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6 Chapter 6 MIDI and Sound Synthesis

6.1 Concepts

6.1.1 The Beginnings of Sound Synthesis

Sound synthesis has an interesting history in both the analog and digital realms. Precursors to today's sound synthesizers include a colorful variety of instruments and devices that generated sound electrically rather than mechanically. One of the earliest examples was Thaddeus Cahill's Telharmonium (also called the Dynamophone), patented in 1897. The Telharmonium was a gigantic 200-ton contraption built of "dynamos" that were intended to broadcast musical frequencies over telephone lines. The dynamos, precursors of the tonewheels to be used later in the Hammond organ, were specially geared shafts and inductors that produced alternating currents of different audio frequencies controlled by velocity sensitive keyboards. Although the Telharmonium was mostly unworkable, generating too strong a signal for telephone lines, it opened people's minds to the possibilities of electrically-generated sound.

The 1920s through the 1950s saw the development of various electrical instruments, most notably the Theremin, the Ondes Martenot, and the Hammond organ. The Theremin, patented in 1928, consisted of two sensors allowing the player to control frequency and amplitude with hand gestures. The Martenot, invented in the same year, was similar to the Theremin in that it used vacuum tubes and produced continuous frequencies, even those lying between conventional note pitches. It could be played in one of two ways: either by sliding a metal ring worn on the right-hand index finger in front of the keyboard or by depressing keys on the six-octave keyboard, making it easier to master than the Theremin. The Hammond organ was invented in 1938 as an electric alternative to wind-driven pipe organs. Like the Telharmonium, it used tonewheels, in this case producing harmonic combinations of frequencies that could be mixed by sliding drawbars mounted on two keyboards.

As sound synthesis evolved, researchers broke even farther from tradition, experimenting with new kinds of sound apart from music. Sound synthesis in this context was a process of recording, creating, and compiling sounds in novel ways. The **musique concrète** movement of the 1940s, for example, was described by founder Pierre Schaeffer as "no longer dependent upon preconceived sound abstractions, but now using fragments of sound existing concretely as sound objects (Schaeffer 1952)." "Sound objects" were to be found not in conventional music but directly in nature and the environment – train engines rumbling, cookware rattling, birds singing, etc. Although it relied mostly on naturally occurring sounds, musique concrète could be considered part of the electronic music movement in the way in which the sound montages were constructed, by means of microphones, tape recorders, varying tape speeds, mechanical reverberation effects, filters, and the cutting and resplicing of tape. In contrast, the contemporaneous **elektronische musik** movement sought to synthesize sound primarily from electronically produced signals. The movement was defined in a series of lectures given in Darmstadt, Germany, by Werner Meyer-Eppler and Robert Beyer and entitled "The World of Sound of Electronic Music." Shortly thereafter, West German Radio opened a studio dedicated to research in electronic music, and the first elektronische music production, *Musica su Due Dimensioni*, appeared in 1952. This composition featured a live flute player, a taped portion manipulated by a technician, and artistic freedom for either one of them to manipulate the composition during the performance. Other innovative compositions followed, and the movement spread throughout Europe, the United States, and Japan.

There were two big problems in early sound synthesis systems. First, they required a great deal of space, consisting of a variety of microphones, signal generators, keyboards, tape recorders, amplifiers, filters, and mixers. Second, they were difficult to communicate with. Live performances might require instant reconnection of patch cables and a wide range of setting changes. "Composed" pieces entailed tedious recording, re-recording, cutting, and splicing of tape. These problems spurred the development of automated systems. The Electronic Music Synthesizer, developed at RCA in 1955, was a step in the direction of programmed music synthesis. Its second incarnation in 1959, the Mark II, used binary code punched into paper to represent pitch and timing changes. While it was still a large and complex system, it made advances in the way humans communicate with a synthesizer, overcoming the limitations of what can be controlled by hand in real-time.

Technological advances in the form of transistors and voltage controllers made it possible to reduce the size of synthesizers. Voltage controllers could be used to control the oscillation (i.e., frequency) and amplitude of a sound wave. Transistors replaced bulky vacuum tubes as a means of amplifying and switching electronic signals. Among the first to take advantage of the new technology in the building of analog synthesizers were Don Buchla and Robert Moog. The Buchla Music Box and the Moog Synthesizer, developed in the 1960s, both used voltage controllers and transistors. One main difference was that the Moog Synthesizer allowed standard keyboard input, while the Music Box used touch-sensitive metal pads housed in wooden boxes. Both, however, were analog devices, and as such, they were difficult to set up and operate. The much smaller MiniMoog, released in 1970, were more affordable and user-friendly, but the digital revolution in synthesizers was already under way.

When increasingly inexpensive microprocessors and integrated circuits became available in the 1970s, digital synthesizers began to appear. Where analog synthesizers were programmed by rearranging a tangle of patch cords, digital synthesizers could be adjusted with easy-to-use knobs, buttons, and dials. Synthesizers took the form of electronic keyboards like the one shown in Figure 6.1, with companies like Sequential Circuits, Electronics, Roland, Korg, Yamaha, and Kawai taking the lead in their development. They were certainly easier to play and program than their analog counterparts. A limitation to their use, however, was that the control surface was not standardized, and it was difficult to get multiple synthesizers to work together.

Figure 6.1 Prophet-5 Synthesizer

In parallel with the development of synthesizers, researchers were creating languages to describe the types of sounds and music they wished to synthesize. One of the earliest digital sound synthesis systems was developed by Max V. Mathews at Bell Labs. In its first version, created in 1957, Mathews' MUSIC I program could synthesize sounds with just basic control over frequency. By 1968, Mathews had developed a fairly complete sound synthesis language in MUSIC V. Other sound and music languages that were developed around the same time or shortly thereafter include CSound (created by Barry Vercoe, MIT, in the 1980s), Structured Audio Orchestras Language (SAOL, part of MPEG 4 standard), Music 10 (created by John

Chowning, Stanford, in 1966), cmusic (created by F. Richard Moore, University of California San Diego in the 1990s), and pcmusic (also created by F. Richard Moore).

In the early 1980s, led by Dave Smith from Sequential Circuits and Ikutaru Kakehashi from Roland, a group of the major synthesizer manufacturers decided that it was in their mutual interest to find a common language for their devices. Their collaboration resulted in the 1983 release of the MIDI 1.0 Detailed Specification. The original document defined only basic instructions, things like how to play notes and control volume. Later revisions added messages for greater control of synthesizers and branched out to messages controlling stage lighting. General MIDI (1991) attempted to standardize the association between program numbers and instruments synthesized. It also added new connection types (e.g., USB, FireWire, and wireless), and new platforms such as mobile phones and video games.

This short history lays the ground for the two main topics to be covered in this chapter: symbolic encoding of music and sound information – in particular, MIDI – and how this encoding is translated into sound by digital sound synthesis. We begin with a definition of MIDI and an explanation of how it differs from digital audio, after which we can take a closer look at how MIDI commands are interpreted via sound synthesis.

6.1.2 MIDI Components

MIDI (Musical Instrument Digital Interface) is a term that actually refers to a number of things:

- A symbolic language of event-based messages frequently used to represent music
- A standard interpretation of messages, including what instrument sounds and notes are intended upon playback (although the messages can be interpreted to mean other things, at the user's discretion)
- A type of physical connection between one digital device and another
- Input and output ports that accommodate the MIDI connections, translating back and forth between digital data to electrical voltages according to the MIDI protocol
- A transmission protocol that specifies the order and type of data to be transferred from one digital device to another

Let's look at all of these associations in the context of a simple real-world example. (Refer to the Preface for an overview of your DAW and MIDI setup.) A setup for recording and editing MIDI on a computer commonly has these five components:

- *A means to input MIDI messages*: a MIDI input device, such as a **MIDI Keyboard** or **MIDI controller.** This could be something that looks like a piano keyboard, only it doesn't generate sound itself. Often MIDI keyboards have controller functions as well, such as knobs, faders, and buttons, as shown in Figure 1.5 in Chapter 1. It's also possible to use your computer keyboard as an input device if you don't have any other controller. The MIDI input program on your computer may give you an interface on the computer screen that looks like a piano keyboard, as shown in Figure 6.2.

