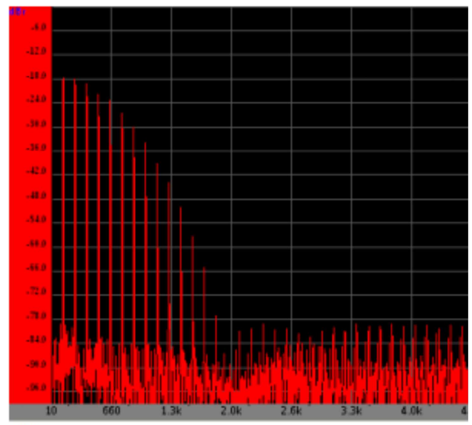
The only place where you can find plots of these waveforms are in an article by Michał Wiernowolski, outlining the technical details of the FS1R. The waveforms aren't even shown on the Yamaha. After our previous discussions, the advantages of these waveforms should be quite apparent (see the plots below). The details of using these waveforms are not explained in any of the Yamaha Montage manuals either.



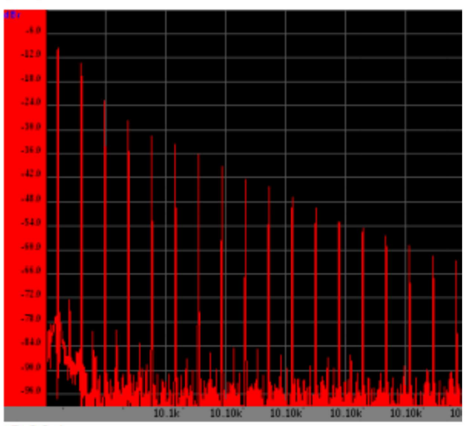
Sine



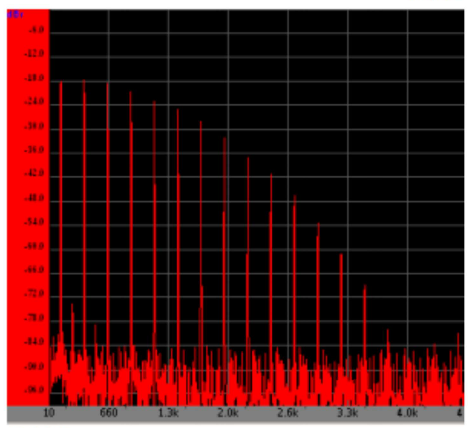
All 1



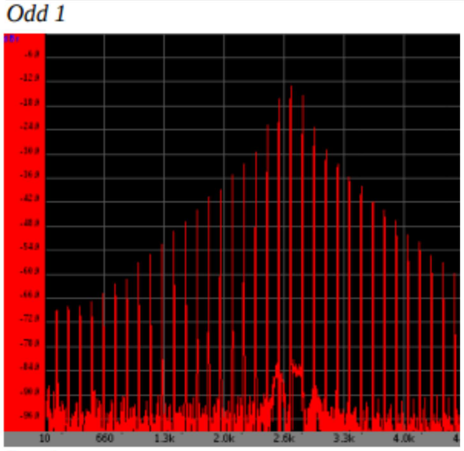
All 2



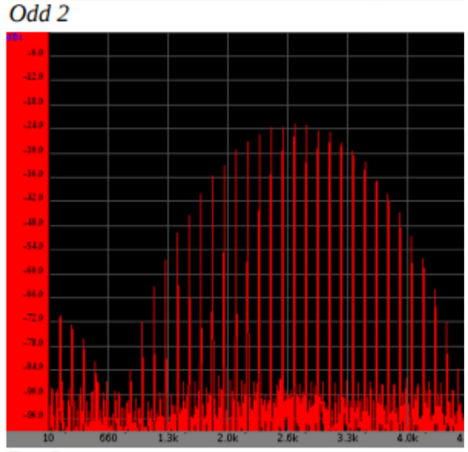
Odd 1



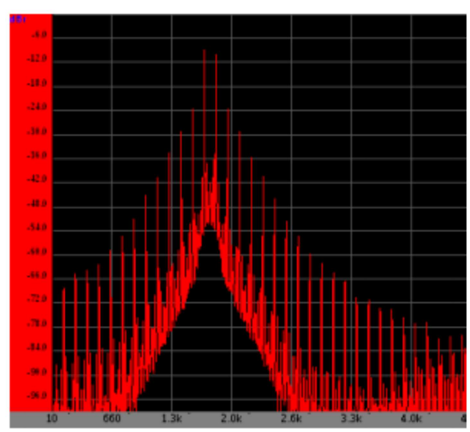
Odd 2



Res 1



Res 2



Formant

Notes on using these waveforms……………………………………………………………………………………….

In these waveforms, the spacing of the harmonics in the ALL1, ALL2, RES1, and RES2 waveforms are determined by the ratio you select. A ratio of 4.0 anchors the sound on the 4th harmonic and spaces the harmonics out by four times the fundamental. It equals a 4:4 FM pair, but with a more controlled set of harmonics.

ALL1 and ALL2 when used as modulators to a sine wave (at a low volume below 60) give you a very high center spike with a much lower skirt of harmonics. The skirt rises as the modulation volume rises. Use these waveforms for modern pads and sweep patches.

In the RES1 and RES2 waveforms the center point of the filter is selected by the Resonance setting which matches the number of the peak it's set to plus one. Set the Resonance to 4 and it moves the filter maximum to the 5th peak in the set. In the plot above, it's set to 21+1=22nd peak. Modulating with these waveforms gets messy, you get a continuously morphing set of harmonics for some reason. These are best used as additive carriers, for reed and string instruments.

In the ODD1 and ODD2 waveforms the harmonics are spaced by twice the fundamental ratio you select. Selecting a ratio of 1.0 anchors the sound on the 1st harmonic and spaces the harmonics out by 2.0. Selecting a ratio of 3.0 anchors the sound on the 3rd harmonic and spaces the harmonics out by six. Used as a modulator at a low volume (just like ALL1 and ALL2) both of these waveforms give you a high center spike with a low fan of harmonics. These give you a Clarinet reed sound.

The width of the harmonic fans are determined by the 'Skirt' setting. A Skirt setting of 0 gives you a sine wave. A Skirt setting of 1 gives you three harmonics, the center fundamental harmonic and one on either side of it. The higher the Skirt setting, the higher the number of harmonics it creates. The harmonics in these waveforms are also uniform, which is an advantage over an FM pair like you see in Figure 1. You get a lot more harmonics with FM-X and much greater control over them.

For the Formant waveform, the harmonics are always anchored on the 1st harmonic and spaced by 1.0. The center maximum of the filter is adjustable to a fixed position determined by the frequency (in hertz) set in the editor. You can make the formant move, but that's a pretty advanced topic. The Formant waveform is for modeling instruments with a resonating body like a Violin or Bassoon. However, this waveform is only available on the FS1R, it's not on the Montage series of synths. We'll address Formants in the next section.

Use the plots above to remind you of the harmonic spectrum each waveform produces. I've found I really haven't used the All1/2 or Odd1/2 waveforms very much in modeling acoustic instruments. The high harmonic content seems to be better suited to making more modern synthesizer tones with harmonic sweeps, pads, and effects rather than reed, brass, or string instruments. You can get some nice harpsichord, organ, and guitar sounds out of them, though. The ones I've found the most useful are the RES1, RES2, and Formant waveforms, which work well in modeling reed and string instruments, and instruments with resonating bodies.

Bear in mind I put these patches together on an FS1R using the software editor, which is why the settings graphics below are what they are.

Example 1……………………………………………………………………………………… English Horn

The waveforms that seem to be the easiest to use for reed instruments are RES1 and RES2. They form either a triangular or an arched set of harmonics which you can center by selecting the resonance harmonic. This's exactly what's needed to match the harmonic structure of lots of instruments quickly. We'll use these two waveforms first to model an English Horn. Select a recording of a note in the <u>middle</u> of the instrument's range to match and capture the harmonic spectrum in Audacity (shown below).

The Fundamental harmonic is pretty strong, so we'll pick it up using Op1 as a Sine wave. The second group of harmonics are in a triangular pattern around the 4th harmonic, so we'll make Op2 a RES1 waveform. You center it on the 4th harmonic by setting the Resonance value to 3 (remember the center is actually at Res+1). For the rest of the harmonic arches, we'll use the RES2 waveform at harmonics 8, 13, 20, and 29. Adjust the widths of the harmonic patterns using the Skirt setting by watching the results on the spectrum analyzer and adjust the amplitudes to match the size of the amplitude drops on the chart below. You can almost set the whole patch up without listening to it, it's that easy.

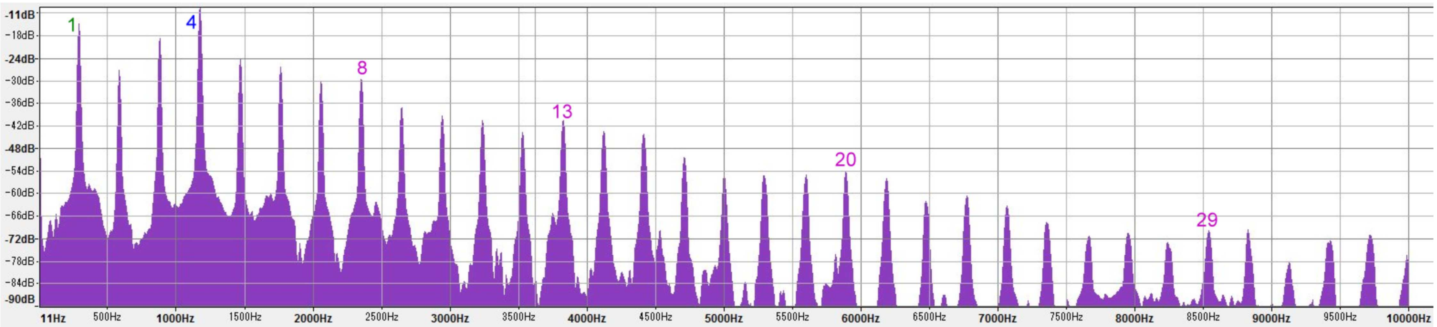


Figure 10. The English Horn Harmonic Spectrum

Operator	**Type**	**Freq**	**Res**	**Skirt**	**Vol**	
Op1	Sine	1.00			85	Fundamental Harmonic #1
Op2	RES1	1.00	3	3	99	Creates a triangle of harmonics at #4
OP3	RES2	1.00	7	2	85	Creates an arch of harmonics at #8
Op4	RES2	1.00	12	7	65	Creates an arch of harmonics at #13
OP5	RES2	1.00	19	2	57	Creates an arch of harmonics at #20
Op6	RES2	1.00	28	7	50	Creates an arch of harmonics at #29

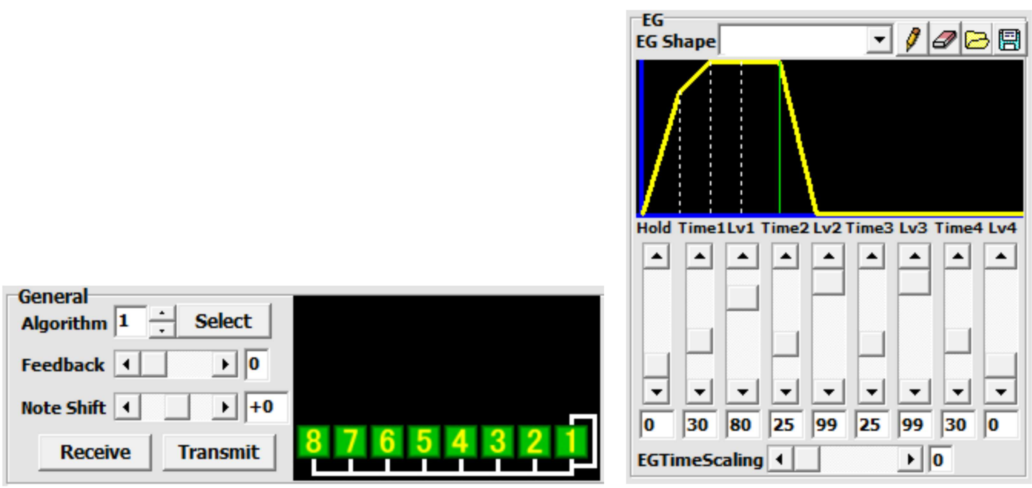


Figure 11. Algorithm #1 and the Operator Envelopes

Once you get the harmonics and the envelopes set up, listen to the recording and your patch, one right after the other, to determine what it needs most. Make changes as necessary to correct the patch. While listening to a different patch, the Oboe patch, it sounded a bit thin and needed more of the lowest two harmonics. I added in another RES2 Operator targeting the 2nd harmonic to add some volume to harmonics 1 & 2 to richen up the patch. You can see this patch in the table in Appendix C in the back.

