YAMAHA

Creating New FM Voices

DX7 II·FD/D

DIGITAL PROGRAMMABLE ALGORITHM SYNTHESIZER

SUPPLEMENTAL BOOKLET

Welcome

This is the second in a series of Supplemental Booklets designed to provide a practical guide to creating your own sounds on the DX7 II. In this booklet, you will discover how to create new voices from scratch. Each section describes a major group of FM parameters and includes a step-by-step procedure illustrating the implementation of these parameters. By following these procedures throughout the booklet, you will create a new and fully programmed voice.

Section 1 provides a brief introduction and outlines the procedure for creating an "initialized" voice.

Section 2 describes the basic operator configurations and the process of algorithm selection.

Section 3 discusses the relationship between operator frequencies and output levels.

Section 4 presents the use of envelope generators.

Section 5 includes the concepts of keyboard level and rate scaling.

Section 6 describes the use of effects such as vibrato, pitch bend, and portamento.

For continuing information concerning the DX7 II FD/D, consult AfterTouch, the official publication of the Yamaha Users Group. Many advanced functions will be discussed in its pages in the coming months. There will also be information regarding the availability of other materials concerning more advanced applications. To receive a free copy of AfterTouch every month, send your request to AfterTouch, P.O. Box 7938, Northridge, CA 91327-7938. On your letter or postcard, be sure to indicate that you are the owner of a DX7 II FD/D.

Contents

1 Introduction

Creating new voices requires a certain understanding of the principles of FM synthesis. This can be achieved in two ways. Understanding the theoretical basis of FM synthesis is essential because it provides a foundation upon which practical experience can be built. Conversely, practical experience in the form of experimentation can also lead to theoretical understanding. However, it is only with practical experience that you can become a skillful FM programmer.

This booklet is intended to provide guidelines by which you can begin creating new voices. The emphasis is primarily on practical experience that illustrates theoretical concepts. It would be helpful if you were already familiar with the first booklet in this series, "Modifying Preset Voices," which serves as an introduction to the material in this booklet. This section presents an introduction to the concepts of FM synthesis on the DX7 II. It also describes the procedure for creating an initialized voice (commonly referred to as "INIT VOICE"), which serves as the starting point for creating a new voice.

Preliminary Information

The process of FM synthesis involves the modulation of one oscillator's frequency with the signal from another oscillator. In the DX7 II, oscillators are known as operators. Those operators which actually produce an audio output are called carriers. Operators which modulate the frequency of other operators are called modulators.

The overall sound that results from combining carriers and modulators depends on many factors, including the specific combination of operators. Another important factor is the output level of each carrier (which affects the volume of the sound) and each modulator (which affects the timbre of the sound). These factors determine the character of the sound, including the harmonic spectrum. As you may recall, any sound can be distilled into a series of sine waves with different frequencies and amplitudes. The original sound can be reconstructed by sounding its component sine waves simultaneously.

The parameters described in the following pages are accessed by pressing the appropriate button on the front panel. Some buttons display their parameters in several LCD "screens," which are selected by pressing the button repeatedly until the desired screen is displayed. The parameter to be edited is selected by positioning the cursor with the cursor keys. The parameter value is changed using the -1/+1 keys or the data entry slider.

Example Voice

In order to illustrate the concepts and techniques presented in this booklet, you will find a step-by-step procedure for creating a new voice. This procedure is presented in parts at the end of each section. Each part illustrates the implementation of the parameters described in that section. The new voice is a basic harpsichord sound. It can be used as is or further modified to suit your own tastes.

The order in which parameters are selected and specified in the course of creating the harpsichord voice is only one of many approaches to FM programming. In fact, some of the parameters are so closely related to each other that their values must be considered with respect to each other. As you will see, this is particularly true of algorithm selection and operator frequency and output level specification.

These concepts are presented separately in this booklet for the purpose of clarity and learning. In practice, knowing the specific harmonic characteristics of the various basic operator configurations is vital to selecting the appropriate algorithm. These characteristics are established by considering the frequency ratios between the operators within a configuration. However, the final operator frequencies are usually specified after an algorithm has been selected.

Don't worry if these concepts seem a bit overwhelming at the moment. They will become clearer as you read this booklet and create the harpsichord voice using the procedures found in each section. Be sure to play some notes as you follow the procedures. Also, feel free to explore values other than those given in the procedures to hear what they do to the sound. This will help you to form an intuitive basis for programming.

Note:

> *The programming procedures assume that you have initialized voice A as described below. They also assume that the DX7 II is in Single Voice Edit Mode. This is achieved by pressing the* SINGLE VOICE MODE SELECT *button followed by the* EDIT *button. Each procedure begins with a reminder to be in Edit mode.*

Initializing a Voice

Voices are created and edited in an area of the DX7 II internal memory known as the Play/Edit Memory, or edit buffer. The following procedure will "initialize" the edit buffer by setting all voice parameters to their default, or most basic values. This creates a voice commonly known as "INIT VOICE" (for "initialized voice"). From there, the world of FM programming awaits.

This procedure assumes that you are starting from either Performance or Voice Play mode. If you have been editing a voice and wish to return to it later, be sure to save it before following this procedure. Any voice in the edit buffer will be lost after initialization.

Procedure

1. Press the EDIT button.
2. Press the TUNE button (#14) repeatedly until the Initialize display appears.
3. Position the cursor at the VoiceA parameter if it isn't already there.
4. Press the +1/YES button to specify INIT VOICE in edit buffer A. The display will ask "Are you sure?"
5. Press the +1/YES button again to confirm your action.

Edit buffer A has now been initialized with the default voice data. The parameter settings are listed in the following table.

INIT VOICE Table

DX7 II-FD/D

Voice name : INIT VOICE
Date : / /

ALGORITHM	
ALG	1
FBL	0
OSC.Sync	ON
Transpose	C3
L F O	
Wave	TRI
Speed	35
Delay	0
Mode	SINGLE
PMS	3
PMD	0
AMD	0
Sync	ON
Pitch E G	
Range	8 oct.
Velocity	OFF
RS	0
R1	99
R2	99
R3	99
R4	99
L1	50
L2	50
L3	50
L4	50

OSCILLATOR OP	1	2	3	4	5	6
Mode	ratio	→				
Coarse·Fine	1.00	→				
Detune	+0	→				
E G OP	**1**	**2**	**3**	**4**	**5**	**6**
RS	0	→				
R1	99	→				
R2	99	→				
R3	99	→				
R4	99	→				
L1	99	→				
L2	99	→				
L3	99	→				
L4	0	→				
Output Level OP	**1**	**2**	**3**	**4**	**5**	**6**
Scaling mode	norm.	→				
Output Level	99	0	0	0	0	0
LD	0	→				
LC	-lin	→				
BP	C3	→				
RC	-lin	→				
RD	0	→				
Sensitivity OP	**1**	**2**	**3**	**4**	**5**	**6**
Velocity	0	→				
AMS	0	→				

Key mode	
Key assign mode	POLY
Unison detune	—
Pitch Bend	
Range	2
Step	0
Mode	normal
Portamento	
Mode	sus. key p retain
Step	0
Time	0
Random pitch S.	
Modulation Wheel	
P. MOD	0
A. MOD	0
EG. B	0
Breath Control	
P. MOD	0
A. MOD	0
EG. B	0
P. Bias	+0
After Touch	
P. MOD	0
A. MOD	0
EG. B	0
P. Bias	+0

Foot control 1	4
P. MOD	0
A. MOD	0
EG. B	0
P. Bias	0
Foot control 2	**7**
P. MOD	0
A. MOD	0
EG. B	0
P. Bias	0
MIDI IN control	
P.MOD	0
A. MOD	0
EG. B	0
P. Bias	0

This voice produces a pure sine wave with a square envelope (instant attack at key-on to maximum level sustain, with instant release at key-off).