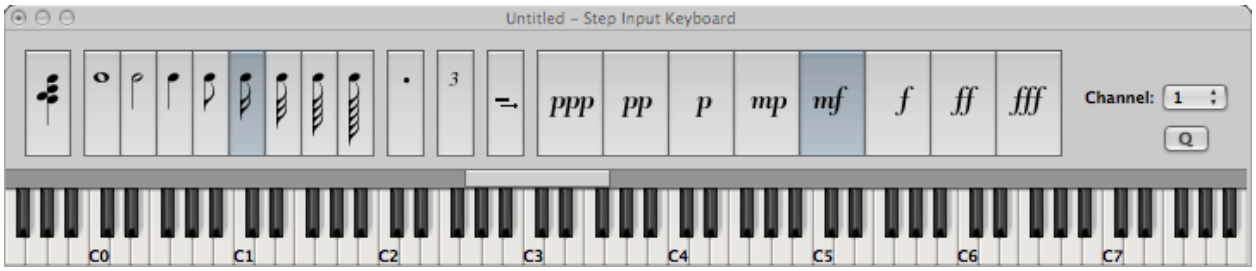


Figure 6.2 Software interface for MIDI controller from Apple Logic

- *A means to transmit MIDI messages*: a cable connecting your computer and the MIDI controller via MIDI ports or another data connection such as USB or FireWire.
- *A means to receive, record, and process MIDI messages*: a **MIDI sequencer**, which is software on your computer providing a user interface to capture, arrange, and manipulate the MIDI data. The interfaces of two commonly used software sequencers – Logic (Mac-based) and Cakewalk Sonar (Windows-based) are shown in Figures 1.29 and 1.30 of Chapter 1. The interface of Reason (Mac or Windows) is shown in Figure 6.3.
- *A means to interpret MIDI messages and create sound*: either a hardware or a software synthesizer. All three of the sequencers pictured in the aforementioned figures give you access to a variety of software synthesizers (**soft synths**, for short) and instrument plug-ins (soft synths often created by third-party vendors). If you don't have a dedicated hardware of software synthesizer within your system, you may have to resort to the soft synth supplied by your operating system. For example, Figure 6.4 shows that the only choice of synthesizer for that system setup is the Microsoft GS Wavetable Synth. Some sound cards have hardware support for sound synthesis, so this may be another option.

> **Aside:** We use the term *synthesizer* in a broad sense here, including samplers that produce sound from memory bands of recorded samples. We'll explain the distinction between synthesizers and samplers in more detail in Section 7.1.6.

- *A means to do digital-to-analog conversion and listen to the sound:* a sound card in the computer or external audio interface connected to a set of loudspeakers or headphones.

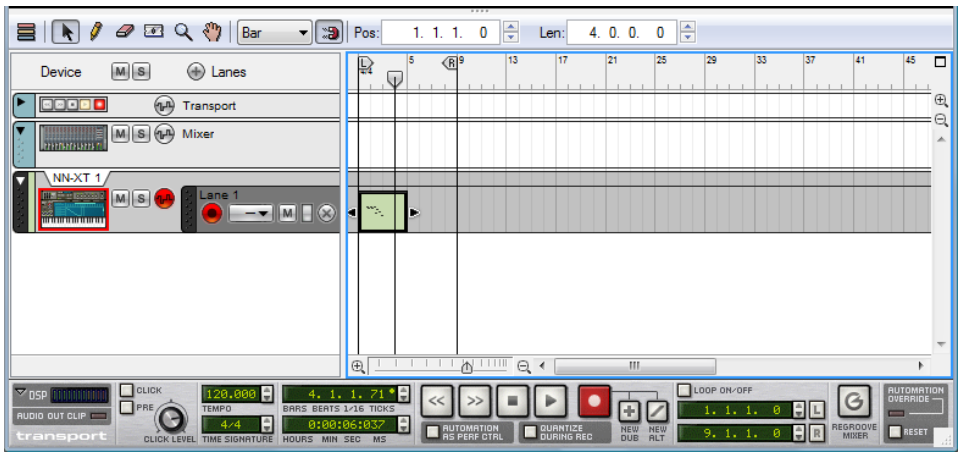


Figure 6.3 Reason's sequencer

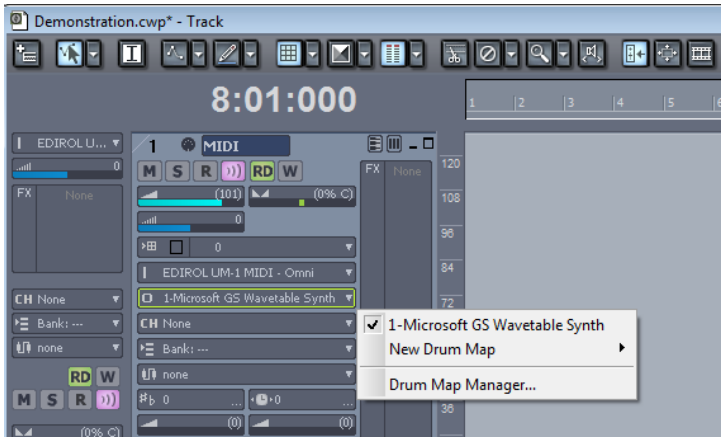


Figure 6.4 Using the operating system's soft synth

The basic setup for your audio/MIDI processing was described in Chapter 1 and is illustrated again here in Figure 6.5. This setup shows the computer handling the audio and MIDI processing. These functions are generally handled by audio/MIDI processing programs like Apple Logic, Cakewalk Sonar, Ableton Live, Steinberg Nuendo, or Digidesign Pro Tools, all of which provide a software interface for connecting the microphone, MIDI controller, sequencer, and output. All of these software systems handle both digital audio and MIDI processing, with samplers and synthesizers embedded. Details about particular configurations of hardware and software are given in Section 6.1.2.