For the English Horn, it took a few minutes of messing with the envelopes to make it sound more like the recording. It still isn't quite perfect, but the RES1/RES2 method doesn't allow you to modify the harmonics as the notes develop so there wasn't much more I could do. A better method would be to use Algorithm #6. Set up a Sine wave at the harmonic you want to target, then modulate it so you can control the harmonic fan. Do this for the first two groups of harmonics so you can use an envelope to modify them as the note progresses, then go back to the simpler RES1/RES2 method for the rest of the harmonics.

Example 2……………………………………………………………………………………… French Horn

Let's model a Brass instrument this time, the French Horn. Unfortunately the easy RES1/RES2 method we used last time doesn't work at all, it ended up sounding like an artificial organ patch. I tried a second time using a mixture of methods and it came out sounding more like a violin, too many high frequency harmonics. I went after a third audio spectrum chart with a clear note higher up in the instrument's register. I also went back to a more traditional FM method, which requires feedback on one pair of the Operators to get a brassy sound out of it. That one worked.

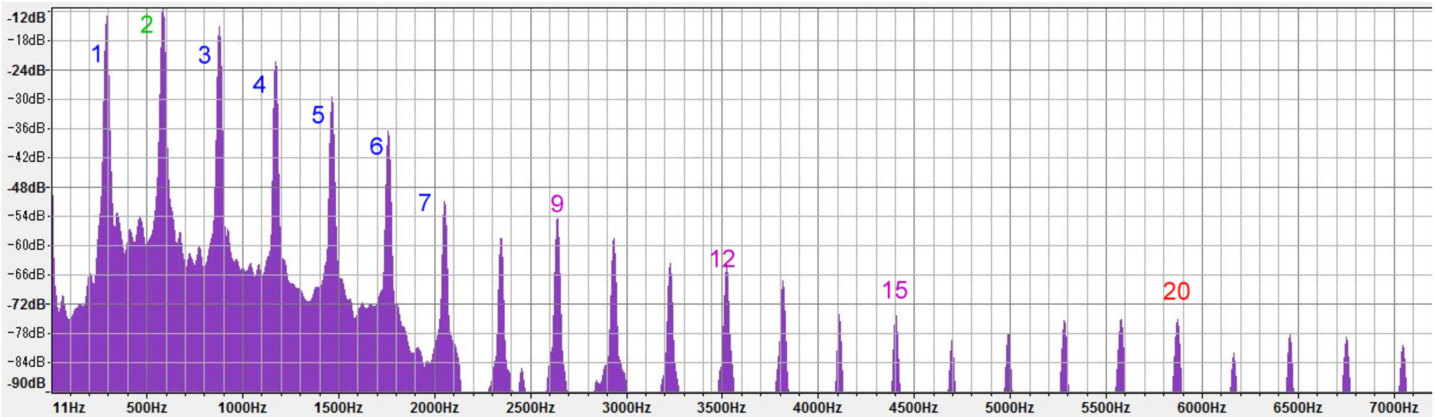
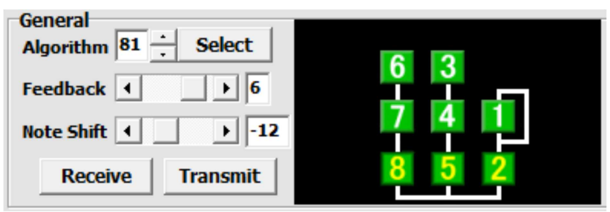


Figure 12. The French Horn Harmonic Spectrum

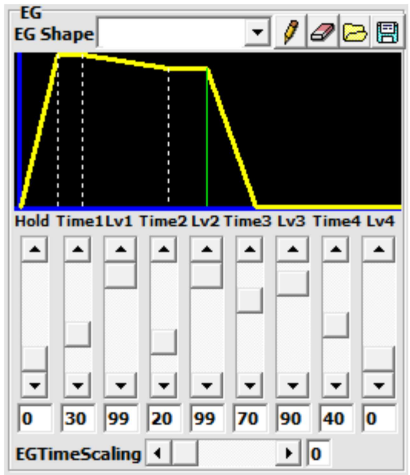
Most brass instruments can be modeled with a single FM pair, with feedback on the modulator. We'll use most of Algorithm 81 this time. Op2 will be the main carrier set at 2.0 to match the highest green peak on the chart above, then it will be modulated with Op1 set to 1.0 with a Feedback level of 6 to create the harmonic fan on the spectrum chart above in Blue. Adjust the Feedback up and down so you can hear what it contributes, it adds the brassiness into the sound.



Algorithm #81

Operator	**Type**	**Freq**	**Res**	**Skirt**	**Vol**	
Op2	Sine	2.00			99	Main Carrier in Green
Op1	Sine	1.00			75	Creates harmonics 1,2,3,4,5,6,7 in Blue, Fbk=6
OP5	Sine	9.00			45	Anchors the set on Harmonic #9
Op4	Sine	3.00			80	Creates harmonics 9,12,15 in Magenta
OP3	Sine	1.00			80	Fills harmonics in between 9,12,15
Op6					0	
Op7					0	
Op8	RES2	1.00	19	4	40	Creates an arch of harmonics at #20 in Red

Set up the second set of harmonics next, to catch the small high frequency peaks around 9, 12, and 15 using the Op3-4-5 stack. These don't contribute much to the patch, just a few high frequency overtones. You could skip these and you wouldn't even notice the difference much. I also threw in Op8 to create the last arch of very high frequency harmonics just for demonstration purposes, but you can't really hear them, not at -75 db. Turn Op5 and Op8 on and off to hear the difference.



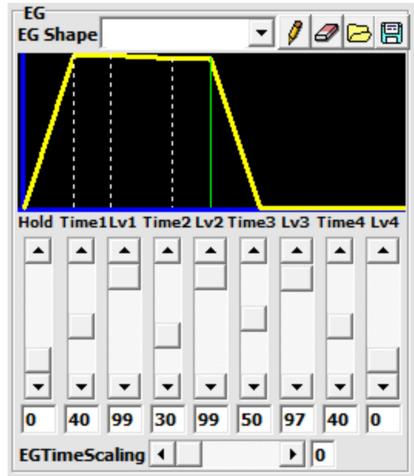


Figure 13. Carrier Envelopes   Modulator Envelopes

As a slight enhancement, add in a little pitch slide at the beginning of the notes, starting from an Lv0 of -2, and sliding up to a Lv1 of 0 in a Time1 of 49 with the Pitch Envelope Generator (PEG). Vibrato helps too, as does a sensitivity of 4 on the AmpVelocity of the carriers.

To create a **Trumpet** patch, lower Op2 to 1.0, raise the Feedback to 7, and you're done.

Example 3…………………………………………………………………………………… Saxophone (ver 2)

Now, let's attempt the second hardest instrument to model there is, the Saxophone (the most difficult is the Piano). It took me forever to get this one figured out, but in the end I think I got it close. I reworked this one quite a bit to fix it. The part everyone seems to miss is the high frequency rasp in the instrument's voice. Without the raspiness, it just doesn't sound right no matter what you do. It needs a LOT more of the high frequency harmonics than any other instrument. The envelopes are critical too, the sound takes time to develop as the reed works up to full vibration. For this patch, start with the frequency spectrum chart below and match the harmonics as closely as possible graphically before trying to tune the sound by ear.

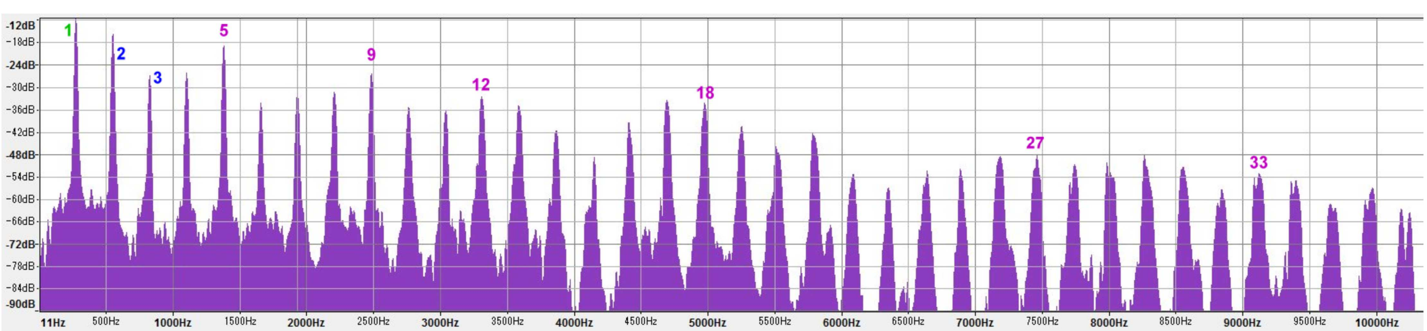
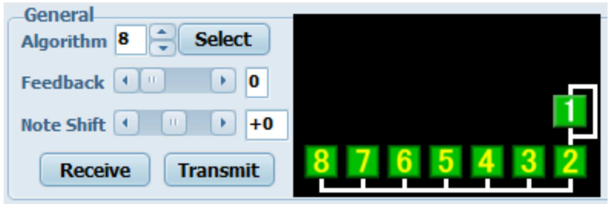


Figure 13. The Saxophone Harmonic Spectrum Revisited



Algorithm #8

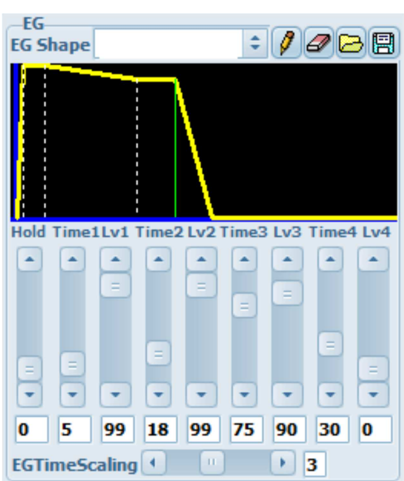
First, we'll model the first three harmonics using a standard FM technique and Algorithm #8. The first three harmonics in green and blue we'll model using a 1:1 FM pair using Op1 and Op2 both at a ratio of 1.0. Adjust the output volume of the Synthesizer so that harmonic #1 hits -12 db in the spectrum analyzer with the Operator set to an output volume of about 94. This makes matching the rest of the amplitudes off the chart above a bit easier. Now adjust the volume of Op1 to get the pattern close to the first three harmonics above in blue on the chart.