Saving Voices

Creating new voices represents a significant investment of your time and energy. It is therefore a good idea to save your new voice periodically as you work on it in order to protect your investment against anything untoward such as a power failure. Voices are saved in the internal memory on in a RAM cartridge. These procedures are presented in the Supplemental Booklet "Memory Management." Please refer to this booklet for the voice saving procedures.

Comparing Versions

As you work on a voice, you may wish to recall the original settings of one or more edited parameters. You can temporarily reset the parameter values to those of the voice you originally selected for editing with the Edit/Compare function. This allows you to compare your latest edits with the last saved version of the voice. Simply press the EDIT/COMPARE button while in Edit Mode. The voice number in the LED display will blink, and the parameter values in the LCD display will reflect those of the latest saved version. Playing the keyboard allows you to compare the old sound with the new one. The edited values and sound are returned by pressing the EDIT/COMPARE button again.

2 Algorithm Selection

The six operators used in a voice can be combined into 32 different configurations of carriers and modulators. These configurations, known as algorithms, are depicted on the front panel of the DX7 II. They determine which operators act as carriers or modulators, as well as their specific combination. In the diagrams shown on the front panel, the operators in the lowest row are carriers. Those appearing above the carriers are modulators. The lines connecting the operators indicate the signal paths.

Algorithms are one of the most fundamental parameters in determining the final sound. This section describes the process by which algorithms are selected. Although this process depends in part on the concepts of operator frequencies and levels presented in the next section, you will find general guidelines for algorithm selection in this section.

Basic Configurations

Algorithms combine the operators by arranging them into one or more basic configurations, or "sub-algorithms." The waveforms of these configurations are mixed together to produce the final sound. There are several distinct types of basic configurations, which give rise to different types of harmonic spectra. The exact nature of these spectra depends on the specific combination of carriers and modulators, as well as the frequency and output level relationships between them. For the moment, the different types of basic configurations are described in general terms below. The spectrum which results from each one will be covered in greater detail in the next section.

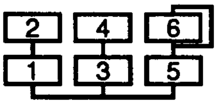
Single Operators

Some algorithms include a single operator. Such an operator acts as an unmodulated carrier, which produces a pure sine wave. Algorithm 32 consists of nothing but single operators.

Single operators can be used to include specific harmonic components in the spectrum of the final sound. In the case of algorithm 32, the entire spectrum is created by directly specifying the frequencies, output levels, and envelopes of the six operators. By using both voices of the DX7 II, a total of twelve operators can be used in this manner.

Simple Pairs

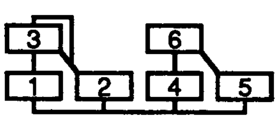
Simple pairs consist of a single carrier modulated by a single modulator. These pairs can be found in most of the algorithms. For example, algorithm 5 contains three simple pairs as illustrated below.



The spectrum resulting from each of these pairs depends on the relationship between the frequencies of the operators and the output level of the modulator.

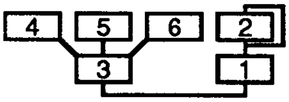
Parallel Carriers

Parallel carrier configurations consist of two or more carriers being modulated by a single modulator. This configuration behaves exactly as two simple pairs in which the modulators are set to the same frequency and output level. Algorithm 21 contains two sets of parallel carriers.



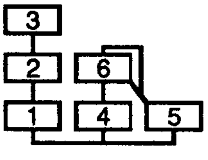
Parallel Modulators

Parallel modulator configurations consist of a single carrier being modulated by two or more modulators. Unlike parallel carriers, this configuration does not behave as a set of simple pairs, because the sine waves from each modulator combine to form a complex wave which is used to modulate the carrier. Modulation by a complex waveform produces a greatly increased number of frequency components in the resulting spectrum. Algorithm 12 contains three parallel modulators and a simple pair.



Stacked Modulators

Stacked modulator configurations consist of modulators being modulated by other modulators. Like parallel modulators, this produces a complex modulating waveform which serves to increase the number of frequency components in the harmonic spectrum. Many of the algorithms incorporate stacked modulators, including Algorithm 19, which consists of two stacked modulators and two parallel carriers.



Selecting an Algorithm

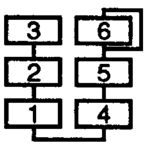
The process of creating a new voice must begin with a careful analysis of the sound you wish to create. This can be accomplished in several ways. For example, you can consider a voice as a sequence of sounds over time. If the beginning, middle, and ending portions of a sound are quite distinct, these elements can be recreated with different basic configurations within an algorithm. The quality of a sound in different pitch ranges should also be considered.

Once the desired sound has been analyzed, the algorithm best suited to its recreation can be selected. Try to find the basic configurations required for different portions of the sound in a single algorithm. Will a simple pair be sufficient for one portion of the sound? Will a particular portion require the broader control offered by a stack of modulators? Can an algorithm be found that offers a variety of means by which a sound can be created? These questions will be answered more easily after you become familiar with the concepts of operator frequencies and levels presented in the next section of this booklet.

Algorithm for Example Voice

The voice used as an example throughout this booklet is a basic harpsichord sound. Harpsichords exhibit the spectrum of the harmonic series in which the components are integral multiples of the fundamental. The overall timbre is somewhat thin and also quite bright, indicating gaps in the spectrum and audible high frequency components. The strings inside the instrument are set into vibration when they are plucked by the internal mechanism in response to the keys being pressed. When a key is released, the mechanism stops the vibration suddenly with a short buzzing sound.

Harpsichords often include two sets of strings, one set tuned an octave above the other set. This suggests an algorithm with two similar basic configurations which can be used to simulate the two sets of strings. The harmonic series is easily achieved with a simple pair, but the thin, bright quality and the buzzing sound at the end of each note must also be considered. Parallel or stacked modulators would allow flexible control over the various portions of the sound. This leads to algorithms 3 and 4. Since feedback on one modulator will probably be sufficient, algorithm 3 seems to be the best choice.



Procedure

1. Press the EDIT button if you are not already in Edit Mode.
2. Press the ALGORITHM button (#7).
3. Position the cursor at the Alg parameter and use the +1 button to select algorithm 3.

Naming Your Voice

The new voice is given a name in the algorithm display as well. The procedure for naming a voice is presented in the Supplemental Booklet "Voicing Parameter Reference Guide." Please refer to this booklet for the naming procedure. Name this voice something like "HARPSICHRD" (voices can be named with up to 10 characters).

3

Operator Frequencies & Levels

The operator frequencies and output levels are the parameters primarily responsible for the harmonic spectrum and resulting timbre of a voice. The tone color of any sound depends on the relationships between the frequencies of carriers and their modulators. Modulator output levels influence the number of harmonics present in a voice. These relationships form the heart of FM programming.