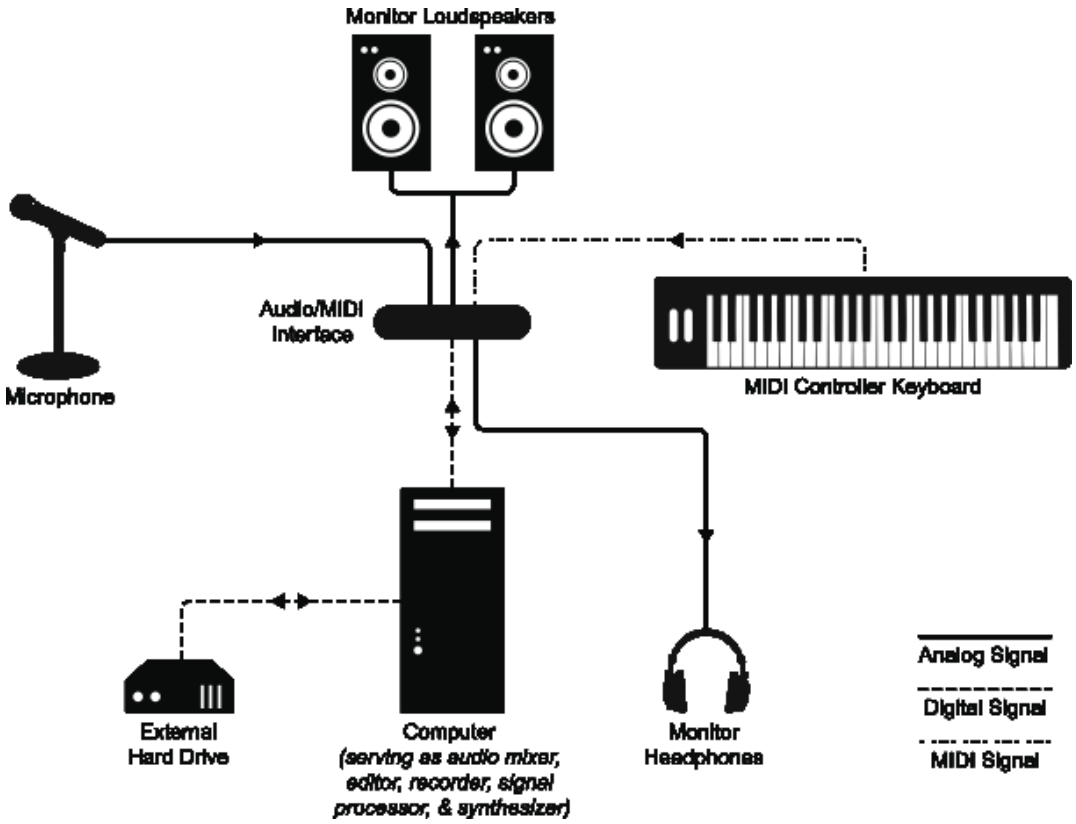


Figure 6.5 Setup for audio/MIDI processing

When you have your system properly connected and configured, you're ready to go. Now you can "arm" the sequencer for recording, press a key on the controller, and record that note in the sequencer. Most likely pressing that key doesn't even make a sound, since we haven't yet told the sound where to go. Your controller may look like a piano, but it's really just an input device sending a digital message to your computer in some agreed upon format. This is the purpose of the MIDI transmission protocol. In order for the two devices to communicate, the connection between them must be designed to transmit MIDI messages. For example, the cable could be USB at the computer end and have dual 5-pin DIN connections at the keyboard end, as shown in Figure 6.6. The message that is received by your MIDI sequencer is in a prescribed MIDI format. In the sequencer, you can save this and any subsequent messages into a file. You can also play the messages back and, depending on your settings, the notes upon playback can sound like any instrument you choose from a wide variety of choices. It is the synthesizer that turns the symbolic MIDI message into digital audio data and sends the data to the sound card to be played.

Figure 6.6 5-pin DIN connection for MIDI

6.1.3 MIDI Data Compared to Digital Audio

Consider how MIDI data differs from digital audio as described in Chapter 5. You aren't recording something through a microphone. MIDI keyboards don't necessarily make a sound when you strike a key on a piano keyboard-like controller, and they don't function as stand-alone instruments. There's no sampling and quantization going on at all. Instead, the controller is engineered to know that when a certain key is played, a symbolic message should be sent detailing the performance information.

In the case of a key press, the MIDI message would convey the occurrence of a Note On, followed by the note pitch played (e.g., middle C) and the velocity with which it was struck. The MIDI message generated by this action is only three bytes long. It's essentially just three numbers, as shown in Figure 6.7. The first byte is a value between 144 and 159. The second and third bytes are values between 0 and 127. The fact that the first value is between 144 and 159 is what makes it identifiable by the receiver as a Note On message, and the fact that it is a Note On message is what identifies the next two bytes as the specific note and velocity. A Note Off message is handled similarly, with the first byte identifying the message as Note Off and the second and third giving note and velocity information (which can be used to control note decay).

Aside: Many systems interpret a Note On message with velocity 0 as "note off" and use this as an alternative to the Note Off message.

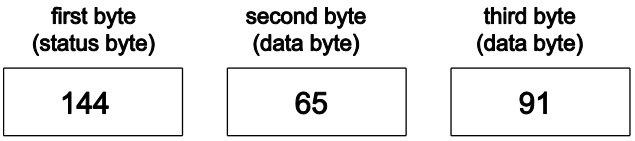


Figure 6.7 Note On message with data bytes

Let's say that the key is pressed and a second later it is released. Thus, the playing of a note for one second requires six bytes. (We can set aside the issue of how the time between the notes is stored symbolically, since it's handled at a lower level of abstraction.) How does this compare to the number of bytes required for digital audio? One second of 16-bit mono audio at a sampling rate of 44,100 Hz requires 44,100 samples/s * 2 bytes/sample = 88,200 bytes/s. Clearly, MIDI can provide a more concise encoding of sound than digital audio.

MIDI differs from digital audio in other significant ways as well. A digital audio recording of sound tries to capture the sound exactly as it occurs by sampling the sound pressure amplitude over time. A MIDI file, on the other hand, records only symbolic messages. These messages make no sound unless they are interpreted to do so by a synthesizer. When we speak of a MIDI "recording," we mean it only in the sense that MIDI data has been captured and stored – not in the sense that *sound* has actually been recorded. While MIDI messages are most frequently interpreted and synthesized into musical sounds, they can be interpreted in other ways (as we'll illustrate in Section 6.1.8.5.3). The messages mean only what the receiver interprets them to mean.

With this caveat in mind, we'll continue from here under the assumption that you're using MIDI primarily for music production since this is MIDI's most common application. When you "record" MIDI music via a keyboard controller, you're saving information about what notes to play, how hard or fast to play them, how to modulate them with pitch bend or with a sustain pedal, and what instrument the notes should sound like upon playback. If you already know how to read music and play the piano, you can enjoy the direct relationship between your input device – which is essentially a piano keyboard – and the way it saves your performance – the notes, the timing, even the way you strike the keys if you have a velocity-sensitive controller. Many MIDI sequencers have multiple ways of viewing your file, including a track view, a piano roll view, an event list view, and even a staff view – which shows the notes that you played in standard music notation. These are shown in Figure 6.8 through Figure 6.11.