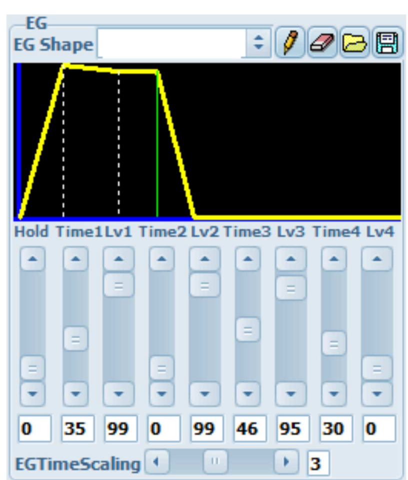
Operator	**Type**	**Freq**	**Vol**	
Op2	Sine	1.00	94	Main Carrier for the Fundamental #1
Op1	Sine	1.00	71	Creates harmonics 1, 2 & 3

Following that, we'll use the RES1 and RES2 waveforms to match the rest of the harmonics. The final values are tabulated below. Set the peaks for each center harmonic so you can form a triangle or arch of harmonics at each location (tabulated below). Until you add the highest frequency groups in the patch doesn't sound like a Saxophone at all. The higher harmonics add in the growl of the reed. I also boosted their volume after initially setting them up to increase the growl so it would sound more like a Saxophone.

Operator	**Type**	**Freq**	**Res**	**Skirt**	**Vol**	
Op3	RES1	1.00	4	0	88	Creates a triangle of harmonics at #5
OP4	RES1	1.00	8	3	84	Creates a triangle of harmonics at #9
Op5	RES2	1.00	11	0	68	Creates an arch of harmonics at #12
Op6	RES2	1.00	17	2	75	Creates a triangle of harmonics at #18
Op7	RES2	1.00	26	4	68	Creates an arch of harmonics at #27
Op8	RES2	1.00	32	4	59	Creates an arch of harmonics at #33

I used three different envelope shapes for the Saxophone patch, one for the Op1, another for the Op2, and a third for the high frequency groups of harmonics created by Operators 3, 4, 5, 6, 7 & 8. The objective was to try and model the hesitation in the reed vibration at the start of each note.





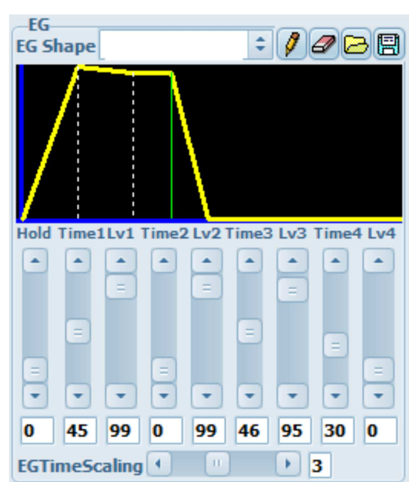


Figure 14. Op1 Envelope Op2 Envelope Op3, 4, 5, 6, 7, 8 Envelopes

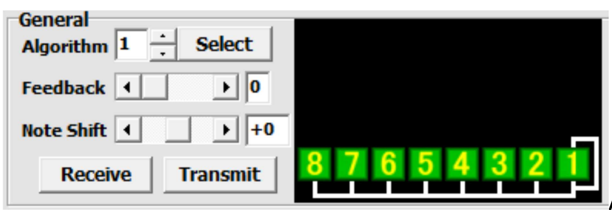
I also added a pitch slide into the notes using the PitchEG. It starts from an Lv0 of -4, then slides quickly with a Time1 of 56 up to a Lv1 of 0. This helps the realism of the model, since the Saxophone recordings have a bit of pitch uncertainty at the start of each note. It would also be great to add in After-Touch control to the patch, allowing you to change the timbre of the notes and the volume as you change the pressure on the keys to add to the expressiveness of your playing.

The Saxophone is notoriously difficult to model with FM. There aren't any decent FM Saxophone patches out there, not that I could find. Overall I think the patch sounds OK, especially the lowest notes on the keyboard. It makes a great bass Saxophone! But like all of the patches, it could still use more work. None of our patches are perfect, but you can get pretty close if you put some effort into it.

Example 4…………………………………………………………………………………… Violin

Let's do one more just for good measure. When I ran the recording of a Violin through the spectrum analyzer I noticed the formants were moving to the right as the Violin played up the scales, very unlike a wood bodied instrument. Later I realized it must be a recording of an electric violin, which doesn't have a resonating body. Electric violins have pickups like an electric guitar mounted near the base of the bridge, or sometimes in the bridge itself. The result is, FM can model the electric violin very closely since the harmonics follow the pitch of the note. You can tell the difference between different types of Violins using spectrum analysis, which is quite interesting.

The tactic we'll use this time is a little different, but not by much. I hope you're seeing there are a lot of different ways to combine our building blocks to construct instrument patches. This time we're going to use simple sine waves for the first two harmonics, then we'll use arches of harmonics created by the Res2 waveform to model the rest of the structure you see below. This is basically the same technique we used on the English Horn.



Algorithm #1

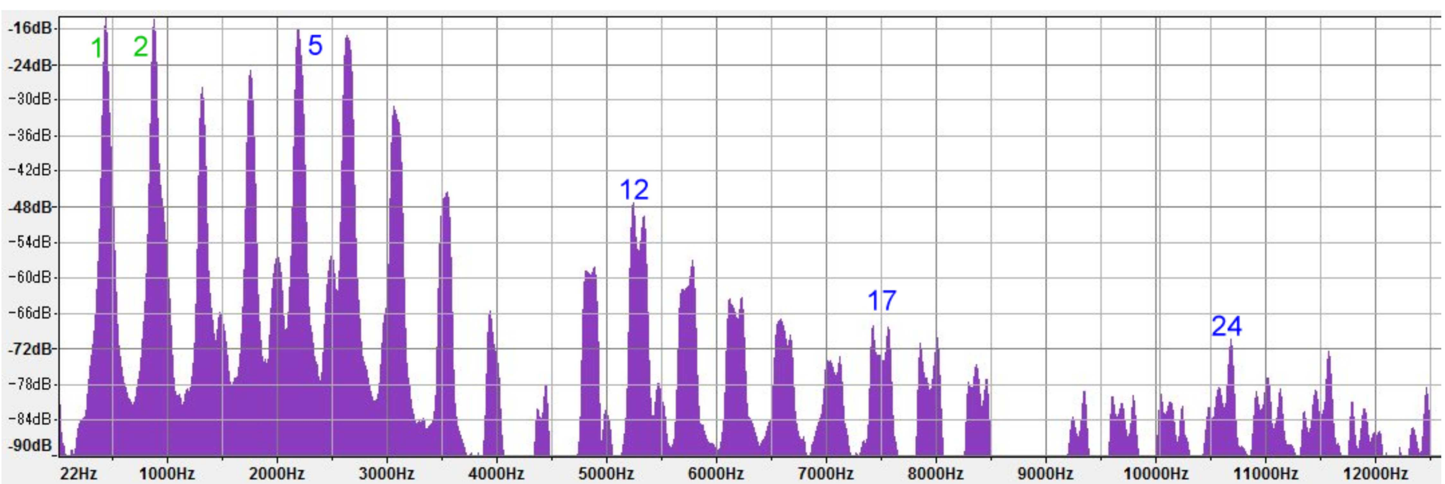


Figure 15. Violin Harmonic Spectrum Chart

Operator	Type	Freq	Res	Skirt	Vol	
Op1	Sine	1.00			87	Fundamental Harmonic #1 in Green
Op2	Sine	2.00			87	Harmonic #2 in Green
OP3	RES2	1.00	4	2	99	Creates an arch of harmonics at #5
Op4	RES2	1.00	11	2	70	Creates an arch of harmonics at #12

OP5	RES2	1.00	16	2	55	Creates an arch of harmonics at #17
OP5	RES2	1.00	23	2	52	Creates an arch of harmonics at #24
Op7	RES2	1.00	32	2	48	Creates an arch of harmonics at #33

I had to reduce the volume of the first two Sine waves a bit after adding in Op3, since Op3 was maxed out at 99. A pure Sine wave has more volume than the arch of harmonics created by Op3 and they needed about the same amplitude in order to sound right. Add in the other arches of harmonics next, then adjust their amplitude visually using the spectrum analyzer. As a note, Op7 at harmonic #33 is off the chart to the right and is the highest frequency set of harmonics we've used so far. The Violin has a high frequency raspiness to it that needed to be added in. On the FS1R I would have used one of the un-voiced operators which are variable frequency noise generators, but FM-X doesn't have that option.

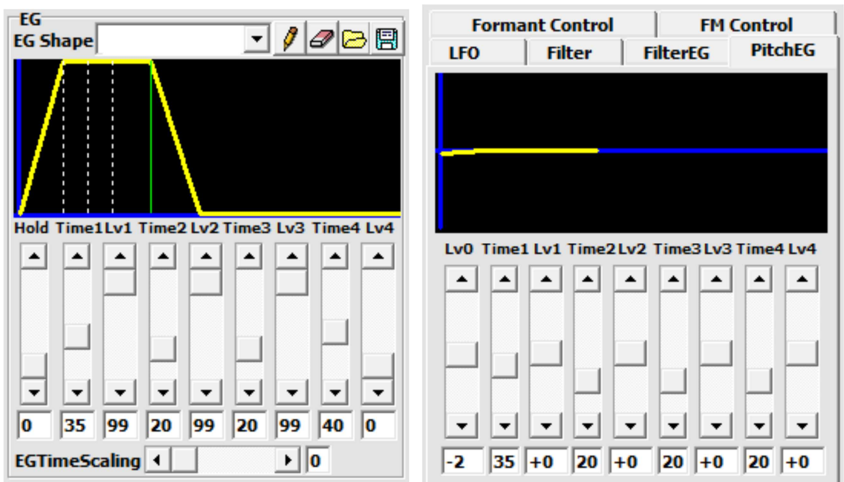


Figure 16. Violin Envelopes Pitch Envelope

The volume envelopes for all of the Operators are the same this time, and a slight pitch slide was set up on the first part of each note to match the sound of a Violin player.

That wasn't too hard, was it? I realize you don't really need to model any of these instruments on the Montage keyboards since the AWM2 patches are so precise, but it is useful to know how to match a stringed instrument if a custom string patch is what you're after. You can use a patch like this one to springboard off of, to create something completely unique. Your own custom sound is the goal, right? Customization is the name of the game with FM.

20. Formants Anyone? The FS1R

Trying to create a high quality Bassoon patch was driving me up the wall. I could get a good Bassoon sound for about one octave with FM, but it kept falling apart outside that narrow range of notes. I suppose you could set up a different patch for each octave and set up a Performance to manage them, but there is a much better way to handle it on the FS1R. When you spectrum analyze the sound of a Bassoon playing different notes up a scale, you can see that a different harmonic is dominant on every other note up the scale. The harmonics move right and widen apart as the notes increase in frequency, but the amplitude groupings stay in the same place/frequency at the same width. How do you model that? Well, that's exactly what a formant does.