A presentation of the mathematics required to calculate the precise frequencies and amplitudes of the harmonic components resulting from a specific combination of operators is beyond the scope of this booklet. However, this section will introduce general guidelines which will help you develop a "feel" for the types of waveforms created by different operator frequency and level settings.

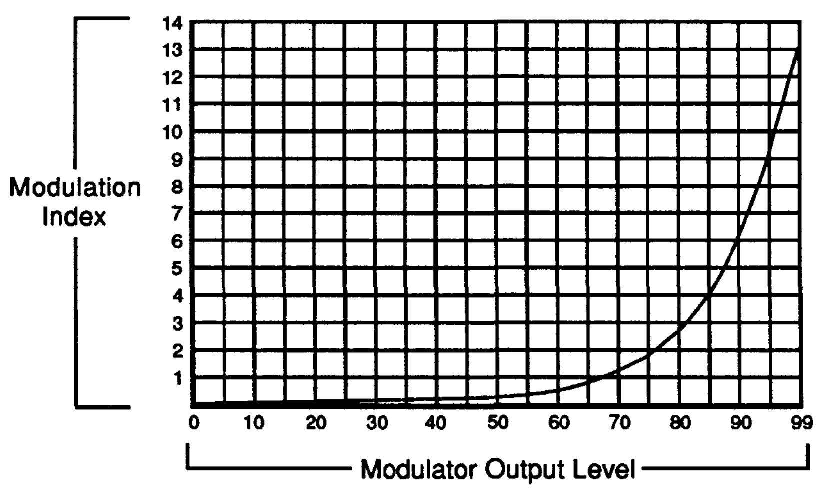
Frequencies & Output Levels

Whenever the frequency of one operator is modulated by a signal with a frequency in the audio range from another operator, an entire harmonic spectrum is generated. Sine waves appear with frequencies above and below the frequency of the carrier. These sine waves are known as side bands, because they appear on both sides of the carrier frequency. The ratios between the frequencies of the carriers and their modulators in any algorithm are instrumental in determining the harmonic content of the resulting waveform.

The output levels of the operators also play an important part in determining the final harmonic spectrum of the sound. These levels specify the amplitude of the signal from each operator. The output level of a carrier determines the overall volume of the sound. The output level of a modulator determines the number of partials evident in the harmonic spectrum. The modulator output level acts like a variable bandpass filter centered at the carrier frequency.

Modulation Index

The modulator output level is related to another FM parameter called the modulation index. This parameter is not used directly in DX7 II terminology, but it is very important to understanding the nature of FM spectra. The modulation index is determined directly by the modulator output level. This relationship is illustrated in the following graph.



For example, the modulation index resulting from a modulator output level of 85 is 4. This number helps to determine the number of partials audibly present in a harmonic spectrum.

Bandwidth

Bandwidth is a term used to describe the "frequency space" occupied by a harmonic spectrum. A spectrum with components in a small frequency range is said to have a narrow bandwidth. A spectrum with components in a large frequency range is said to have a wide bandwidth. This term is useful in describing the effect of operator frequencies and output levels on the harmonic spectrum of a sound.

Symbols

Used in conjunction, the frequency ratios and modulation index determine the final waveform. The general effect of these parameters is described below. These descriptions use certain symbols to represent the frequencies and output levels.

c = carrier frequency
m = modulator frequency
I = modulation index as defined by the modulator output level

Frequency ratios are specified in the form c:m = 1:2. This example indicates that the ratio of the carrier frequency to the modulator frequency is 1 to 2. In other words, the carrier frequency is half the modulator frequency.

General Effects

Although the final waveform is influenced by the configuration of carriers and modulators, some general effects of the carrier frequency, modulator frequency, and modulator output level can be described.

1. By increasing the modulator frequency with respect to the carrier frequency, the bandwidth widens as the components in the harmonic spectrum "spread out." That is, they become more and more separated in frequency from each other. The overall number of partials in the spectrum does not change.
2. By increasing the carrier frequency with respect to the modulator frequency, the entire harmonic spectrum shifts upward. The number of partials and the relationships between them remains largely unchanged.
3. By increasing the modulator output level, the bandwidth widens as the number of components in the harmonic spectrum increases. This is like opening a bandpass filter, allowing more and more components to pass through.

Integer Ratios

The frequencies of a carrier and its modulator can be in any ratio. Of particular interest are the ratios of integers, or whole numbers. When the frequencies of a carrier and its modulator are in a ratio of integers, the components in the harmonic spectrum generally form what is called the harmonic series. This special spectrum consists of partials with frequencies which are integral (whole number) multiples of the fundamental, or lowest partial. This type of spectrum is characteristic of most wind and string instruments, and therefore is useful when such instruments are to be simulated by a voice on the DX7 II.

Frequency in the DX7 II

As you may know, the frequency of each operator in a voice can be specified in one of two ways: as a ratio or a fixed frequency. In ratio mode, the frequency of an operator is specified as being in a ratio with the pitch of the key being played. For example, if an operator's ratio frequency is set to 1.00 and the C3 key is played, the absolute frequency produced by that operator will be that of C3. If the operator's ratio frequency is set to 2.00, the absolute frequency will be that of C4, one octave higher than the key being played. Fixed frequencies are specified in cycles per second, or Hertz (Hz). Operators with a fixed frequency produce the same pitch regardless of the key played.

Basic Configurations

The basic operator configurations of which the algorithms consist are described in the previous section of this booklet. In addition to the frequency ratios and modulation indices, the final harmonic spectrum also depends on the configuration of carriers and modulators. The specific spectral characteristics of the basic configurations are described below.

Simple Pairs

Simple pairs are the most basic configurations in FM synthesis. Understanding the spectra resulting from simple pairs is a major step towards understanding FM synthesis. While a rigorous treatment of the required mathematics is beyond the scope of this booklet, the general concept can be presented.

As you know, side band frequencies appear whenever the frequency of one operator is modulated by an audio range signal from another operator. These side band frequencies form the components of a harmonic spectrum. This spectrum also includes the carrier frequency itself. The specific frequencies of these components can be calculated in the following manner (remember that c = carrier frequency and m = modulator frequency).

Carrier Frequency:	c
1st Order Side Band Frequency:	c + m c - m
2nd Order Side Band Frequency:	c + 2m c - 2m
3rd Order Side Band Frequency:	c + 3m c - 3m

etc.

Notice that the "order" of the side band frequencies indicates the number by which the modulator frequency is multiplied before being added to or subtracted from the carrier frequency.

This process of calculation could continue ad infinitum. However, it would be pointless to do so, because the amplitude of the harmonic components become inaudible after a certain point. At what order of side band frequencies should the calculations stop? The modulation index provides the answer to this question. As you'll recall, the modulation index acts like a variable bandpass filter. The higher the index, the more harmonic components are audible. The calculations should stop after the side band frequency order number reaches I + 2. Side band frequencies of higher order will be inaudible.

Example

For example, suppose that the carrier frequency c is specified as 100 Hz and the modulator frequency m is 200 Hz. The ratio of carrier frequency to modulator frequency would be 100:200 or 1:2. Further suppose that the modulation index is 2 (which results from a modulator output level of about 76). The frequencies of the harmonic components resulting from this ratio are calculated in the following manner.