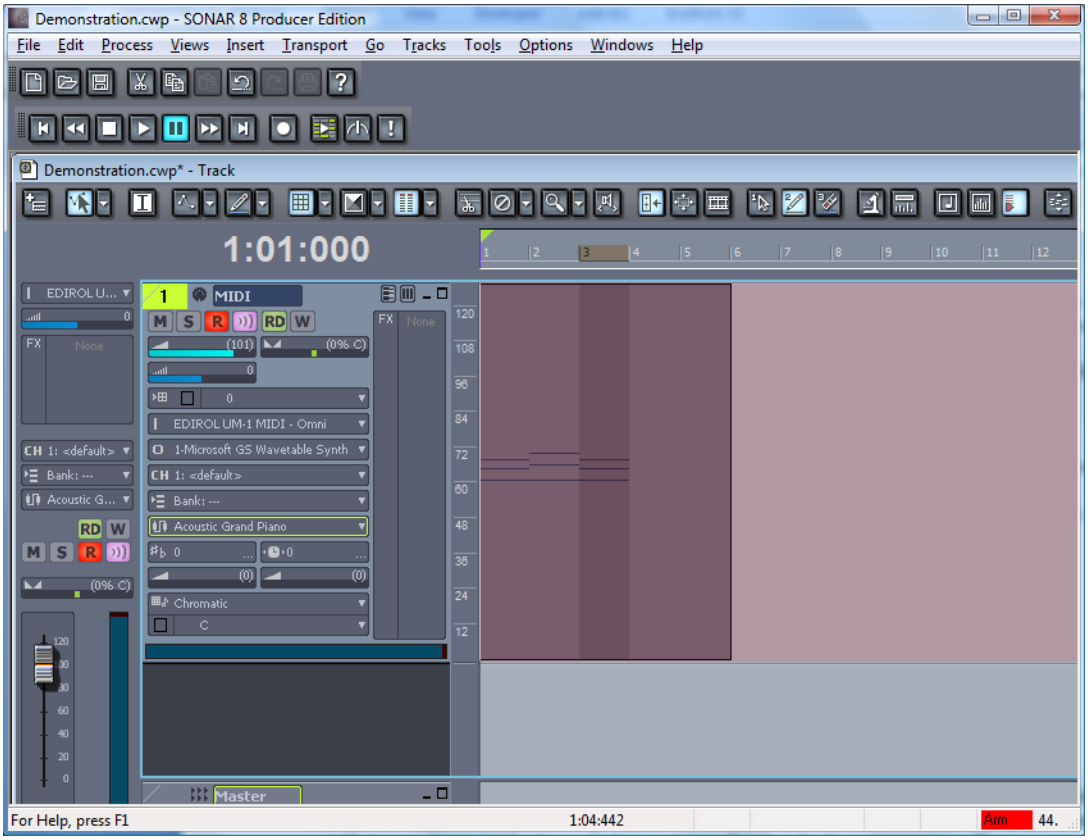


Figure 6.8 Track view in Cakewalk Sonar

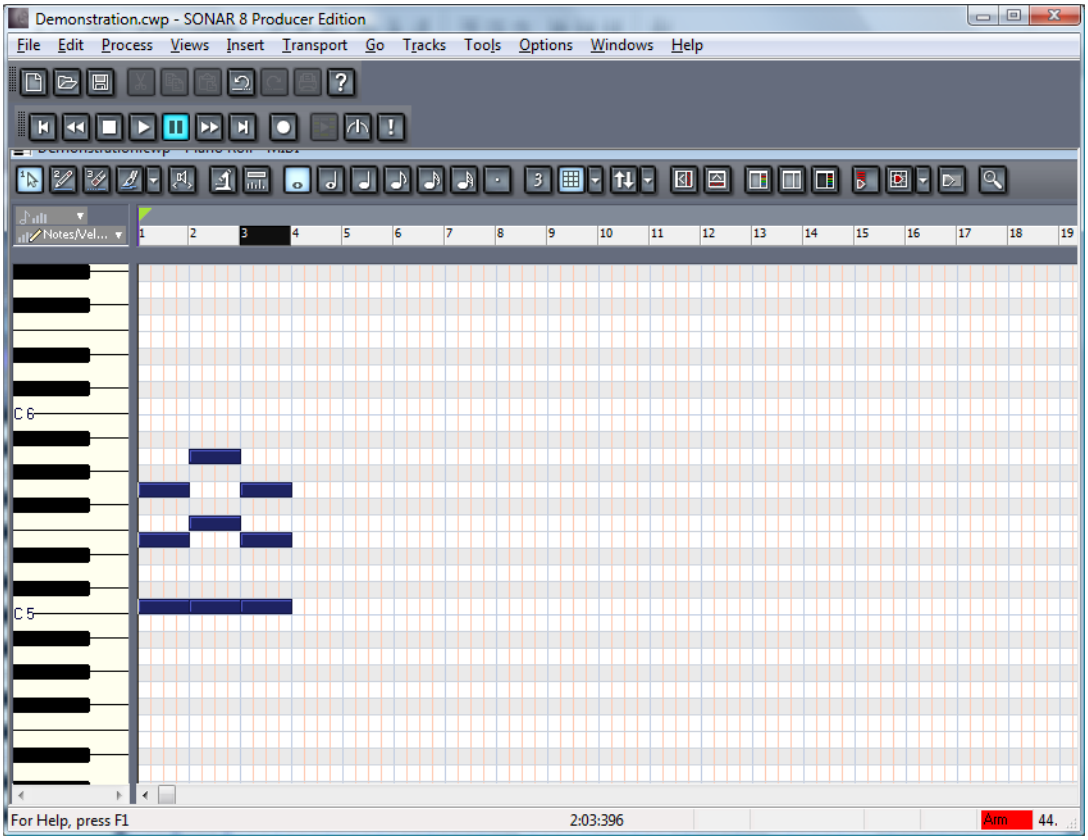


Figure 6.9 Piano roll view in Cakewalk Sonar

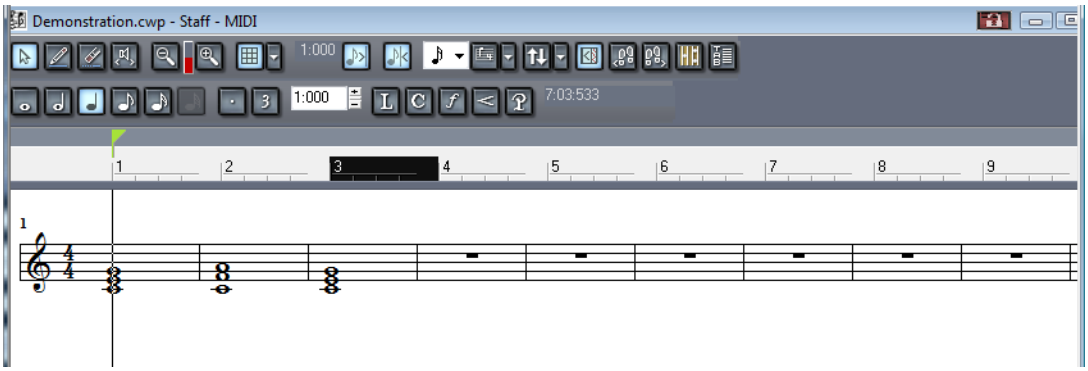


Figure 6.10 Staff view in Cakewalk Sonar

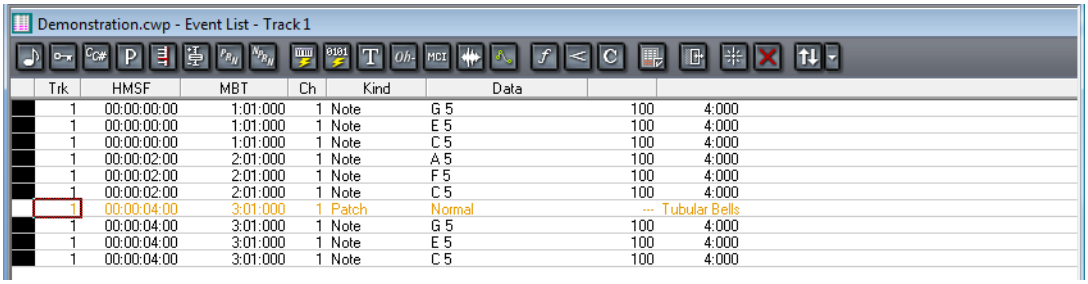


Figure 6.11 Event list view in Cakewalk Sonar

Another significant difference between digital audio and MIDI is the way in which you edit them. You can edit uncompressed digital audio down to the sample level, changing the values of individual samples if you like. You can requantize or resample the values, or process them with mathematical operations. But always they are values representing changing air pressure amplitude over time. With MIDI, you have no access to individual samples, because that's not what MIDI files contain. Instead, MIDI files contain symbolic representations of notes, key signatures, durations of notes, tempo, instruments, and so forth, making it possible for you to edit these features with a simple menu selection or an editing tool. For example, if you play a piece of music and hit a few extra notes, you can get rid of them later with an eraser tool. If your timing is a little off, you can move notes over or shorten them with the selection tool in the piano roll view. If you change your mind about the instrument sound you want or the key you'd like the piece played in, you can change these with a click of the mouse. Because the sound has not actually been synthesized yet, it's possible to edit its properties at this high level of abstraction.