But first, let's discuss two major classes of instruments for a minute. We're going to divide instruments up into two categories, instruments without resonating bodies like a pipe organ or piano, and a second class with resonating bodies for instruments like the violin. Instruments <u>without</u> resonating bodies include the organ, piano, marimba, the brass instruments, the electric guitar, and the clarinet, saxophone and the flute fit into this category. These instruments have a tone generator that changes dimensions for each and every note they play. On a marimba the wooden keys change size for each note. On a pipe organ, there is a different length pipe for each note. On the electric guitar, the strings change length for each note as you move your fingers. On a flute, the tube changes it's effective length for every finger hole. Frequency Modulation works great on these instruments since FM creates a harmonic structure that follows each note.

The second class of instruments are those <u>with</u> resonating bodies. Violins, violas, cellos, and basses all have a wooden body that is fixed in size, it doesn't change size for each note. Bassoons and a few other reed instruments are the same, using a fixed resonating body. The acoustic guitar is also in this category, along with the human voice tract. You can also see formants in the lowest registers of some of the Brass instruments. FM doesn't work all that well on most of these instruments. You can match the character of the instrument over about one octave with FM, but the match falls apart as you get further and further away from the center of the patch. That's because FM moves the harmonic structure around with the notes, it doesn't model the fixed nature of the resonating body. For that hat trick you need to use Formants.

FS1R

So what are Formants you ask? Well, the simple answer is a Formant is a custom waveform. It's created by using a waveform like the ALL1 waveform which has all of the harmonics in it, then you add a band pass filter to it with a fixed center frequency. The fixed frequency filter mimics the resonance of the cavity, like you have with a cello. The cavity is a fixed size, hence the fixed frequency of the band pass filter. The sound going through it, the vibrations from the strings of the cello, resonate in the cavity. The cavity resonance gives the instrument it's unique sound. When you mix four or five Formants together at the right set of frequencies and amplitudes you get the signature sound of a Cello, the Bassoon, or the vowel sounds of speech.

The FS1R has this unique capability, which most people seem to overlook and don't understand. It has Formants built right in! Yamaha really came up with a powerful combination with this innovation. By combining Formants with FM the FS1R can model almost every possible type of instrument there is. Personally, I think integrating formants is the single biggest innovation in FM in decades. Yamaha really hit the ball out of the park with this one!

It surprises me that Yamaha didn't include many formant patches in the factory set to model real acoustic instruments. Almost all of the formant patches are synthesizer sounds, vox patches and sweeps. In fact, most of the factory patches came up from the DX7 and use only standard FM. We can fix that.

Free FS1R Editor (the one I'm using): http://synth-voice.sakura.ne.jp/fs1r_editor_english.html

An Improved Commercial Editor (42 Euros): http://zeeedit.free.fr/

Example 1………………………………………………………………………………… Bassoon

The Bassoon is one of the harder instruments to model, so let's start with that one. We can put together an FS1R patch using the formants obtained from the frequency spectrum chart from the Bassoon shown in Figure 17 below. Using the cursor in Audacity, map out the peaks of harmonic groups in the spectrum, both the center frequencies and the amplitudes for the formant groups. The formants form the arches in the spectrum chart below. These arches are fixed on the harmonic spectrum chart. As the instrument plays up a scale the little harmonic spikes spread out and move, but the overall shapes and arches stay in the same place. That's the behavior of a Formant.

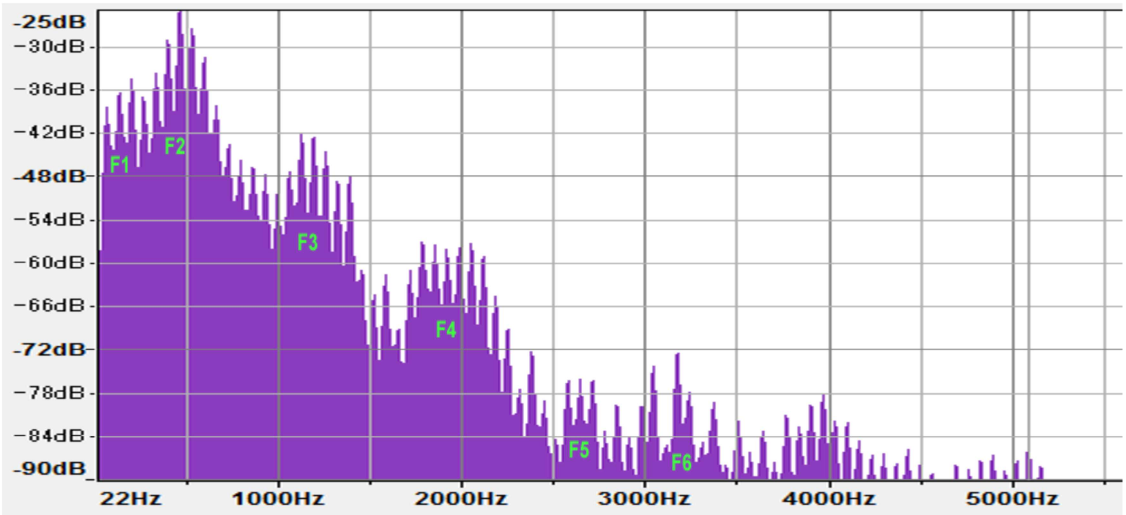
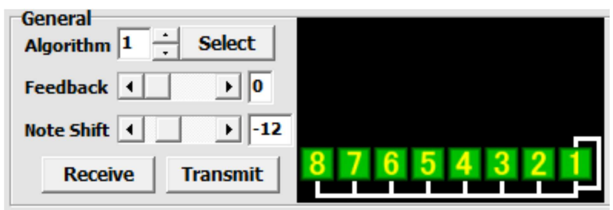


Figure 17. The Bassoon Harmonic Spectrum, Formants in Green

Select Algorithm 1 next and we'll get started building our patch. We want to add all of the formants together to get our sound, so we'll use Algorithm 1 which does this for us on the FS1R. Note that I ended up shifting the patch down one octave so I could hit some even lower notes on my keyboard by using a NoteShift of -12 (see below).



Algorithm #1

I came up with the following set of frequency groupings off the spectrum chart in Figure 17. The location of each formant group is labeled in green on the chart. F7 would have been too quiet at -78 db and too high pitched to hear so I skipped it.

Formant	**Frequency**	**Amplitude**	**Adjusted**	**BandWidth**
F1	197	-35 db	86 (121-35=86)	30
F2	462	-22	**99 (121-22=99)**	40 (or swap to a Skirt of 3 instead)
F3	1192	-42	79 (121-42=79)	40
F4	1984	-57	64 (121-57=64)	60
F5	2645	-72	44 (turned off)	35
F6	3176	-71	50 (turned off)	40

'Adjust' the amplitudes by adding a larger number to them (121 in this example) so the highest number in the adjusted set hits 99 so they can be used to set up the synthesizer. These will become the volumes we will start from for each of the formants we're going to define. Now select 'frmt' as the wave form for six of the voiced Operators, setting the frequency near the frequencies in the table above using both the coarse and fine sliders in the editor. Set the volumes to the 'Adjusted' values in our table.

Next, you have to go through each one, one at a time, and listen while trying out different band widths. The band widths I found that worked the best are listed in the table above. Too low/narrow and you hear some nasty metallic resonances, too high/wide and the tone changes from a nice ‘ah’ sound to and higher toned ‘ee’ sound. The Bassoon has a low ‘ah’ sound to it. Adjust these, one voiced Operator at a time, by muting the others momentarily while you make adjustments. F5 and F6 didn’t contribute much that was audible, so I turned them off by setting the output volume to 0. You can also set the BandWidths by watching the results in the spectrum analyzer. You will also have to re-adjust the volumes for each Formant now that the BandWidths have changed them on you.

Making adjustments to the shape of the Formant waveform is done using two different parameters, the BandWidth and the Skirt setting. The BandWidth parameter creates an arch-shaped group of harmonics. The higher the bandwidth, the wider the rounded arch gets. The Skirt setting creates a triangle-shaped group of harmonics. The higher the number the wider the triangle gets. Use the Skirt setting for emphasizing the center harmonics if they stick out prominently like they do on F2 since the BandWidth setting can round them off too much. When you combine the BandWidth with Skirt at the same time, you get an arched top with triangular flares on the sides. We’ve used just the BandWidth setting for our Bassoon.

The sound envelopes are next. Go in and set up the EG Shapes for each Operator, setting all of them up the same. I used the values below. These settings mimic a reed instrument like the Bassoon.

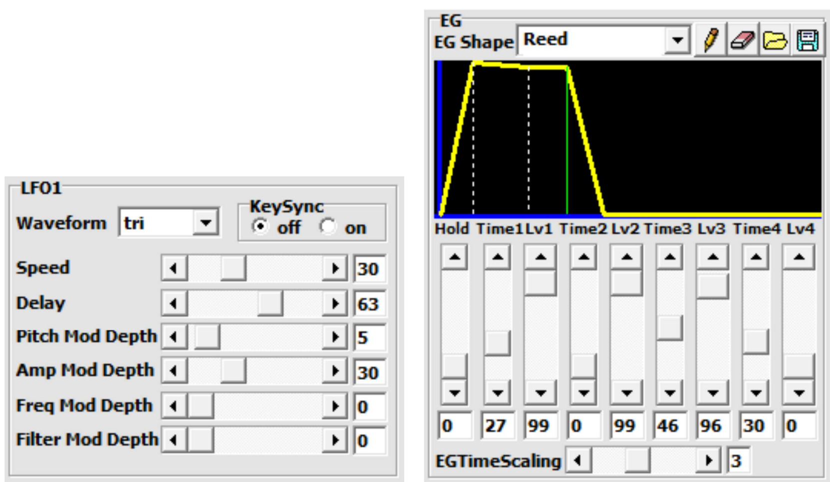


Figure 18. LFO1 and Envelope Generator Settings

For keyboard control, for <u>each carrier Operator</u> set the AmpVelocity to +4, the AmpMod = 4, and the PitchMod = 4, which are located just to the right of the Envelope Generator panel in the free software editor. For vibrato, set LFO1 up like I did in the figure above. We’re going to create a vibrato using both the pitch and amplitude of the sound. The delay is set long so vibrato will start in the second measure of a piece, giving us a flat Bassoon sound until a very long note is held. Adjust these to your liking. You can also trigger vibrato using the Modulation Wheel on the ‘Performance Control’ panel on the FS1R, but we won’t go over that. You can play with that on your own.

Pretty stellar Bassoon patch isn’t it, all done using formants! You could also add in a little bit of air hiss noise at the start of the notes using a couple of the Unvoiced Operators just for fun. You could also modify the envelopes for a few of the formants like we did the modulators on the 4-OP synth patches to add more character to the patch. Make sure you save your patch in one of the Internal Voice slots so you don’t lose your work.

Another nice effect you can create is to use mixes of instruments, taking on qualities of each to create something brand new. Mix the new Bassoon patch in a Performance with the DX-Clari 2 (H-88) voice patch which is much throatier in the lower registers to get a new reed instrument with a unique sound. Layering similar instruments in a Performance can really increase the depth of the sound.