Carrier Frequency:	100
1st Order Side Band Frequency:	100 + 200 = 300 100 - 200 = -100
2nd Order Side Band Frequency:	100 + 400 = 500 100 - 400 = -300
3rd Order Side Band Frequency:	100 + 600 = 700 100 - 600 = -500
4th Order Side Band Frequency:	100 + 800 = 900 100 - 800 = -700

The side band frequencies beyond this point will be inaudible (I + 2 = 2 + 2 = 4). The negative frequencies are merely sine waves which are out of phase with the positive frequencies. Since our ears are not very sensitive to phase relationships, you can simply consider these to be positive frequencies as well.

Note:

The amplitudes of the side band frequencies are determined using the modulation index in conjunction with mathematical constructs called Bessel functions. These functions even influence the amplitude of the carrier apart from its output level setting. For this example and the rest of the booklet, it is only important to know that the side bands with similar (but oppositely signed) frequencies may add together to produce a harmonic component of greater amplitude. On the other hand, they may tend to cancel each other out and produce a harmonic component of lesser amplitude. This may also affect the carrier amplitude.

This example illustrates that a simple pair in the ratio c:m = 1:2 and a modulation index of 2 produces a total of five different audible harmonic components. These components are all odd integer multiples of the carrier frequency, which is acting as the fundamental of the harmonic spectrum in this case. This configuration, like any simple pair in an integer ratio, produces the special spectrum known as the harmonic series. In this case, the series consists only of odd harmonics.

The technique of calculation used above can be applied to operator frequencies specified as ratios in the DX7 II. The results of such calculations will be simply the ratios formed by the harmonic components and the base pitch of the key being played on the keyboard. Negative ratios are merely out of phase with their positive counterparts.

Basic Waveforms

The basic waveforms used in older analog synthesizers can be recreated on the DX7 II using a simple pair. In the following table, the carrier frequency is 1.00 (specified in ratio mode) with an output level of 99.

Waveform	Modulator Output Level	Modulator Frequency Ratio	Feedback Level
Sine	≤40	Any	0
Triangle	40 to 70	1.00	0
Sawtooth	≥70	1.00	Any
Square	≥40	2.00 3.00 5.00 etc.	0
Noise	≥93	Any	7

This corresponds well with the example of calculation above. With a c:m ratio of 1:2 and a modulation index of 2, a square wave (which consists only of odd harmonics) is produced.

Parallel Carriers

As you'll recall, parallel carriers are formed when two or more carriers are modulated by one modulator. Each carrier behaves exactly as a simple pair with the modulator. The calculations are performed in the same manner. One of the only differences between this configuration and that of independent simple pairs is that, in this case, the modulation index for each pair is the same. Another difference is that the c:m ratio can only be changed by the carrier frequency. This configuration can be very helpful in economical programming if a single modulation index is sufficient for the sound component being created.

Parallel Modulators

The configuration of parallel modulators introduces a new wrinkle into FM programming. Since there are two or more independent operators modulating one carrier, there are multiple indices which bring about a much greater number of harmonic components than any configuration considered so far. The signal of the carrier is being modulated by a complex (that is, non-sine) waveform, which results by combining the independent sine waves of the modulators. This also contributes to the increased number of harmonic components.

The entire harmonic spectrum of a parallel modulator configuration consists of several parts. As with parallel carriers, the simple side band frequencies arising from the pairs formed by each modulator and the single carrier are present. In addition, harmonic components known as combination side band frequencies appear as a result of the complex modulating waveform.

An interesting phenomenon occurs with a configuration of two parallel modulators when the frequency of one of the modulators is very low. The simple side band frequencies of the pair including the low frequency modulator appear close to either side of the side band frequencies resulting from the high frequency modulator. The high frequency modulator's side bands act as local carriers for the side bands of the low frequency modulator. As the frequencies of the modulators become closer, this relationship becomes less clear as the harmonic components overlap and increase in amplitude.

Stacked Modulators

The harmonic components arising from stacked modulators are essentially similar to those resulting from parallel modulators. The reason is that, like parallel modulators, the waveform being used to modulate the carrier is complex. This is due to the fact that the modulator immediately above the carrier is itself being modulated. In the case of two stacked modulators where the upper modulator is at a low frequency, side band frequencies appear around each of the harmonic components arising from the high frequency modulator except the carrier itself.

Note:

> ***If a modulator in a stack has an output level of 0, there will be no effect from any modulator above it in the stack.***

Feedback

Certain FM spectra can be created by feeding the signal from a single operator back to its own modulation input. This technique is called feedback, and is used in a wide variety of applications. Each of the algorithms includes a feedback path, which indicates the feedback signal flow. In all but one of the algorithms, a modulator provides the feedback signal. This signal is usually directed to the modulator's own input. In algorithms 4 and 6, the feedback signal is directed to the input of a modulator higher in the stack. Algorithm 32 is the only one in which a carrier signal is fed back into its own input.

One of the characteristics of single operator feedback is that the frequency ratio is always 1:1. This is because the frequency of the modulating signal is always the same as the signal frequency of the modulated operator (these signals being in fact the same signal). The feedback level (from 0 to 7) provides the equivalent of a modulation index for the feedback system. By analyzing a single operator feedback system with the calculations described for simple pairs, you can see that a sawtooth wave with odd and even components in the harmonic series is generated.

The applications of feedback range from the generation of sawtooth-like spectra (useful as the basis of string sounds) to unpitched noise (used, for example, to simulate the sound of a player's breath at the beginning of a flute note). This noise is particularly easy to produce if the feedback operator is in a non-integral ratio with another operator in its configuration. The reason for this rich harmonic content is the fact that the modulating waveform generated by a feedback operator is itself complex. A feedback level of 7 produces 23 audible components, which act as effective parallel modulators. The final waveform and spectrum of such a configuration is thus extremely complex.

Sample Rate & Aliasing

The DX7 II operators are digital oscillators that produce a stream of numbers. This stream is converted into an audio signal by a digital-to-analog convertor. The rate at which these numbers are generated is called the sample rate. The DX7 II generates numbers about 60,000 times per second. This represents a sample rate of 60 kHz. The sample rate is one of the most fundamental aspects of any digital audio system.

It is well known that higher frequencies are less accurately represented by digital audio systems than lower frequencies. A basic concept known as the Nyquist Theorem states that no digital audio system can reproduce frequencies greater than half the sample rate. With a sample rate of 60 kHz, the DX7 II can reproduce frequencies up to 30 kHz. If frequencies above this limit are present, a phenomenon known as aliasing occurs. This normally introduces unwanted noise into a signal.

Aliasing occurs in the following way. Suppose that, by virtue of its programming, a voice tries to produce a harmonic spectrum with a component at 31 kHz. This is quite possible using parallel or stacked modulators and feedback. The component at 31 kHz is "reflected" around the 30 kHz Nyquist frequency and appears at 29 kHz. A component at 40 kHz is reflected and appears at 20 kHz. These reflections can interfere with the harmonic components that are supposed to be at or near the reflection frequencies.

It is important to be aware of aliasing in order to avoid it in your programming. It is usually a problem only as higher keys are played. Aliasing is eliminated by the use of level scaling to reduce the modulator output level as higher keys are played. Lowering the modulation index in this way decreases the number of high frequency components in the higher notes. Level scaling is described in Section 5.