> **Aside:** The word *sample* has different meanings in digital audio and MIDI. In MIDI, a sample is a small sound file representing a single instance of sound made by some instrument, like a note played on a flute.

MIDI and digital audio are simply two different ways of recording and editing sound – with an actual real-time recording or with a symbolic notation. They serve different purposes. You can actually work with both digital audio and MIDI in the same context, if both are supported by the software. Let's look more closely now at how this all happens.

6.1.4 Channels, Tracks, and Patches in MIDI Sequencers

A software MIDI sequencer serves as an interface between the input and output. A **track** is an editable area on your sequencer interface. You can have dozens or even hundreds of tracks in your sequencer. Tracks are associated with areas in memory where data is stored. In a multitrack editor, you can edit multiple tracks separately, indicating input, output, amplitude, special effects, and so forth separately for each. Some sequencers accommodate three types of tracks: audio, MIDI, and instrument. An audio track is a place to record and edit digital audio. A MIDI track stores MIDI data and is output to a synthesizer, either a software device or to an external hardware synth through the MIDI output ports. An instrument track is essentially a MIDI track combined with an internal soft synth, which in turn sends its output to the sound card. It may seem like there isn't much difference between a MIDI track and an instrument track. The main difference is that the MIDI track has to be more explicitly linked to a hardware or software synthesizer that produces its sound, whereas an instrument track has the synth, in a sense, "embedded" in it. Figure 6.12 shows each of these types of tracks.

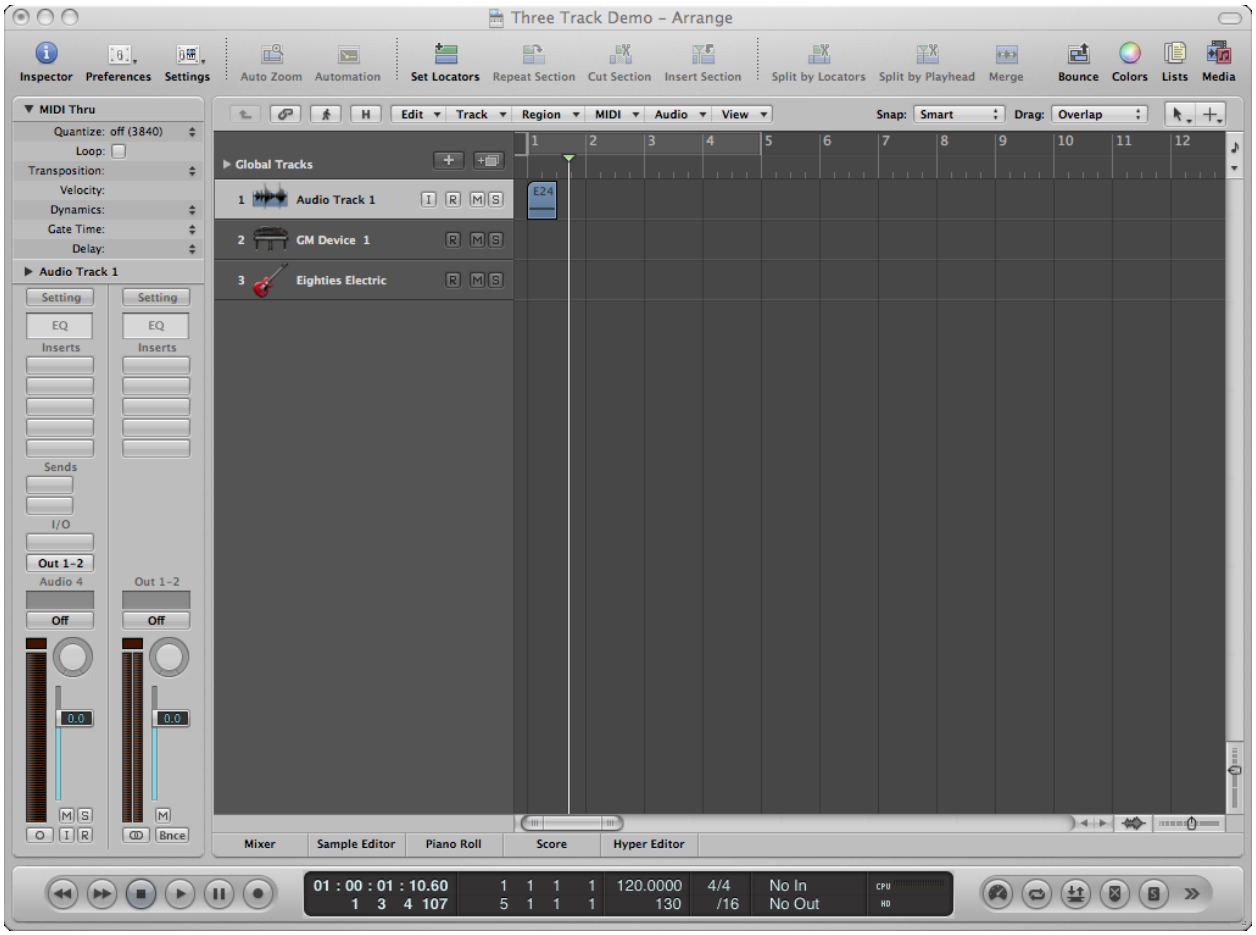


Figure 6.12 Three types of tracks in Logic

A **channel** is a communication path. According to the MIDI standard, a single MIDI cable can transmit up to 16 channels. Without knowing the details of how this is engineered, you can just think of it abstractly as 16 separate lines of communication.

There are both input and output channels. An incoming message can tell what channel it is to be transmitted on, and this can route the message to a particular device. In the sequencer pictured in Figure 6.13, track 1 is listening on all channels, indicated by the word OMNI. Track 2 is listening only on Channel 2.

The output channels are pointed out in the figure, also. When the message is sent out to the synthesizer, different channels can correspond to different instrument sounds. The parameter setting marked with a patch cord icon is the **patch**. A patch is a message to the synthesizer – just a number that indicates how the synthesizer is to operate as it creates sounds. The synthesizer can be programmed in different ways to respond to the patch setting. Sometimes the patch refers to a setting you've stored in the synthesizer that tells it what kinds of waveforms to combine or what kind of special effects to apply. In the example shown in Figure 6.13, however, the patch is simply interpreted as the choice of instrument that the user has chosen for the track. Track 1 is outputting on Channel 1 with the patch set to Acoustic Grand Piano. Track 2 is outputting on Channel 2 with the patch set to Cello.