Example 2……………………………………………………………………………………. Cello

For round 2, let's try a Cello. I also play the Cello, by the way. Something interesting showed up in the spectrum analyzer when comparing the Cello to the Violin and Viola, the Cello harmonics are muted somewhat. Now why would that be? Well, think about how the Cello is held, it's gripped between the knees and held into the chest. This damps some of the harmonics the wood body of the instrument is capable of compared to a Violin or Viola. If the musician were to let go of the instrument it would ring more freely, brightening up the tone of the instrument. Something you cellists should think about.

This patch was more difficult to zero in on, it took a lot of fiddling with the bandwidths before it started to sound like it should. I should have started it visually using the real-time spectrum analyzer. After reviewing the patch, several of the chart formants proved to be very soft when soloed by themselves, so F5, F6, F7 and F8 could be turned off. On second thought, they may need to be left on since they come more into play when the Cello plays high notes. We should probably turn the two formants in the Bassoon patch back on as well.

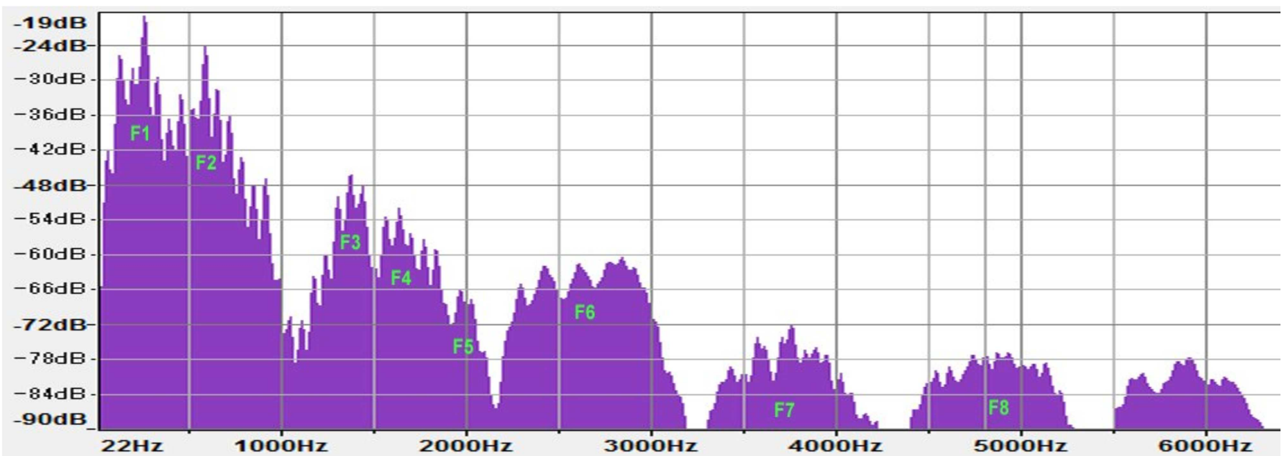
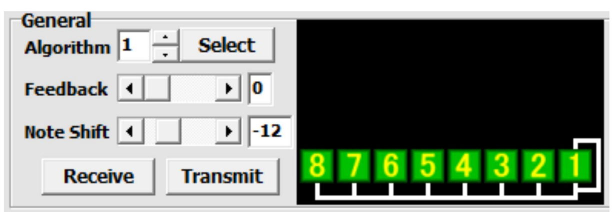


Figure 19. The Cello Harmonic Spectrum, Formants in Green



Algorithm #1

Formant	**Frequency**	**Amplitude**	**Adjusted**	**BandWidth**
F1	262	-19	**99 (118-19=99)**	60
F2	589	-24	94 (118-24=94)	47
F3	1371	-45	73	50
F4	1635	-52	66	66
F5	2024	-67	51	35
F6	2606	-61	57	55
F7	3758	-71	42	55
F8	4869	-76	40	55

After doing a few more Formant patches I've noticed that the Adjusted volumes are not quite correct. The fundamental frequency formant on the left of the chart needs to be boosted some in volume, so I've been adding in another Formant at the same frequency using a Skirt setting of about 3 to boost the volume of the loudest harmonics. Also, the high frequency Formants on the right are too loud sometimes and need the volume reduced a bit. These adjustments are best made using a real-time spectrum analyzer, comparing the differences in amplitude of the real-time peaks to the recording plot. Calculate the db drop between a main peak and the next one down, then adjust the volumes to get the correct <u>difference</u> in db. On the chart of the recording above, the F2 peak is at -24 db and the F3 peak is at -48 db. The difference is a drop of 24 db, which is what you target on the spectrum analyzer to adjust the volume of F3. Got it?

You might have noticed that the Cello patch sounds a bit too clean. It's too nice and neat, it doesn't have enough variation in it to match the recording of the real instrument quite like we want it to. To fix this, go into a few of the Operators and detune them to add some variation into the patch. You can leave the two main formants alone, but detune the higher pitched ones a couple of units up or down.

The envelope settings are shown below, along with the LFO1 vibrato settings.

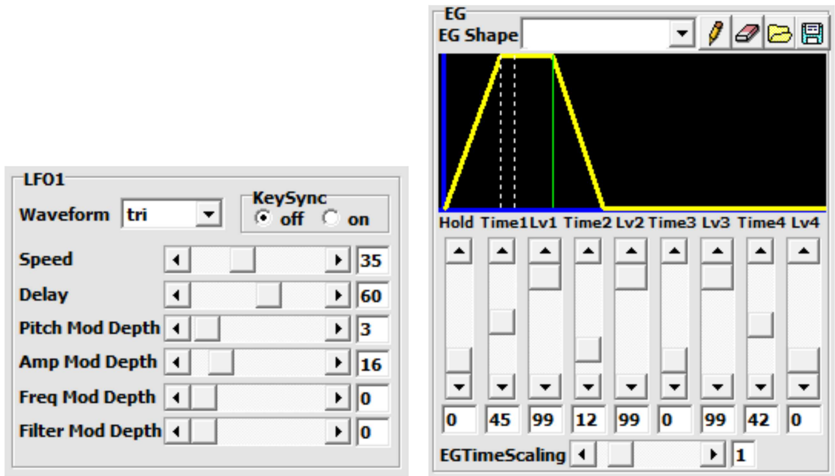


Figure 20. LFO1 and Envelope Generator Settings

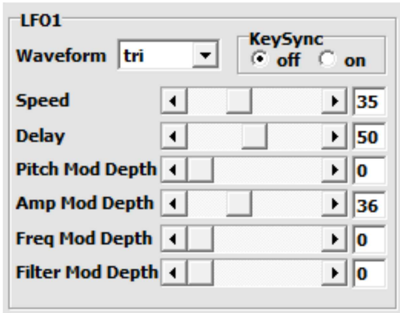
Example 3……………………………………………………………………………………… Soprano "Hoo"

For round 3, let's try something completely different and set up a voice/vox patch. This is what formants were mainly developed for, modeling human speech.

When most of us sing we select the pitch with our vocal chords and that's it. To sing louder we increase the air volume which is more akin to screaming. Listen to rock singer like Axel Rose or the lead singer from AC/DC Brian Johnson, this is the result of too much air flow on the vocal chords. A trained singer like Luciano Pavorotti or Christina Aguilera however, tune the cavities in their vocal tract in addition to selecting a note with their vocal chords. They achieve a quality of note far and above the rest of us and much more volume. The harmonic resonances pop out well above the background formants creating a clear, pure tone.

Looking at the spectrum chart for our female singer patch below, you can see the main harmonic spikes are very prominent over the formant shapes. The harmonics on the plot are harmonics of the note she is singing, with the set being the spikes at F1, F2, F4, F5, and so on, all evenly spaced. We'll model these using a 1:1 FM pair like we've done before in previous sections.

For the second part of the patch, we'll model the formants underneath using the FS1R formant waveforms. The formants shape the sound underneath, at F1, maybe F2, and under F3. The formant under F5 is so weak you'll never hear it. There may also be a formant around 3000 hz which is called the "singer's formant" but I didn't keep it in the patch. The patch sounded cleaner without it. For a more realistic, raspier voice, leave the 3000 hz formant in the patch. You can't really hear it much, not at -78db, but you could increase the volume on purpose to add some character into the patch. As it is now, it comes out sounding very flute like. First we're going to select Algorithm 6 to test with.



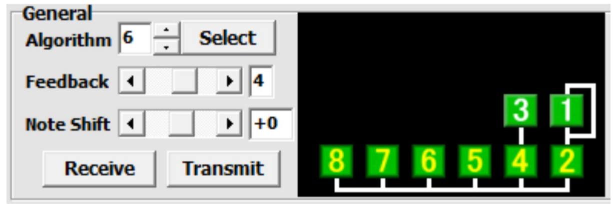


Figure 21. LFO1 Settings

Algorithm #6 Selection

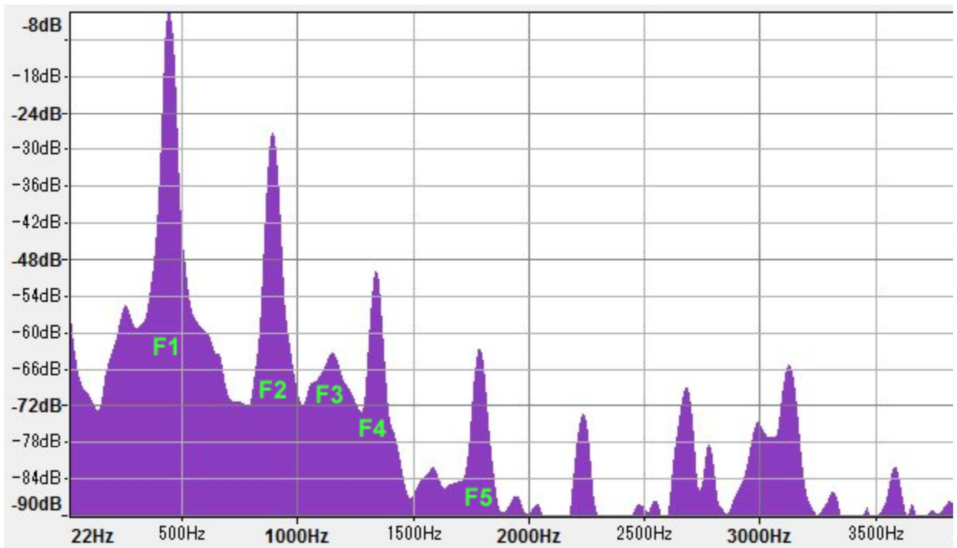


Figure 22. The Soprano Harmonic Spectrum, Singing 'Hoo'

Operator	**Wave**	**Freq**	**Vol**	
VOp2	Sine	1.00	99	Main Carrier, at F1
VOP1	Sine	1.00	65	Modulator, Creates the harmonics F1, F2, F4, F5
VOp5	frmt	300 hz	99	Voice Formant 1 near F1
VOp6	frmt	700 hz	80	Voice Formant 2 near F2
VOp7	frmt	1100 hz	50	Voice Formant at F3

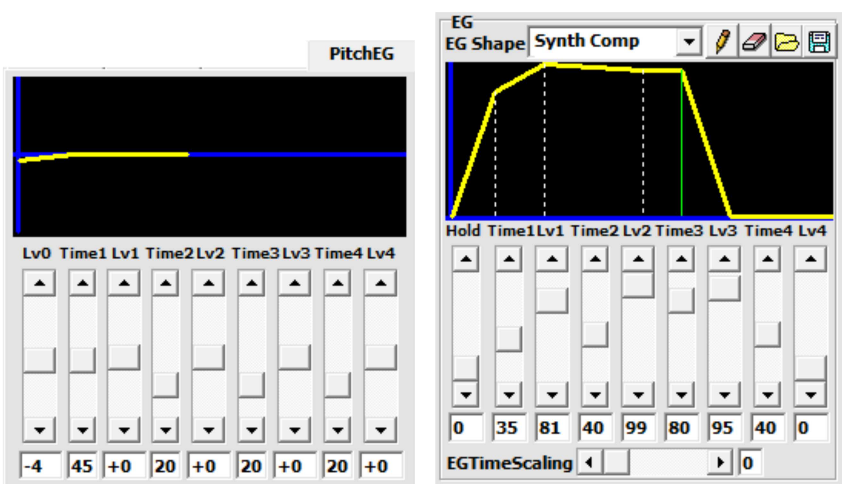


Figure 23. Pitch EG and Voiced Operator Envelope Generator Settings

The Pitch Envelope Generator (PEG) in Figure 21 is set up to give some variability to the first part of each note, much like a singer uses to correct the pitch as she sings a note. I also used the exact same pitch slide on my Trombone patch, which worked quite well. It enhances my string patches a bit too, using an LV0 of -2 or -3.