Fixed Frequencies

Operator frequencies are usually specified in the ratio mode. This causes the actual frequencies generated by the operators to vary according to the keys being played. Specifying a frequency in the fixed mode causes that operator to generate a sine wave at one frequency regardless of the key played. Although they may not be as intuitively clear, there are many uses for fixed frequencies.

Carriers

Specifying a fixed frequency for a carrier can be used to establish a formant in the sound. Formants are regions of resonance in a harmonic spectrum which remain relatively fixed with respect to the notes being played. The human voice exhibits several formants. Many other natural sounds also exhibit formants.

Another application of fixed frequencies is to create low frequency carriers. By specifying a fixed frequency of 1 or 2 Hz for a carrier and a ratio frequency value for its modulator, pairs of harmonic components will appear very close together in frequency. These pairs will be separated by only 2 to 4 Hz, and therefore will exhibit beats. This can be used to create a rotating speaker or chorusing effect.

Modulators

Fixed frequencies are used with modulators in a variety of applications as well. At very low frequencies, modulators can provide additional low frequency oscillators (LFOs), which are used to create vibrato by slowly modulating the frequency of a carrier. Low fixed frequency modulators found in the middle of a stack can be used to simulate reverberation.

In the middle range, fixed modulator frequencies result in a ring modulator effect. This produces a clangorous timbre useful for bell sounds. In this case, the timbre changes in different keyboard ranges due to the changing ratio between the carrier and the fixed frequency modulator. As different keys are played, the frequency of the carrier changes while the frequency of the modulator does not.

Frequencies & Levels for Example Voice

The algorithm chosen for the harpsichord voice is algorithm 3. Its two three-operator stacks are used to simulate the sound of two sets of strings one octave apart. As noted in the previous section, the sound of a harpsichord is somewhat thin and bright. It's spectrum consists of components in the harmonic series.

You may recall that a simple pair in the ratio c:m = 1:2 results in a spectrum containing only odd harmonics. This is a bit too thin for a harpsichord sound. It turns out that a ratio of c:m = 1:3 generates a spectrum in which every third harmonic is missing. This is better suited to the voice you are creating by following the procedures at the end of each section.

The lower harmonics in each set of strings are generated by the pair formed by the carrier and the lower modulator. The higher harmonics (responsible for the bright quality) are generated by the higher modulator in the stack. In order to maintain the spectral density of every third harmonic missing, the three operators in the stack reproducing the low set of strings should be in the ratios $c:m_1:m_2 = 1:3:9$. Notice that the ratio between the modulators is equivalent to that of the lower pair (3:9 = 1:3). The stack responsible for the high set of strings should double these values, i.e. $c:m_1:m_2 = 2:6:18$, to produce the same spectrum one octave higher.

Output levels are generally arrived at after some experimentation. This is particularly true of modulator levels. Carrier levels are often set at their maximum value of 99 to provide the greatest possible dynamic range for the voice. In general, carrier levels are used to balance the different portions of the overall sound. Modulator levels can have a significant influence on the sound using lower output levels. In fact, modulators high in a stack need less output than do those located lower in the stack to achieve a given degree of impact on the sound.

In a harpsichord, both sets of strings are equally loud and include a similar number of harmonics. The output levels for corresponding operators in the two stacks can therefore be identical.

Procedure

1. Press the EDIT button if you are not already in Edit Mode.
2. Press the OSCILLATOR button (#8).
3. Operator 1 has the desired frequency value of 1.00. The left stack will reproduce the sound of the low set of strings.
4. Press the OPERATOR SELECT button #2 (#2).
5. Position the cursor at the Coarse parameter.
6. Use the +1 button to specify a value of 3.00.
7. Press the OPERATOR SELECT button #3 (#3).
8. Use the data entry slider or the +1 button to specify a Coarse value of 9.00.
9. Press the OPERATOR SELECT button #4 (#4).
10. Use the +1 button to specify a value of 2.00. The right stack will reproduce the sound of the set of strings one octave higher than the low set.
11. Press the OPERATOR SELECT button #5 (#5).
12. Use the data entry slider or the +1 button to specify a Coarse value of 6.00.

13. Press the OPERATOR SELECT button #6 (#6).
14. Use the data entry slider or the +1 button to specify a Coarse value of 18.00.
15. Press the OUTPUT LEVEL button (#10) repeatedly until the Outlvl display appears.
16. Press the OPERATOR SELECT button #1 (#1). Notice that operator 1 is set to a level of 99. This is the desired setting for this carrier.
17. Press the OPERATOR SELECT button #2 (#2).
18. Position the cursor at the Level parameter.
19. Specify an output level of 84.
20. Press the OPERATOR SELECT button #3 (#3).
21. Specify an output level of 80. This modulator needs less output than operator 2 to be as effective.
22. Repeat the levels of operators 1, 2, and 3 for operators 4, 5, and 6 respectively. In terms of bandwidth, this stack behaves exactly as the first one.
23. Press the ALGORITHM button (#7)
24. Position the cursor at the Fbl parameter and use the +1 button or the data entry slider to specify a feedback level of 6. This will brighten the sound a bit and add a metallic quality reminiscent of metal harpsichord strings.

The timbre should now roughly approximate that of an acoustic harpsichord. However, the envelope is more like that of an organ. The procedure for creating an appropriate envelope for this sound is found at the end of the next section.

4 Envelopes

Each operator includes an envelope generator (EG), which is used to specify how the operator's output level will vary over time from the moment a key is pressed until after it is released. This is used to create the envelope or "shape" of a sound.

Envelopes

For example, think of the difference between the sound of a note played by striking a piano key and by bowing on a violin string. The volume of the piano note peaks almost immediately and decays very slowly until the key is released. The violin sound reaches its peak more slowly and retains a high level as the bow moves across the string.

All acoustic instruments have characteristic envelopes that shape the volume, timbre, and even the pitch of each note. The EG associated with each operator provides a means of simulating these envelopes closely in any DX7 II voice.

Note:

> *Applying an EG to the output level of a carrier affects the way in which a voice's volume changes during each note. Applying an EG to the output level of a modulator affects the way in which the bandwidth (number of harmonics) changes during each note. These two types of envelopes can remain independent of each other within any voice.*

Each operator's EG is specified by four rates and four levels. These parameters are explained in detail below. The diagram on the front panel of the DX7 II illustrates the output of a typical EG.

Rates & Levels

Each envelope consists of four levels, which affect the output level of the selected operator. These envelope levels are relative to the overall operator output level. The time it takes to reach an EG level during the course of a note is determined by the corresponding rate. Used in conjunction with EG levels, rates provide a completely flexible means of creating various volume and timbre envelopes. These can be used to simulate acoustic instrument envelopes or to invent new envelopes never before heard.

The envelope architecture employed by the DX7 II EGs provides much more flexibility than the ADSR envelopes found on older analog synthesizers. For example, the volume envelope of a flute cannot be simulated using an ADSR EG. However, it is a simple matter to simulate such an envelope with the EGs found on the DX7 II.

Copying EG Values

With nine parameters to adjust for each operator's EG, specifying envelopes for several operators can be quite a chore. In many cases, however, identical or nearly identical envelopes can be used for most or all of the operators in an algorithm. This is because there is often a correlation between the volume and timbre envelopes in natural voices. Once one EG is set, you can copy its parameter values to any other operator using the EG COPY button.