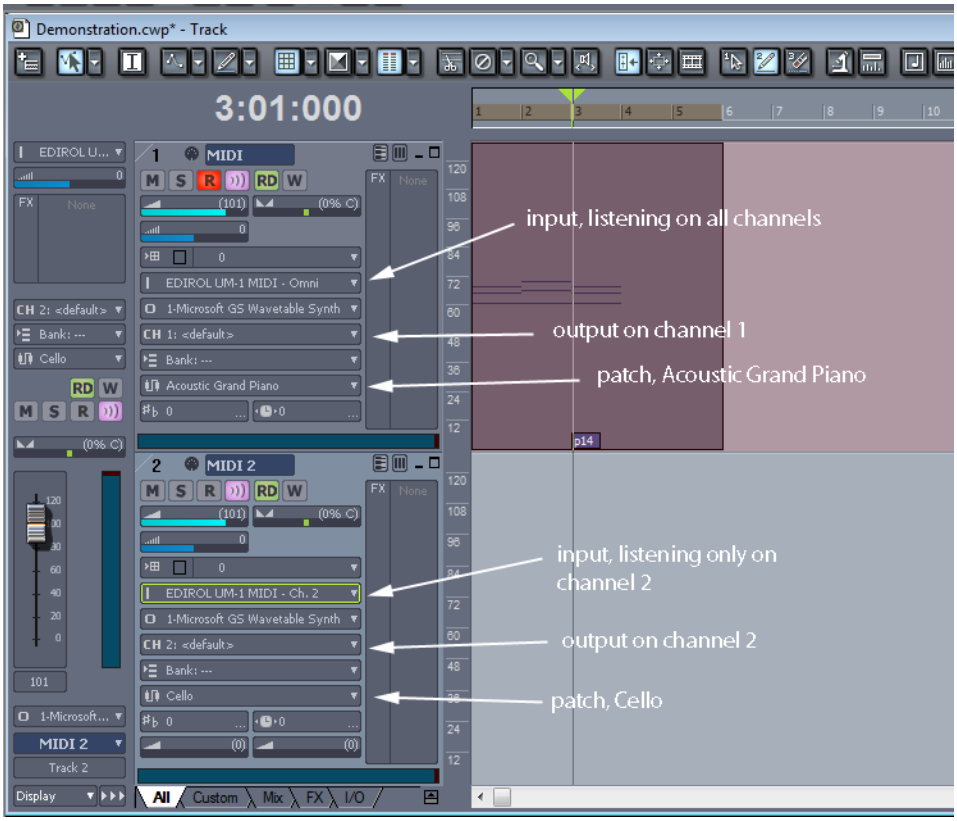


Figure 6.13 Parameter settings in Cakewalk Sonar – input and output channels and patch

It's possible to call for a patch change by means of the Program Change MIDI message. This message sends a number to the synthesizer corresponding to the index of the patch to load. A Program Change message is inserted into track 1 in Figure 6.13. You can see a little box with *p14* in it at the bottom of the track indicating that the synth should be changed to patch 14 at that point in time. This is interpreted as changing the instrument patch from a piano to tubular bells. In the setup pictured, we're just using the Microsoft GS Wavetable Synth as opposed to a more refined synth. For the computer's built-in synth, the patch number is interpreted according to the General MIDI standard. The choices of patches in this standard can be seen when you click on the drop down arrow on the patch parameter, as shown in Figure 6.14. You can see that the fifteenth patch down the list is tubular bells. (The Program Change message is still 14 because the numbering starts at 0.)

In a later section in this chapter, we'll look at other ways that the Program Change message can be interpreted by a synthesizer.

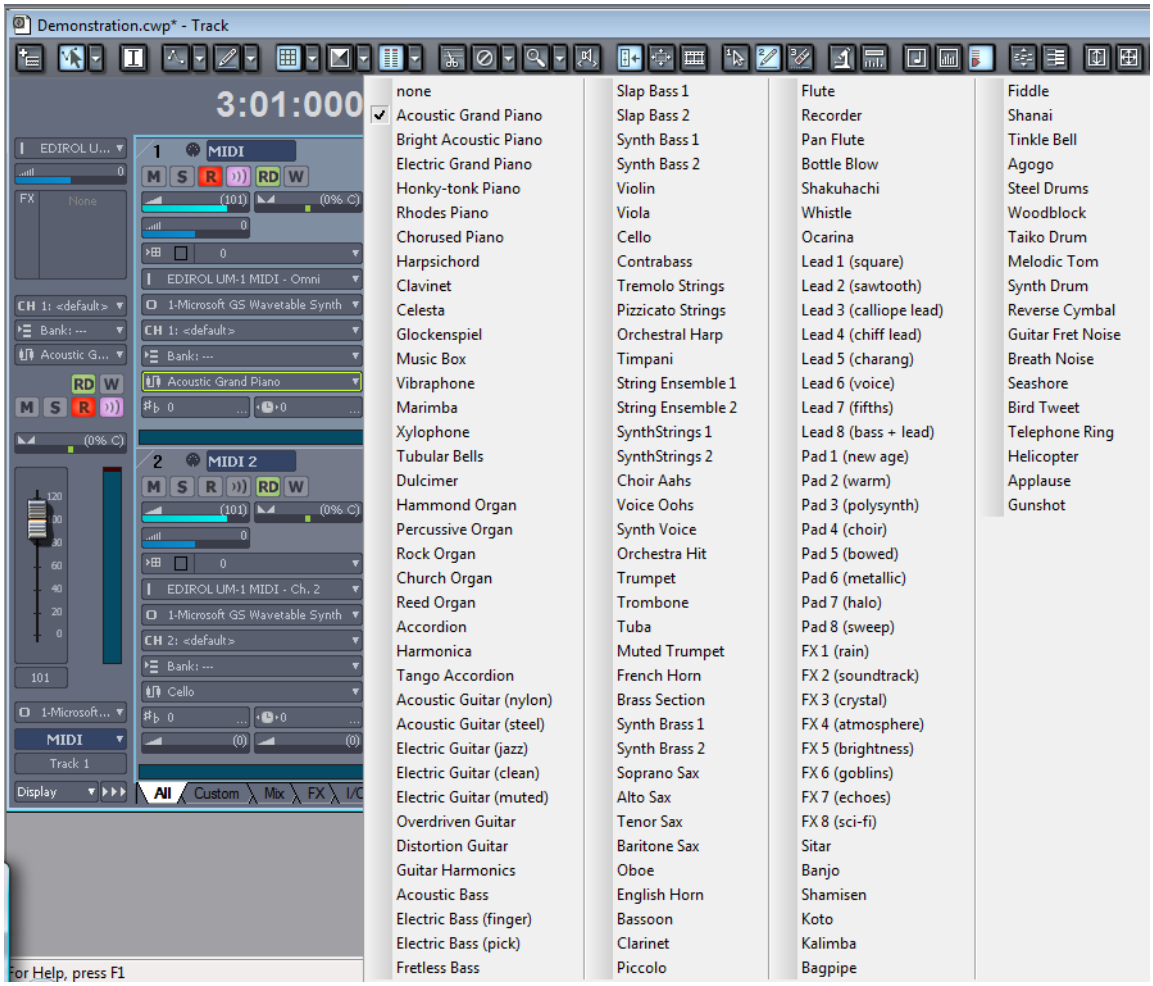


Figure 6.14 Patch assignments in the General MIDI standard, as shown in Cakewalk Sonar

6.1.5 A Closer Look at MIDI Messages

6.1.5.1 Binary, Decimal, and Hexadecimal Numbers

When you read about MIDI message formats or see them in software interfaces, you'll find that they are sometimes represented as binary numbers, sometimes as decimal numbers, and sometimes as hexadecimal numbers (hex, for short), so you should get comfortable moving from one base to another. Binary is base 2, decimal is base 10, and hex is base 16. You can indicate the base of a number with a subscript, as in 01111100_2, $7C_{16}$, and 124_{10}. Often, 0x is placed in front of hex numbers, as in 0x7C. Some sources use an H after a hex number, as in 7CH. Usually, you can tell by context what base is intended, and we'll omit the subscript unless the context is not clear. We assume that you understand number bases and can convert from one to another. If not, you should easily be able to find a resource on this for a quick crash course.

Binary and hexadecimal are useful ways to represent MIDI messages because they allow us to divide the messages in meaningful groups. A **byte** is eight bits. Half a byte is four bits, called a **nibble**. The two nibbles of a MIDI message can encode two separate pieces of information. This fact becomes important in interpreting MIDI messages, as we'll show in the

next section. In both the binary and hexadecimal representations, we can see the two nibbles as two separate pieces of information, which isn't possible in decimal notation.