Let's discuss the human voice for a minute. The vocal tract is shaped by the lips, jaw, tongue, and throat to form chambers that alter/filter the sound produced. These chambers are simulated like we did the fixed body shape of the Cello using a formant. Voice formants create different vowel sounds, which are outlined in the table below in Figure 25. Vowels can be distinguished by using just the first two voice formants (out of five) which is really fascinating when you think about it. You can change the frequencies of voice formants 1 and 2 allowing you to change which vowel sound the patch sings, which is what I did above. I selected the formant frequencies for a strong 'oo' sound off the chart below, 300 hz and 700 hz. Voice Formants move as you articulate speech, which we're going to ignore for now.

Vowel sounds seem to be shaped more by the difference between the first two formants, which is 400 hz in this case (700-300=400). A third formant was selected at 1100 hz in order to reinforce the 400 hz difference, and to possibly clarify the vowel enunciation some. Your voice box can't do this, but we can on our synthesizers. This seemed to work.

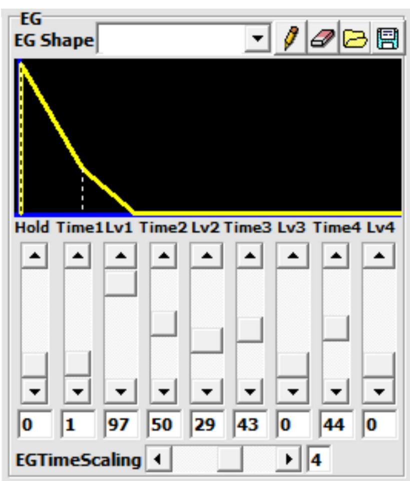


Figure 24. Unvoiced Operator Envelope

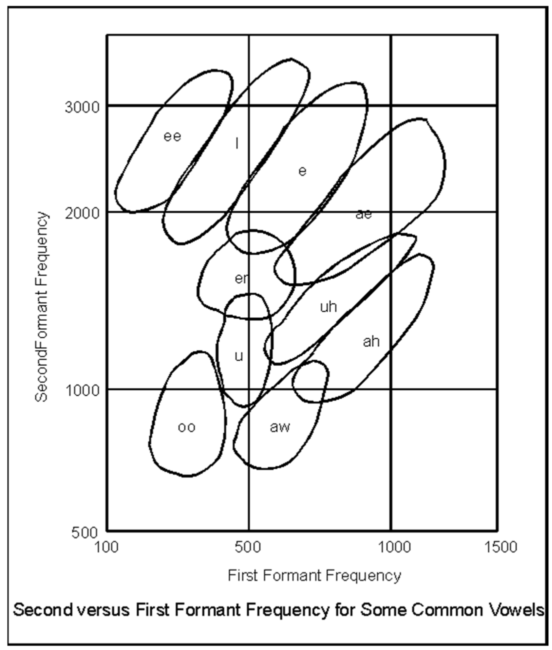


Figure 25. Formant Vowel Sound Settings

Two additional features that are used to enhance most of the Yamaha factory vox patches are vibrato and a slight pitch change on the first part of the note, which we discussed above. The settings for LFO1 and the PEG are shown in the figures above. We're also going to add a bit of breath noise to the patch, using two Unvoiced Operators. The envelopes for the Unvoiced Operators are shown in Figure 24 above.

Operator	**Freq**	**Vol**	
UOp1	552 hz	69	Breath Noise, "H" in the word Who
UOP2	880 hz	42	

For a male voice patch, listen to the 'Man_Eh' patch which is Voice Preset B-92 on your FS1R synth. It has a fantastic male bass voice in the lower register. Like the Soprano patch we developed, it works well over about two octaves, which is pretty wide for a singer. Christina Aguilera has about a 2-1/2 octave voice range, which is exceptional.

Playing with just this one aspect of the FS1R will keep you busy for a while. Try changing the vowel to an 'ah' or an 'e' just for fun by changing the frequencies for the F1, F2, and F3 voice formants.

Example 4……………………………………………………………………………………… Christina1

The last voice patch wasn't really all that convincing was it, it lacks realism. After a number of tries on the FS1R to model voices, I finally bought a Yamaha PLG100-SG card so I could learn how Yamaha sets up life-like voice patches on it. That worked. I learned a lot by analyzing the voices on the SG card, and was able to improve on them with the FS1R. The SG card voices have a kind of tinny sound to them though, and only sound good over a very narrow range of notes. Similar patches on the FS1R are much better behaved, and sound cleaner and more realistic. This patch is one of my own, I analyzed the voice of Christina Aguilera and selected her lower voice register as a model for this patch. It doesn't sound exactly like her, but it is a very high quality voice patch. I think you'll like it. Here's a link to a Jazz piece using the Christina1 patch. https://www.youtube.com/watch?v=IvKTYFWN2Uk

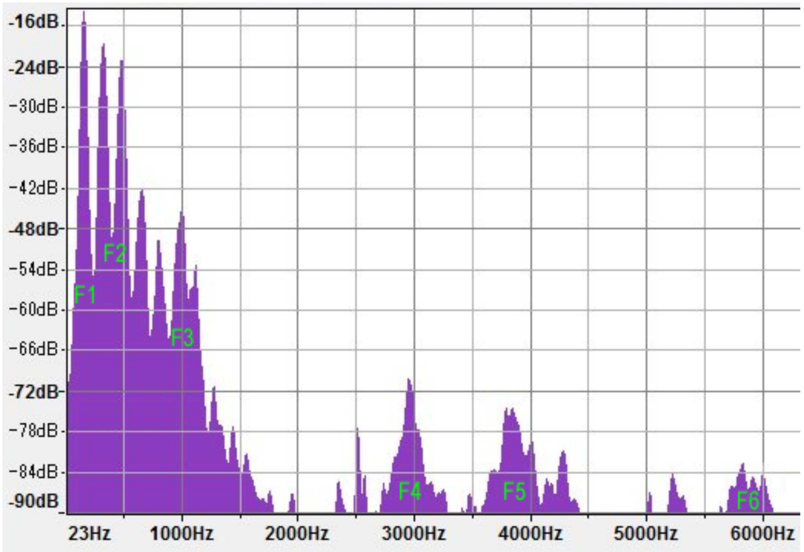


Figure 26. Christina1 Spectrum Chart

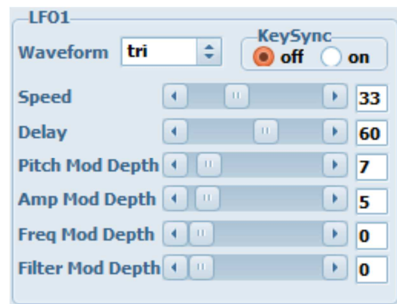


Figure 27. LFO1 Vibrato Settings

Operator	Wave	Freq	Vol	BandWidth	
VOp1	frmt	180 hz	85	20	Lowest Voice Formant F1
VOp2	frmt	485 hz	83	30	Voice Formant F2
VOp3	frmt	1008 hz	61	41	Voice Formant at F3
VOp1	frmt	2961 hz	23	40	Voice Formant at F4
VOp2	frmt	3840 hz	20	20	Voice Formant at F5
VOp3	frmt	5827 hz	20	50	Voice Formant at F6

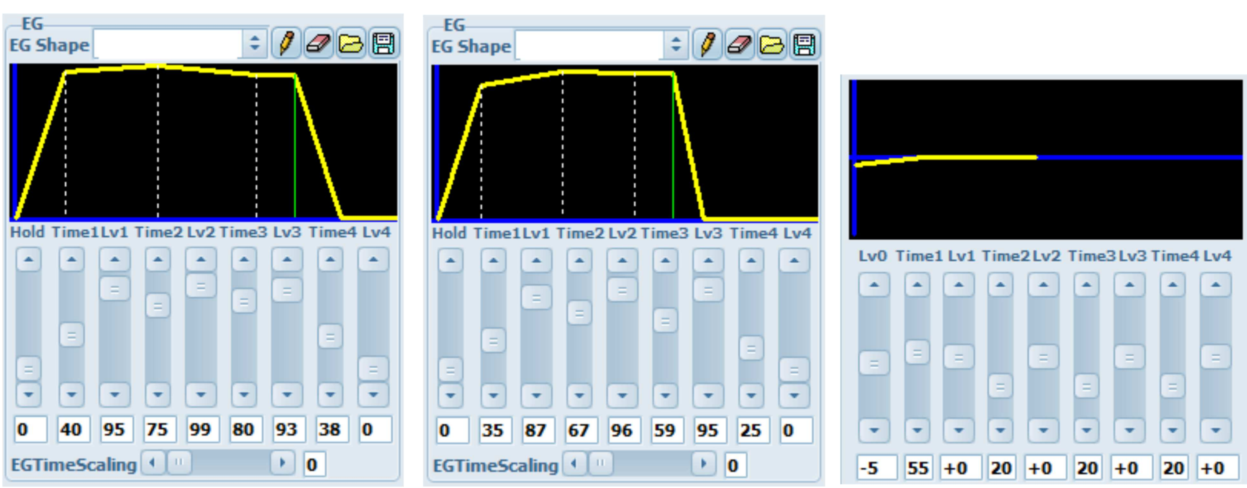


Figure 27. Voiced Operator Envelopes and Un-Voiced Operator Envelopes, and PitchEG Settings

Realistic voices are much more difficult than you think. The key seems to be in the control characteristics of the patch more than the timbre of the formants. It requires a LOT of Portamento. Turn the Portamento ON, select 'mono' for the playing mode, set the level to about 75, and select 'fingerd' so it only triggers a Portamento slide when you play connected notes. The notes need to overlap in the midi file in order for this to work. Leave in a gap where you want to start a new musical phrase (with no Portamento).

The patch also requires quite a bit of reverb, I set the send level to 80 and used 'Hall1'.