After selecting one operator's EG for specification and copying, press and hold the EG COPY button. While holding the EG COPY button, select the desired operator by pressing the appropriate OPERATOR SELECT/EG COPY button (#1~6). Releasing the EG COPY button completes the operation.

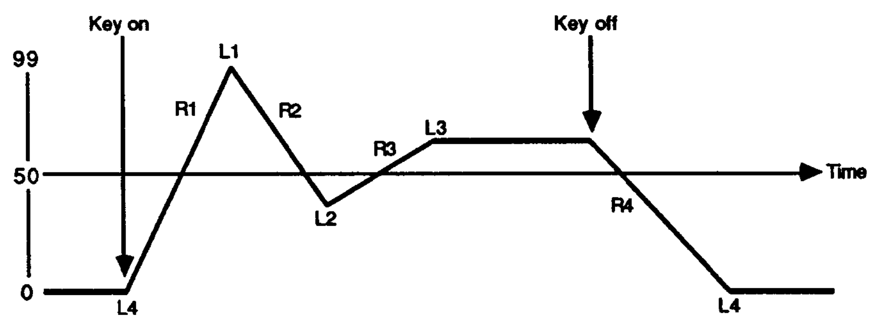
Modulator Envelopes

By providing automatic control over the bandwidth of a spectrum, modulator envelopes are one of the keys to good FM programming. Although the volume and timbre envelopes of natural sounds are often similar, the only points at which they must correspond are the moments at which a key is pressed and released. A common technique is to copy carrier envelopes to the modulators and adjust them to suit the sound.

For example, the volume of a sound might reach its full value almost instantly after a key is pressed. The timbre of the same voice might develope more slowly, requiring a longer attack rate (R1 in DX7 II EGs). This technique is used to simulate the volume and tonal attack characteristics of brass instruments.

Pitch EG

Many musical sounds vary in pitch as well as in volume or timbre during the course of a note. The DX7 II includes a separate envelope generator which is applied to frequency in addition to the individual operator level EGs. As with each operator's EG, the pitch EG is specified by four rates and four levels. The following diagram illustrates the output of the pitch EG.



Note:

The pitch EG affects only those operators set to the ratio frequency mode.

The levels of the pitch EG are very similar to those of the operator EGs. The main difference lies in the fact that the pitch EG levels refer to pitch rather than output level. A pitch level of 50 represents the standard pitch of the instrument.

EG Values for Example Voice

A harpsichord exhibits a very fast attack because it produces its sound by plucking a string inside the instrument. The sound then dies away slowly as the key is held down. Once the key is released, the sound disappears quickly. Before it dies completely, you can hear a very short buzzing sound as the plectrum touches the string on the way back to its starting position. This damps the vibration of the string and causes the buzzing sound for a moment before the string becomes still.

The EGs for all operators include rather fast attack rates (R1). The carrier in the stack reproducing the upper set of strings uses a slightly faster attack rate than that of the other stack. This is generally true of higher strings plucked at the same time and with the same force as lower strings. The attack levels (L1) are all set at their maximum values to assure maximum effect. The release rates while the key is held down (R2 & R3) are slow. The first release levels (L2) are set fairly high to maintain the integrity of the sound during its first portion. The second release levels (L3) are set to 0 so that the sound will die away if a key is held for a long time. This simulates the behavior of an acoustic harpsichord.

It is with the key-off release rates and levels (R4 & L4) that things get quite interesting. The volume of a harpsichord sound drops off sharply after the key is released, but not before the plectrum touches the string, damping the vibration and creating the buzzing sound. R4 of the carriers must be relatively fast, but not so fast that the buzzing sound cannot be heard. This buzzing sound is reproduced by suddenly increasing the modulation index after the key has been released. This is accomplished by setting a high L4 value for the modulators to increase the number of harmonics in the sound briefly as the sound disappears. The modulators' R4 must be set faster than the carriers' R4 so that the buzzing sound will be heard. The carriers' L4 is set to 0 in order to achieve silence as the note finally dies.

The EG values presented here were arrived at after consideration of the points above and a certain amount of experimentation. As with any of the parameter values in this booklet, you should feel free to alter them to your own liking.

Procedure

1. Press the EDIT button if you are not already in Edit Mode.
2. Press the EG button (#9).
3. Press the OPERATOR SELECT button #1 (#1).
4. Position the cursor at the R1 parameter and use the -1 button or the data entry slider to specify a value of 90.
5. Position the cursor at the R2 parameter and use the -1 button or the data entry slider to specify a value of 36.
6. Position the cursor at the R3 parameter and use the -1 button or the data entry slider to specify a value of 20.

7. Position the cursor at the R4 parameter and use the -1 button or the data entry slider to specify a value of 48.
8. Position the cursor at the L1 parameter. Its initialized value is 99. Do not adjust this value.
9. Position the cursor at the L2 parameter and use the -1 button or the data entry slider to specify a value of 80.
10. Position the cursor at the L3 parameter and use the -1 button or the data entry slider to specify a value of 0.
11. Position the cursor at the L4 parameter and use the -1 button or the data entry slider to specify a value of 0.

```
OP1   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 90 36 20 48 99 80  0  0
```

12. Press the OPERATOR SELECT button #2 (#2).
13. Repeat steps 4 through 11 using the following values.

```
OP2   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 99 55 25 70 99 90  0 99
```

14. Press the OPERATOR SELECT button #3 (#3).
15. Repeat steps 4 through 11 using the following values.

```
OP3   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 85 36 30 90 99 85  0 99
```

16. Press the OPERATOR SELECT button #1 (#1).
17. Press and hold the EG COPY/STORE button. The message "EG & Scaling Copy OP1 to OP?" will appear.
18. Continue to hold the EG COPY/STORE button. Press the OPERATOR SELECT button #4 (#4). The EG specified for operator 1 will be copied to operator 4.

19. Press the OPERATOR SELECT button #2 (#2).
20. Press and hold the EG COPY/STORE button. The message "EG & Scaling Copy OP1 to OP?" will appear.
21. Continue to hold the EG COPY/STORE button. Press the OPERATOR SELECT button #5 (#5). The EG specified for operator 2 will be copied to operator 5.
22. Press the OPERATOR SELECT button #3 (#3).
23. Press and hold the EG COPY/STORE button. The message "EG & Scaling Copy OP1 to OP?" will appear.
24. Continue to hold the EG COPY/STORE button. Press the OPERATOR SELECT button #6 (#6). The EG specified for operator 3 will be copied to operator 6.
25. Make the following modifications to the EG rates of operators 4 and 5 by pressing the appropriate OPERATOR SELECT button and positioning the cursor at the rate parameters. The levels are used without modification.

```
OP4   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 95 49 32 50 99 80  0  0
```

```
OP5   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 86 30 25 70 99 90  0 99
```

```
OP6   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   0 85 36 30 90 99 85  0 99
```

The voice should now be approaching a reasonable simulation of a harpsichord. However, as you play keys in the upper range of the keyboard, notice the distortion that begins to appear. This distortion is caused by aliasing. It can be eliminated by the use of level scaling, which is described in the next section.

5

Keyboard Scaling

Many musical sounds change as notes in different pitch ranges are played. Some become less bright in the high range. Others lengthen their envelopes as lower notes are played. These effects can be recreated in DX7 II voices using keyboard scaling. Keyboard scaling can be applied to each operator independently, providing very flexible control over the sound in different pitch ranges.