A convenient way to move from hexadecimal to binary is to translate each nibble into four bits and concatenate them into a byte. For example, in the case of 0x7C, the 7 in hexadecimal becomes 0111 in binary. The C in hexadecimal becomes 1100 in binary. Thus 0x7C is 01111100 in binary. (Note that the symbols A through F correspond to decimal values 10 through 15, respectively, in hexadecimal notation.)

6.1.5.2 MIDI Messages, Types, and Formats

In Section 6.1.3, we showed you an example of a commonly used MIDI message, Note On, but now let's look at the general protocol.

MIDI messages consist of one or more bytes. The first byte of each message is a **status byte**, identifying the type of message being sent. This is followed by 0 or more **data bytes**, depending on the type of message. Data bytes give more information related to the type of message, like note pitch and velocity related to Note On.

All status bytes have a 1 as their most significant bit, and all data bytes have a 0. A byte with a 1 in its most significant bit has a value of at least 128. That is, 10000000 in binary is equal to 128 in decimal, and the maximum value that an 8 bit binary number can have (11111111) is the decimal value 255. Thus, status bytes always have a decimal value between 128 and 255. This is 10000000 through 11111111 in binary and 80 through FF in hex.

MIDI messages can be divided into two main categories: Channel messages and System messages. Channel messages contain the channel number. They can be further subdivided into voice and mode messages. Voice messages include Note On, Note Off, Polyphonic Key Pressure, Control Change, Program Change, Channel Pressure/Aftertouch, and Pitch Bend. A mode indicates how a device is to respond to messages on a certain channel. A device might be set to respond to all MIDI channels (called Omni mode), or it might be instructed to respond to polyphonic messages or only monophonic ones. **Polyphony** involves playing more than one note at the same time.

System messages are sent to the whole system rather than a particular channel. They can be subdivided into Real Time, Common, and System Exclusive messages (SysEx). The System Common messages include Tune Request, Song Select, and Song Position Pointer. The system real time messages include Timing Clock, Start, Stop, Continue, Active Sensing, and System Reset. SysEx messages can be defined by manufacturers in their own way to communicate things that are not part of the MIDI standard. The types of messages are diagrammed in Figure 6.15. A few of these messages are shown in Table 6.1.

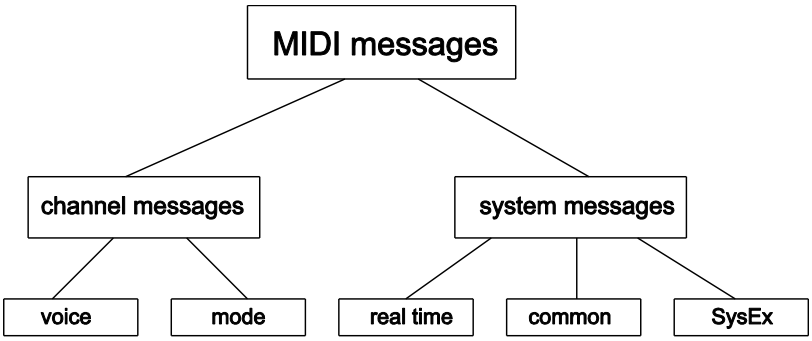


Figure 6.15 Types of MIDI messages

Hexadecimal*	Binary**	Number of Data Bytes	Description
Channel Voice Messages			
8n	1000mmmm	2	Note Off
9n	1001mmmm	2	Note On
An	1010mmmm	2	Polyphonic Key Pressure/Aftertouch
Bn	1011mmmm	2	Control Change***
Cn	1100mmmm	1	Program Change
Dn	1101mmmm	1	Channel Pressure/Aftertouch
En	1110mmmm	2	Pitch Bend Change
Channel Mode Messages			
Bn	1101mmmm	2	Selects Channel Mode
System Messages			
F3	11110011	1	Song Select
F8	11111000	0	Timing Clock
F0	11110000	variable	System Exclusive

*Each n is a hex digit between 0 and F.
**Each m is a binary digit between 0 and 1.
***Channel Mode messages are a special case of Control Change messages. The difference is in the first data byte. Data byte values 0x79 through 0x7F have been reserved in the Control Change message for information about mode changes.

Table 6.1 Examples of MIDI messages

Consider the MIDI message shown in all three numerical bases in Figure 6.16. In the first byte, the most significant bit is 1, identifying this as a status byte. This a Note On message. Since it is a channel message, it has the channel in its least significant four bits. These four bits can range from 0000 to 1111, corresponding to channels 1 through 16. (Notice that the binary value is one less than the channel number as it appears on our sequencer interface. A Note On message with 0000 in the least significant nibble indicates channel 1, one with 0001 indicates channel 2, and so forth.)

A Note On message is always followed by two data bytes. Data bytes always have a most significant bit of 0. The note to be played is 0x41, which

translates to 65 in decimal. By the MIDI standard, note 60 on the keyboard is middle C, C4. Thus, 65 is five semitones above middle C, which is F4. The second data byte gives the velocity of 0x5B, which in decimal translates to 91 (out of a maximum 127).

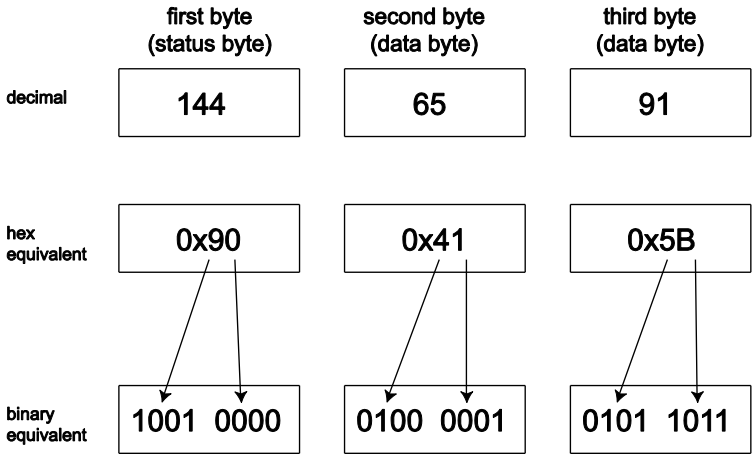


Figure 6.16 Note On message with data bytes

6.1.6 Synthesizers vs. Samplers

As we've emphasized from the beginning, MIDI is a symbolic encoding of messages. These messages have a standard way of being interpreted, so you have some assurance that your MIDI file generates a similar performance no matter where it's played in the sense that the instruments played are standardized. How "good" or "authentic" those instruments sound all comes down to the synthesizer and the way it creates sounds in response to the messages you've recorded.

We find it convenient to define **synthesizer** as any hardware or software system that generates sound electronically based on user input. Some sources distinguish between samplers and synthesizers, defining the latter as devices that use subtractive, additive AM, FM, or some other method of synthesis as opposed to having recourse to stored "banks" of samples. Our usage of the term is diagrammed in Figure 6.17

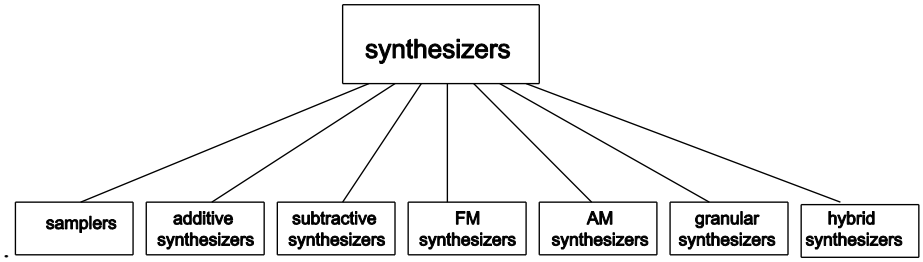


Figure 6.17 Types of synthesizers

A **sampler** is a hardware or software device that can store large numbers of sound clips for different notes played on different instruments. These clips are called **samples** (a different use from this term, to be distinguished from individual digital audio samples). A repertoire of samples stored in memory is called a **sample bank**. When you play a MIDI data stream via a sampler, these samples are pulled out of memory and played – a C on a piano, an F on a cello, or

whatever is asked for in the MIDI messages. Because the sounds played are actual recordings of musical instruments, they sound realistic.