To add in some air hiss to the voice, use the Un-Voiced Operators. Set them all to 'LinkFF' so the white noise is in the same frequency bands as the Formants are. Set the levels all to around 15 and leave the bandwidths all at 20. You can go through each one and fine tune the volumes to further tune the breath noise.

21. FS1R Tips & Tricks

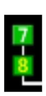
Trick 1……………………………………………………………………………………… Formant Sequences

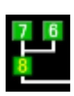
When you get tired of fixed tone formants, the FS1R has a few more tricks up it's sleeve. The biggest one is that it can play back entire formant sequences where every aspect of the formants in the synth can be scripted. It can simulate speech or sing phrases this way. Download and test out the FSeqEdit programs available on the web, which can take a recording and turn it into a formant sequence you can play back on the synth. They can also turn recordings of more modern, changing instrument patches like sweeps and morphing pads into FSeq's as well, with interesting results. There is also a Flash FSeq Creator I found on the web which allows you to hear your scripted sequences as you edit them, before you go to the trouble of exporting them to the FS1R. I recommend the Fseq Editor I programmed, it has better tonal quality than the others. It's only on Windows, though. Sorry Apple users.

Thor's FS1R Fseq Editor:	http://javelinart.com/fs1r-fseq-editor.html
Nifterick's FS1R FSeq Editor:	https://niff.home.xs4all.nl/fs1r/
Flash FSeq Creator:	http://blog.zacharcher.com/2011/03/14/a-formant-sequence-editor-in-flash/

Trick 2……………………………………………………………………………………… Modulation Bite

The FS1R has less bite when modulating FM Algorithms, not as much growl as the older instruments. A volume of 99 on the FS1R is only equal to about 91 on a DX7, and only about 83 on a DX100. To increase the modulation bite, double up on modulators. Use two modulators in parallel to modulate one carrier. This increases the amount of modulation (the modulation Index) substantially.

In place of 

use this: 

Trick 3……………………………………………………………………………………… Unvoiced Operators as Oscillators

Unvoiced Operators on the FS1R can be used as additional voiced operators if you get the settings right. This provides you with <u>8 more</u> Sine wave oscillators to use in Additive Synthesis, for a total of 16! Select an Unvoiced Oscillator, set the Bandwidth to 1 and the Resonance to 7 to stabilize the tone, then increase the volume so you can hear it. Set the frequency to 261.6 hz for the Fundamental Harmonic or select a frequency off the chart below. You now have a fixed frequency oscillator. To get the oscillator to follow the keys as you play, increase the FreqScaling up to 99. On the spectrum analyzer this looks just like a 1.0 ratio Sine wave. Tuning the oscillator to the harmonic you want it to follow is a bit trickier than simply selecting a ratio, use the chart below to tune your oscillators to the correct harmonic.

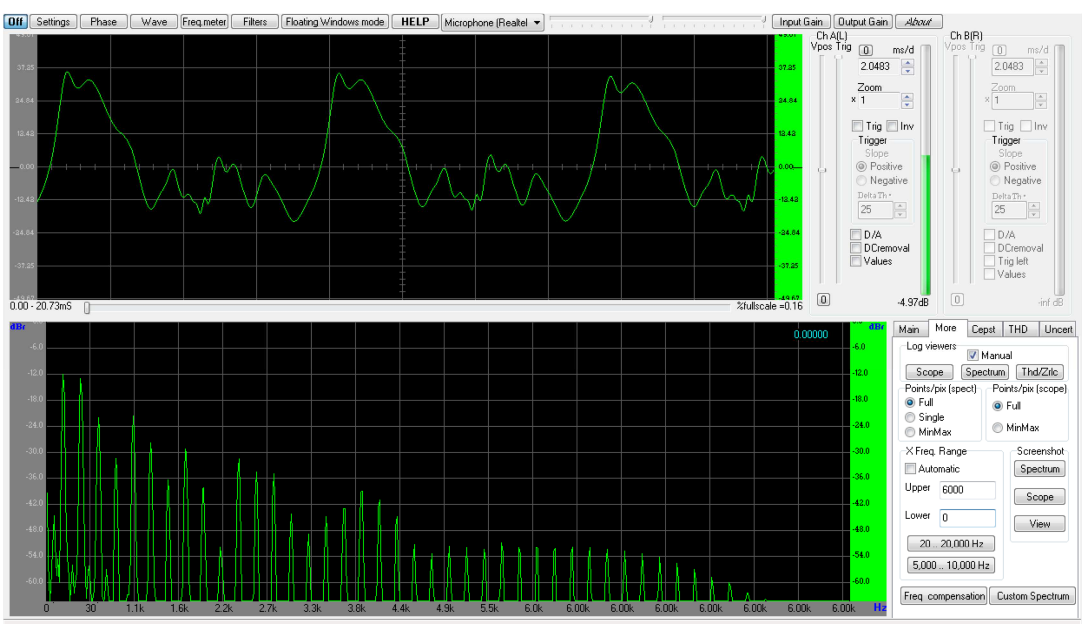
Harmonic	Note	Freq	Interval
1	C4	261.6	Unison, Middle C
2	C5	523.3	Octave
3	G	784.0	Perfect Fifth
4	C6	1047	Octave
5	E	1319	Major Third
6	G	1568	Perfect Fifth
7	Bb	1865	Minor Seventh
8	C7	2093	Octave
9	D	2349	Major Second
10	E	2637	Major Third
11	F#	2960	Augmented Fourth
12	G	3136	Perfect Fifth
13	A	3520	Minor Sixth
14	Bb	3729	Minor Seventh
15	B	3951	Major Seventh
16	C8	4186	Octave

Example: 4th Harmonic	
Freq	1047
FreqScaling	99
BandWidth	1
Resonance	7
Skirt	0
Output Level	80

I use this trick all the time on complex patches. I run out of Operators and need more harmonics to accomplish my goal without resorting to layering in a second patch in a performance. Sometimes I start by using these as sine wave Operators right from the start, knowing I'm going to run out of Operators later on.

22. Afterward

We covered a lot of ground in this guide, hopefully we didn't go too fast. I put in lots of examples to try and illustrate the concepts of programming an FM synth, and you'll find a lot more in the appendix to work with. I hope you find this instruction book useful, I know I learned a lot of things I didn't know while putting this together. Hopefully you didn't get too confused as I rattled on. Learning how to construct a patch from scratch all by yourself is a major accomplishment. Pat yourself on the back! You are now one of a small, select handful of people who actually know how to program an FM synth!



The FM-X / FS1R Saxophone Patch

I thought about putting together a section on special effects, the effects you use to get sweeps and to morph your patches to add trajectories into the sounds. The problem with that is it's very hardware specific. It's accomplished differently on every synthesizer due to the hardware. This guide is meant to be broader in scope than just one synthesizer, so I decided to leave that out of this guide to keep it more general. I hope you don't mind.

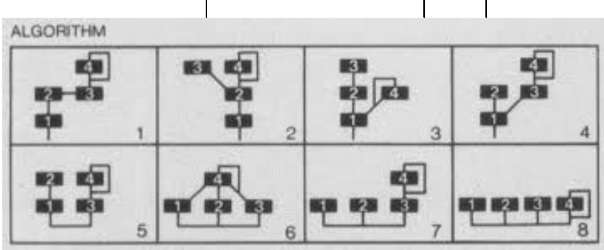
Maybe I'll write an Advanced FM manual for the FS1R later on. I'm working on duplicating the Yamaha VP1 patches on the FS1R right now, which is really stretching the envelope. The timbre of the VP1 changes dramatically as you go up the keyboard and is really making me think. For example, on the low end the ResoMetal patch has a plucked Piano tone which turns into an Asian bell sound like a Gamelan above C3. Morphing the harmonics this dramatically as you go up and down the keyboard is quite a challenge.

Have fun!

Best Regards,
Thor Z.

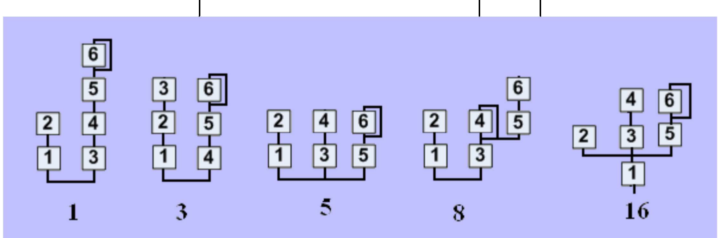
APPENDIX A. DX11 Instrument Patch Examples (4-OP)

Name:	Alg	F1	F2	F3	F4	Fbk	vol1	vol2	vol3	vol4	W1	W2	W3	W4	AR	D1R	D1L	D2R	RR	
ZBassoon	5	5	1	2	1	4	99	76	94	71	1	2	2	1	14	31	15	2	9	Carr.
															14	31	15	0	8	Mod.
ZClarinet	5	3	2	1	2	4	90	75	99	69	1	1	1	1	13	15	14	2	9	Carr.
															13	15	15	0	8	Mod.
ZFlute	5	2	2				99	73	0	0	1	1			13	13	14	2	9	Carr.
															12	13	14	7	8	Mod.
ZFluteWood	3	1	1	1	3		90	65	50	20	1	1	1	1	12	15	15	0	9	Carr.
															31	10	14	2	8	Mod.
ZEnglHrn	5	4	1	1	1	4	99	78	94	85	1	1	1	1	14	31	15	2	9	Carr.
															14	31	15	0	8	Mod.
ZOboe	1	2	3	1	1	4	99	69	67	68	1	1	1	1	15	31	15	0	9	Carr.
															15	31	15	2	5	Mod.
ZSaxophone	3	1	4	1	2	5	92	66	73	66	1	1	1	1	13	31	15	2	9	Carr.
															14	4	12	0	8	Mod.
ZTuba	3	1	1	1	1		99	69	54	99	1	1	1	1	15	31	15	0	8	OP123
															31	23	0	0	8	Op4
ZTrombone	3	2	1	1	1		99	66	67	60	1	1	1	1	14	31	15	1	8	OP123
															31	16	10	1	8	Op4
ZFrenchHrn	3	2	1	1	1		99	66	58	50	1	1	1	1	16	31	15	1	8	OP123
															31	15	8	1	8	Op4
ZTrumpet	3	1	1	1	1		99	80	60	99	1	1	1	1	16	31	15	0	8	OP123
															31	23	0	0	8	Op4
ZBassBowed	3	2	6	1	1	1	99	72	62	65	1	1	1	1	14	31	15	2	8	Carr.
															14	31	15	0	8	Mod.
ZBassPizz	3	2	6	1	1	1	99	60	69	69	1	1	1	2	25	15	15	13	9	OP13
															11	14	12	0	8	OP24
ZCelloViola	3	1	7	1	1	1	99	46	78	78	1	1	2	2	14	31	15	2	8	OP13
															15	31	15	2	6	OP24
ZCelloPizz	5	1	1	2	1	1	99	64	75	67	1	4	1	2	25	22	15	15	8	OP13
															11	21	15	13	6	OP24
ZViolin	3	1	4	1	1	1	99	73	50	74	1	1	2	1	14	31	15	2	8	OP13
															15	31	15	2	6	OP24