Level Scaling

Normal level scaling is used to increase or decrease the output level of any operator as ascending or descending keys are played. The keyboard is divided into two sections which meet at one key called the break point. This key is independently specified for each operator. Each section of the keyboard is assigned a scaling curve which determines how the levels will change as keys are played in the direction away from the break point. The four curves are listed and defined below.

–lin Operator output will decrease linearly with increasing distance from the break point.

–exp Operator output will decrease exponentially with increasing distance from the break point.

+exp Operator output will increases exponentially with increasing distance from the break point.

+lin Operator output will increases linearly with increasing distance from the break point.

The degree to which an operator responds to level scaling is determined by the scaling depth parameters. The right depth parameter specifies the effect of scaling on the operator's output level to the right of the break point. The left depth performs the same function to the left of the break point. A diagram illustrating the scaling curves and depths can be found on the front panel of the DX7 II.

Applying level scaling to a carrier causes the volume of the sound to increase or decrease as keys further from the break point are played. Modulator level scaling affects the bandwidth of the sound. One important application of modulator level scaling is to eliminate aliasing. If no modulator level scaling is included in a voice, the bandwidth can remain relatively constant. If the bandwidth is wide and high notes are played, aliasing is almost sure to become apparent as the high spectral components are reflected about the Nyquist frequency (half the sample rate). Scaling the modulator output level with a negative curve to the right of the break point decreases the modulation index in the high range of the keyboard. This in turn decreases the bandwidth in the upper range, reducing the number of high components and thus eliminating aliasing.

Fractional Scaling

The DX7 II includes another type of level scaling known as fractional scaling. This scaling mode provides extremely accurate and completely flexible control of an operator's output level based on the keys played. The keyboard is divided into groups of three keys each. The level scaling factor for each group is specified independently. This allows you to create any scaling curve you wish across the entire keyboard. In addition, each group's scaling factor can be specified over a range of 0 to 255. This provides greater scaling resolution than that available in the normal scaling mode. For further information on fractional scaling, consult the Supplemental Booklets "Understanding Fractional Scalings" and "Programming Fractional Scalings."

Rate Scaling

Rate scaling determines the extent to which EG rates will speed up as higher notes are played on the keyboard. Rate scaling is used to simulate the change in the envelope of higher pitched notes evident in many acoustic instruments. For example, if you play the lowest and highest notes on a piano with equal force and hold the keys down, you will notice that the highest note dies away long before the lowest note. As higher notes are played, the EG rates become faster. Rate scaling allows you to duplicate this effect in your own DX7 II voices. This aids in creating more natural sounding voices.

Scaling for Example Voice

Normal level scaling works well for the harpsichord voice. Scaling is applied to the carriers in order to achieve a good volume balance across the keyboard. The left curves are +lin and the right curves are –lin. The right and left depths are small because only a bit of scaling is needed in this case.

The upper modulator output levels are scaled in order to eliminate the aliasing you heard after specifying the operator frequencies and levels at the end of the last section. These are the modulators responsible for the highest harmonics. This requires a –lin right curve and a high right depth. The left curve is +lin to increase the brightness in the lower range. The left depth is small because the increase in brightness should be small.

Rate scaling is applied primarily to the operators in the second stack. This enhances the simulation of higher strings, which have similar but shorter envelopes than their lower counterparts.

Procedure

1. Press the EDIT button if you are not already in Edit Mode.
2. Press the OUTPUT LEVEL button (#10) repeatedly until the Outlvl display appears.
3. Press the OPERATOR SELECT button #1 (#1).
4. Position the cursor at the Ld (left depth) parameter and use the +1 button or the data entry slider to specify a value of 20.
5. Position the cursor at the Lc (left curve) parameter and use the +1 button or the data entry slider to specify the +lin curve.
6. Position the cursor at the Bp (break point) parameter and use the -1 button or the data entry slider to specify a break point at C2.
7. Position the cursor at the Rd (right depth) parameter and use the +1 button or the data entry slider to specify a value of 6.

```
OP1    Outlvl >Level>Ld >Lc  >Bp  >Rc >Rd
alg 3 111111     99 20 +lin  C2  -lin  6
```

8. Press the OPERATOR SELECT button #4 (#4).
9. Repeat steps 4 through 7.

```
OP4    Outlvl >Level>Ld >Lc  >Bp  >Rc >Rd
alg 3 111111     99 20 +lin  C2  -lin  6
```

10. Press the OPERATOR SELECT button #3 (#3).
11. Repeat steps 4 through 7 with the following values.

```
OP3    Outlvl >Level>Ld >Lc  >Bp  >Rc >Rd
alg 3 111111     80 10 +lin  G3 -lin 60
```

12. Press the OPERATOR SELECT button #6 (#6).
13. Repeat step 11.

```
OP6    Outlvl >Level>Ld >Lc  >Bp  >Rc >Rd
alg 3 111111     80 10 +lin  G3 -lin 60
```

Now that the level scaling has been set, rate scaling is next.

14. Press the EG button (#9).
15. Press the OPERATOR SELECT button #1 (#1).
16. Position the cursor at the Rs (rate scaling) parameter and use the +1 button to specify a value of 3. Even the low set of strings shorten their envelopes as higher notes are played.

```
OP1   EG      >Rs>R1>R2>R3>R4>L1>L2>L3>L4
alg 3 111111   3 90 36 20 48 99 80  0  0
```

17. Press the OPERATOR SELECT button #2 (#2).
18. Use the +1 button to specify a value of 1.
19. Press the OPERATOR SELECT button #3 (#3).
20. Use the +1 button to specify a value of 1.
21. Press the OPERATOR SELECT button #4 (#4).
22. Use the +1 button to specify a value of 6.
23. Press the OPERATOR SELECT button #5 (#5).
24. Use the +1 button to specify a value of 4.
25. Press the OPERATOR SELECT button #6 (#6).
26. Use the +1 button to specify a value of 6.

The voice should now sound quite like a harpsichord. Notice that there is no aliasing as you play the higher keys. In the next section you will complete the voice by adding a few finishing touches.

6

Effects

Effects provide the finishing touches for a voice. You can control many of the effects in real time. This allows you to include a strong element of expression in your playing. Other effects are under programmed control. This section presents the effects available on the DX7 II, and incorporates some of them in finishing the harpsichord voice.

Vibrato/Tremolo

Vibrato and tremolo effects are achieved using a low frequency oscillator (LFO). The DX7 II LFOs generate a slow sine, triangle, sawtooth, square, or random waveform at a frequency usually well below the range of human hearing. This waveform is used to modulate the frequency or output level of the operators in a voice. For example, applying LFO to a carrier's frequency produces vibrato. Modulating operator output levels with the LFO produces tremolo (for carriers) or a "wah-wah" effect (for modulators).

The LFO parameters are accessed by pressing the LFO button (#12). They include speed, which regulates the rate of the LFO, and delay time, which delays the onset of the LFO for a specified time after a key is pressed. The DX7 II has a total of 16 LFOs, which can be applied individually to each note as it is played. The pitch modulation depth and amplitude modulation depth determine the potential strength of the LFO effect on the operator's frequency and output level. These parameters act like LFO output level controls.