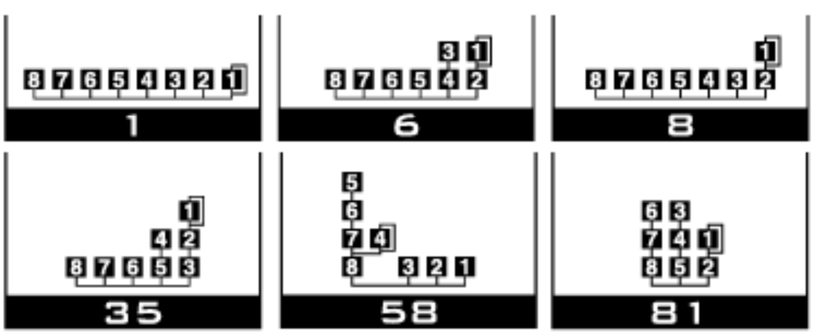
APPENDIX B. DX7 Instrument Patch Examples (6-OP / Converted 4-OP Patches)

Name:	Alg	F 1	F 2	F 3	F 4	F 5	F 6	Fbk	Vol 1	Vol 2	Vol 3	Vol 4	Vol 5	Vol 6	R1	R2	R3	R4	L1	L2	L3	L4	
ZBassoon	3	5	1	2	2	2	1	4	99	84	60	94	60	79	52	99	16	64	99	99	91	0	Carrier
															52	99	10	58	99	99	94	0	Modul
ZClarinet	5			3	2	1	2	4	0	0	90	83	99	77	49	55	16	64	99	96	88	0	Carr.
															49	55	10	58	99	99	94	0	Mod.
ZFlute	5			2	2				0	0	99	81	0	0	49	49	16	64	99	96	88	0	Carr.
															46	49	31	58	99	96	80.5	0	Mod.
ZFluteWood	8			1	3	1	1		0	0	90	28	73	58	46	55	10	64	99	99	94	0	Carr.
															99	40	16	58	99	96	88	0	Mod.
ZEnglHrn	5			4	1	1	1	4	0	0	99	86	94	93	52	99	16	64	99	99	91	0	Carr.
															52	99	10	58	99	99	94	0	Mod.
ZOboe	1			2	3	1	1	4	0	0	99	77	75	76	55	99	10	64	99	99	94	0	Carr.
															55	99	16	40	99	99	91	0	Mod.
ZSaxophone	8			1	2	4	1	5	0	0	92	74	74	81	49	99	16	64	99	99	91	0	Carr.
															52	22	10	58	99	87	82	0	Mod.
ZTuba	8			1	1	1	1		0	0	99	107	77	62	55	99	10	58	99	99	94	0	OP345
															99	79	10	58	99	0	0	0	Op6
ZTrombone	8			2	1	1	1		0	0	99	68	74	75	52	99	13	58	99	99	92.5	0	OP345
															99	58	13	58	99	78	71.5	0	Op6
ZFrenchHrn	8			2	1	1	1		0	0	99	58	74	66	58	99	13	58	99	99	92.5	0	OP345
															99	55	13	58	99	70	63.5	0	Op6
ZTrumpet	8			1	1	1	1		0	0	99	107	88	68	58	99	10	58	99	99	94	0	OP345
															99	79	10	58	99	0	0	0	Op6
ZBassBowed	8			2	1	6	1	1	0	0	99	73	80	70	52	99	16	58	99	99	91	0	Carr.
															52	99	10	58	99	99	94	0	Mod.
ZBassPizz	8			2	1	6	1	1	0	0	99	77	68	77	85	55	49	64	99	99	74.5	0	OP35
															43	52	10	58	99	87	82	0	OP46
ZCelloViola	16	1		1	7	1	2	1	99	0	86	54	86	60	52	99	16	58	99	99	91	0	OP14
															55	99	16	46	99	99	91	0	OP356
ZCelloPizz	5			1	1	2	1	1	0	0	99	72	75	75	85	76	55	58	99	99	71.5	0	OP35
															43	73	49	46	99	99	74.5	0	OP46
ZViolin	8			1	1	4	1	1	0	0	99	82	81	58	52	99	16	58	99	99	91	0	OP35
															55	99	16	46	99	99	91	0	OP46



APPENDIX C. Montage and FS1R FM Instrument Patch Examples (8-OP)

Name:	Alg	F1	F2	F3	F4	F5	F6	F7	F8	Fbk/Shft	vol1	vol2	vol3	vol4	vol5	vol6	vol7	vol8
ZBassoon	1	1	1	1	1	1	1				99	70	64	60	43	40		
		Res1	Res1	Res2	Res2	Res2	Res2											
ZClarinet	8	1	1								45	99						
			Odd2															
ZFlute	8	1	1								87	99						
ZEnglHorn	1	1	1	1	1	1	1				85	99	85	65	57	50		
			Res1	Res2	Res2	Res2	Res2											
ZOboe	1	1	1	1	1						83	99	70	45				
		Res1	Res1	Res1	Res1													
ZSaxophone	8	1	1	1	1	1	1	1	1		71	94	88	84	68	75	59	68
				Res1	Res1	Res2	Res2	Res2	Res2									
ZTuba	6	1	1	1	4	1				6/-12	80	99	48	60	99			
ZTrombone	6	1	2	1	4					7/-12	77	99	60	72				
ZFrenchHrn	81	1	2	1	3	9			1	6	75	99	80	80	45			40
									Res2									
ZTrumpet	81	1	1	1	3	9			1	7	75	99	80	80	45			40
									Res2									
ZBassBowed	58				1		1	6	2	7/-12				80		62	72	99
ZBassPizz	58				1		1	6	2	7/-12				75		62	72	99
Zcello	6	1	1	1	6	1	1	1	3		75	99	70	80	65	52	99	80
			All2			Res2	Res1	All1										
Zviola	6	1	1	1	5	4	1	1	1		82	99	55	90	91	80	61	60
		All2					Res1	Res2	Res2									
ZStringPizz	6	1	5	1	8	1	2	3			62	65	75	45	75	95	50	
ZViolin	1	1	2	1	1	1	1	1			87	87	99	70	55	52	48	
				Res2	Res2	Res2	Res2	Res2										

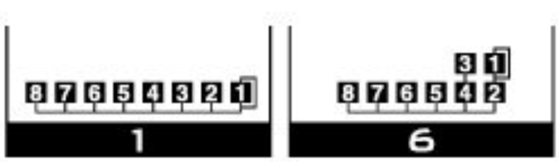


Resonance/Skirt

Name:	RS1	RS2	RS3	RS4	RS5	RS6	RS7	RS8	hld	Tm1	Lv1	Tm2	Lv2	Tm3	Lv3	Tm4	Lv4	Op
ZBassoon	3/2	7/1	10/1	15/2	19/2	26/3			0	30	73	25	93	25	99	30	0	
ZClarinet		/2							0	30	84	25	99	25	99	30	0	
ZFlute									0	8	99	0	99	60	80	33	0	1
									0	38	99	0	99	65	99	33	0	2
ZEnglHrn		3/2	7/2	12/2	19/4	28/4			0	30	80	25	99	25	99	30	0	
ZOboe	1/1	3/1	7/2	13/2					0	30	80	25	99	25	99	30	0	
ZSaxophone			4/0	8/3	11/0	17/2	35/4	26/4	0	5	99	18	99	75	90	30	0	1
									0	35	99	0	99	46	95	30	0	2
									0	45	99	0	99	46	95	30	0	3-8
ZTuba									0	30	99	20	95	67	90	40	0	1-4
									0	10	99	0	99	0	99	25	0	5
ZTrombone									0	40	99	64	98	60	97	40	0	1
									0	30	99	20	99	67	91	40	0	2-4
ZFrenchHrn								19/4	0	30	99	20	99	70	90	40	0	2,5,8
									0	40	99	30	99	50	97	40	0	1,3,4
ZTrumpet								19/4	0	30	99	20	99	70	90	40	0	2,5
									0	40	99	30	99	50	97	40	0	1,3,4
ZBassBowed									0	45	99	20	99	20	99	40	0	6,7
									0	35	99	20	99	60	90	45	0	4,8
ZBassPizz									0	13	99	20	99	60	60	40	0	4
									0	13	99	20	99	20	99	40	0	6,7
									0	13	99	9	99	58	23	40	0	8
ZCelloViola		/0			11/2	15/0	/0		0	35	99	20	99	20	99	40	0	
Zviola	/0					9/2	13/2	18/3	0	35	99	20	99	20	99	40	0	
ZStringPizz									0	8	98	20	92	45	0	35	0	1-4,7
									0	8	98	20	92	50	0	35	0	5,6
ZViolin			4/2	11/2	16/2	23/2	32/3		0	35	99	20	99	20	99	40	0	

APPENDIX D. FS1R Formant Instrument Patch Examples (FS + 8-OP)

Name:	Alg	F1	F2	F3	F4	F5	F6	F7	F8	Fbk/Shft	vol1	vol2	vol3	vol4	vol5	vol6	vol7	vol8
FBassoon	1	188	462	1192	1983	2643	3177			4	86	99	79	64	44	50		
FEnglHorn	1	294	1179	2359	3010	3840	5891	6782			85	99	78	69	59	55	30	
FOboe	1	1070	2082	3010	2657						99	73	75	54				
FFrenchHrn	6	1.0	2.0			482	1849			7	80	99			99	62		
FTrumpet	6	1.0	2.0				1365			7	80	99				90		
FBassBowed	1	302	150	302	532	1205	2005	2805		/-12	80	99	98	91	71	69	40	
Fcello	1	261	589	1372	1761	2027	2600	3758	4874		99	94	73	66	35	55	42	40
Fviola	1	440	440	752	1951	3010	4010				90	99	86	74	51	44		
FViolin	1	440	520	2082	5008	7807					99	99	91	65	42			
Soprano	6	1.0	1.0			301	709	1105			65	99			99	70	50	
Christina1	1	180	485	1008	2961	3840	5827				85	83	61	23	20	20		



	Bandwith/Skirt																	
Name:	Bw1	Bw2	Bw3	Bw4	Bw5	Bw6	Bw7	Bw8	hld	Tm1	Lv1	Tm2	Lv2	Tm3	Lv3	Tm4	Lv4	Op
FBassoon	30	40	40	60	60	70			0	27	99	0	99	46	95	30	0	
FEnglHrn	45/2	0/3	60	60	50/2	70	50		0	40	99	0	99	99	85	36	0	1-4
									0	48	99	0	99	90	97	36	0	5-7
FOboe	40/3	20/2	65/2	60					0	30	99	0	99	90	95	36	0	
FFrenchHrn					/4	/3			0	8	99	66	92	60	90	40	0	1
									0	30	99	20	99	67	81	40	0	2
									0	25	99	0	99	0	99	30	0	5,6
FTrumpet						/3			0	8	99	68	95	60	94	40	0	1
									0	30	99	20	99	67	81	40	0	2
									0	25	99	0	99	0	99	30	0	6
FBassBowed	/2	40	40	40	45	45	45		0	50	99	0	99	0	99	42	0	2-7
																		1
Fcello	60	47	50	70	99	70	56	67	0	48	99	0	99	0	99	42	0	
Fviola	/4	50	47	50	50	50			0	45	99	47	99	51	92	40	0	1
									0	45	99	0	99	0	99	42	0	2-6
FViolin	/2	60	60	60	60				0	45	99	0	99	0	99	42	0	
Soprano	60	47			50	70	65		0	35	82	40	92	80	99	40	0	1,2
									0	35	92	66	75	99	80	40	0	5-7
Christina1	20	30	41	40	20	50			0	40	95	75	99	80	93	38	0	1-6