Real Time Control

Real time controllers such as the modulation wheel and aftertouch can be used to vary the intensity of the programmed LFO effect as you play. In addition, they can be used to affect the frequency and output level of the operators directly. This allows you to bring vibrato into your voice or brighten the sound by increasing the bandwidth at will. The available real time controllers are described below. The parameters for each one are displayed by pressing the button numbered in parentheses.

Pitch Bend (#24)

The pitch bend wheel does exactly as its name implies. The range over which the pitch is bent by the wheel is programmable. It can also be programmed to bend the pitch in discrete steps instead of smoothly.

Mod Wheel (#25)

You can program the modulation wheel to invoke LFO and provide output level control. It cannot be programmed to bend the pitch. This is accomplished with the neighboring pitch bend wheel. Like most of the real time controllers, the mod wheel parameters determine the degree to which the effect will be evident as the wheel is engaged.

Aftertouch (#25)

Aftertouch is activated by pressing down on the keys after a note has been played. This can be used to affect the frequency and output level of the operators in a voice. The harder a key is pressed down, the greater the effect.

Breath Controller (#25)

A breath controller is plugged into the jack located next to the headphone jack at the front of the DX7 II. It is a device into which air is blown in order to achieve wind-like articulations as the keys are played. Like aftertouch, a breath controller can be used to control operator frequencies and levels.

Foot Controllers (#26)

Foot controllers are used to regulate various effects in a manner similar to the breath controller, aftertouch, and modulation wheel. Two continuous foot controllers can be connected to the DX7 II rear panel foot controller jacks 1 and 2. The foot control parameters have no effect unless FC-7 foot controllers are connected to the corresponding jacks on the rear panel of the DX7 II.

Continuous Sliders (#27)

The continuous sliders, labeled CS1 and CS2, are found at the left of the front panel on the DX7 II. A total of 105 different parameters are available to apply with each of the continuous sliders. This lets you use these sliders as very fine and flexible performance controls.

The following parameters are available for selection in each of the CS screens. In the edit mode, CS2 becomes the data entry slider which is used to select the parameter to be affected by either of the continuous sliders back in the play mode. Using a slider in the edit mode to select the effect it will have in the play mode may be a bit confusing at first, but you'll soon get the hang of it.

The available parameters are listed from the top to the bottom position of the data entry slider as it is used to select them.

Operator 6~1 Output level
Operator 6~1 AMS
Operator 6~1 Key velocity
Operator 6~1 EG level 4~1 (L4~L1)
Operator 6~1 EG Rate 4~1 (R4~R1)
Operator 6~1 Osc. detune
Operator 6~1 Frequency fine tune
Operator 6~1 Frequency coarse tune
Portamento time
Pitch EG level 4~1
Pitch EG rate 4~1
LFO AMD
LFO PMD
LFO PMS
LFO Delay
LFO Speed
LFO Waveform
Feedback level
Algorithm
Dual detune
Pan select
Output balance (A/B)
Total volume
No effect

In the dual or split voice modes, the selected parameters can be controlled for either or both of the A and B voices. In single mode, the A parameter must be on for CS control to be effective. Notice that in selecting Output balance (A/B), only the A parameter is turned on and off.

Portamento

Portamento is an effect found on even the earliest synthesizers. It allows you to "glide" smoothly from one note to another. For example, a piano cannot be played with portamento. A violin, trombone, or even a human voice can glide from note to note very easily. The DX7 II provides very flexible portamento parameters accessed by pressing the PITCH BEND/PORTAMENTO button (#24). These parameters include the time it takes to glide between notes, and discrete steps.

Detune

The detune parameter is specified in the OSCILLATOR button (#8) display. It provides a "super fine" frequency control for each operator. It is used to shift the frequency of an operator by as much as 2 cents up or down. This parameter is independent of the coarse and fine parameters.

Detuning has many uses, particularly when simulating acoustic instruments. By nature, such instruments are never in mathematically perfect tune anyway. Small detune values induce beats between the harmonic components, while large values create a chorus effect useful for large ensemble voices such as brass or strings.

Sensitivity

The sensitivity parameters are displayed by pressing the SENSITIVITY button (#11). They provide separate controls over the intensity of effect the LFO and key velocity have on each operator. Following the analogy drawn above, these parameters act as input attenuation controls for each individual operator.

Panning

The pan functions are among the most impressive effects in the DX7 II. You may be familiar with the pan controls on audio mixers. These controls allow sound to be shifted anywhere between the extreme left and extreme right of the stereo field. The pan parameters of the DX7 II control the placement of the voice(s) in the audio output stereo image.

There are three basic pan functions.

1. Independent output of voices A and B from the two audio jacks in dual and split voice modes.
2. A panning function that allows you to move the stereo position of an A+B voice mix (or a single voice) between the two audio outputs in any voice mode.
3. A level control for each of the voice A and B volume levels in dual and split voice modes.

While all three functions cannot be used simultaneously, functions 1 and 3 always work in conjunction with each other. Functions 2 and 3 can be regulated by LFO, key velocity, key position, and EG.

The pan parameters are accessed by pressing the PAN button (#30). They are effective only with stereo amplification or using headphones connected directly to the DX7 II. Also, the small LED above the right cursor button must be on. The right cursor button acts as the pan on/off button only in play mode. In edit mode, it must be available to position the cursor.

Note:

> *If the pan function has been turned off (indicated by no light above the right cursor button), the two audio outputs and both channels of the headphone output will deliver the same sound (a mix of voices A and B). No panning effects are possible in this case.*

The panning effects can be controlled by the LFO, key velocity, or key pitch. In addition, a separate pan EG is available to control the panning effects.

Effects for Example Voice

A harpsichord sound has very few of the effects described in this section. They respond slightly to key velocity, although the mechanism limits this sensitivity. To simulate this, the velocity sensitivity is adjusted primarily for the carriers. By detuning the operators, you can simulate the inevitable inaccuracies in any acoustic instrument. This is also outlined in the following procedure.

Procedure

1. Press the EDIT button if you are not already in Edit Mode.
2. Press the SENSITIVITY button (#11).
3. Press the OPERATOR SELECT button #1 (#1).
4. Position the cursor at the Velocity parameter and use the +1 button to specify a value of 4.
5. Press the OPERATOR SELECT button #2 (#2).
6. Use the +1 button to specify a value of 1.
7. Press the OPERATOR SELECT button #3 (#3).
8. Use the +1 button to specify a value of 1.
9. Press the OPERATOR SELECT button #4 (#4).
10. Use the +1 button to specify a value of 3.

11. Press the OPERATOR SELECT button #5 (#5).
12. Use the +1 button to specify a value of 1.

Leave the velocity sensitivity of operator 6 at 0. Now, on to detuning.

13. Press the OSCILLATOR button (#8).
14. Press the OPERATOR SELECT button #1 (#1).
15. Position the cursor at the Detune parameter and use the -1 button to specify a value of −1.
16. Press the OPERATOR SELECT button #2 (#2).
17. Use the +1 button to specify a value of +1.
18. Press the OPERATOR SELECT button #4 (#4).
19. Use the +1 button to specify a value of +3.
20. Press the OPERATOR SELECT button #5 (#5).
21. Use the +1 button to specify a value of +1.
22. Press the OPERATOR SELECT button #6 (#6).
23. Use the -1 button to specify a value of −1.

You are encouraged to continue developing this sound. It provides a reasonable harpsichord sound, but many other modifications are possible. Use your imagination and enjoy!

Notes