The NN-XT sampler from Reason is pictured in Figure 6.18. You can see that there are WAV files for piano notes, but there isn't a WAV file for every single note on the keyboard. In a method called **multisampling**, one audio sample can be used to create the sound of a number of neighboring ones. The notes covered by a single audio sample constitute a **zone**. The sampler is able to use a single sample for multiple notes by pitch-shifting the sample up or down by an appropriate number of semitones. The pitch can't be stretched too far, however, without eventually distorting the timbre and amplitude envelope of the note such that the note no longer sounds like the instrument and frequency it's supposed to be. Higher and lower notes can be stretched more without our minding it, since our ears are less sensitive in these areas.

There can be more than one audio sample associated with a single note, also. For example, a single note can be represented by three samples where notes are played at three different velocities – high, medium, and low. The same note has a different timbre and amplitude envelope depending on the velocity with which it is played, so having more than one sample for a note results in more realistic sounds.

Samplers can also be used for sounds that aren't necessarily recreations of traditional instruments. It's possible to assign whatever sound file you want to the notes on the keyboard. You can create your own entirely new sound bank, or you can purchase additional sound libraries and install them (depending on the features offered by your sampler). Sample libraries come in a variety of formats. Some contain raw audio WAV or AIFF files which have to be mapped individually to keys. Others are in special sampler file formats that are compressed and automatically installable.

Figure 6.18 The NN-XT sampler from Reason

A **synthesizer**, if you use this word in the strictest sense, doesn't have a huge memory bank of samples. Instead, it creates sound more dynamically. It could do this by beginning with basic waveforms like sawtooth, triangle, or square waves and performing mathematical

operations on them to alter their shapes. The user controls this process by knobs, dials, sliders, and other input controls on the control surface of the synthesizer – whether this is a hardware synthesizer or a soft synth. Under the hood, a synthesizer could be using a variety of mathematical methods, including additive, subtractive, FM, AM, or wavetable synthesis, or physical modeling. We'll examine some of these methods in more detail in Section 6.3.1. This method of creating sounds may not result in making the exact sounds of a musical instrument. Musical instruments are complex structures, and it's difficult to model their timbre and amplitude envelopes exactly. However, synthesizers can create novel sounds that we don't often, if ever, encounter in nature or music, offering creative possibilities to innovative composers. The Subtractive Polyphonic Synthesizer from Reason is pictured in Figure 6.19.

Aside: Even the term *analog synthesizer* can be deceiving. In some sources, an analog synthesizer is a device that uses analog circuits to generate sound electronically. But in other sources, an analog synthesizer is a digital device that emulates good old fashioned analog synthesis in an attempt to get some of the "warm" sounds that analog synthesis provides. The Subtractor Polyphonic Synthesizer from Reason is described as an analog synthesizer, although it processes sound digitally.

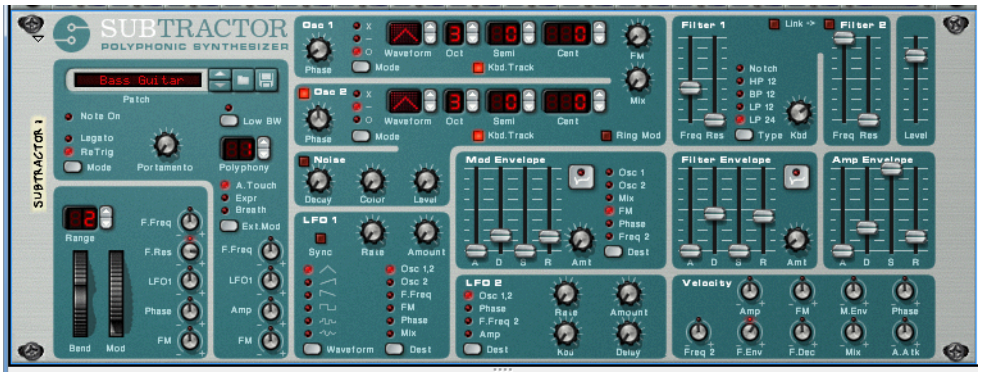


Figure 6.19 Subtractor Polyphonic Synthesizer from Reason

In reality, there's a good deal of overlap between these two ways of handling sound synthesis. Many samplers allow you to manipulate the samples with methods and parameter settings similar to those in a synthesizer. And, similar to a sampler, a, synthesizer doesn't necessarily start from nothing. It generally has basic patches (settings) that serve as a starting point, prescribing, for example, the initial waveform and how it should be shaped. That patch is loaded in, and the user can make changes from there. You can see that both devices pictured allow the user to manipulate the amplitude envelope (the ADSR settings), apply modulation, use low frequency oscillators (LFOs), and so forth. The possibilities seem endless with both types of sound synthesis devices.

Practical Exercise: Sampler Programming

6.1.7 Synthesis Methods

There are several different methods for synthesizing a sound. The most common method is called **subtractive synthesis**. Subtractive synthesizers, such as the one shown in Figure 6.19, use one or more oscillators to generate a sound with lots of harmonic content. Typically this is a sawtooth, triangle, or square wave. The idea here is that the sound you're looking for is hiding

somewhere in all those harmonics. All you need to do is subtract the harmonics you don't want, and you'll expose the properties of the sound you're trying to synthesize. The actual subtraction is done using a filter. Further shaping of the sound is accomplished by modifying the filter parameters over time using envelopes or low frequency oscillators. If you can learn all the components of a subtractive synthesize,r you're well on your way to understanding the other synthesis methods because they all use similar components.

The opposite of subtractive synthesis is **additive synthesis**. This method involves building the sound you're looking for using multiple sine waves. The theory here is that all sounds are made of individual sine waves that come together to make a complex tone. While you can theoretically create any sound you want using additive synthesis, this is a very cumbersome method of synthesis and is not commonly used.

Another common synthesis method is called **frequency modulation (FM) synthesis**. This method of synthesis works by using two oscillators with one oscillator modulating the signal from the other. These two oscillators are called the **modulator** and the **carrier**. Some really interesting sounds can be created with this synthesis method that would be difficult to achieve with subtractive synthesis. The Yamaha DX7 synthesizer is probably the most popular FM synthesizer and also holds the title of the first commercially available digital synthesizer. Figure 6.20 shows an example of an FM synthesizer from Logic Pro.

Figure 6.20 A FM synthesizer from Logic Pro

Wavetable synthesis is a synthesis method where several different single-cycle waveforms are strung together in what's called a wavetable. When you play a note on the keyboard, you're triggering a predetermined sequence of waves that transition smoothly between each other. This synthesis method is not very good at mimicking acoustic instruments, but it's very good at creating artificial sounds that are constantly in motion.

Other synthesis methods include granular synthesis, physical modeling synthesis, and phase distortion synthesis. If you're just starting out with synthesizers, begin with a simple

subtractive synthesizer and then move on to a FM synthesizer. Once you've run out of sounds you can create using those two synthesis methods, you'll be ready to start experimenting with some of these other synthesis methods.

6.1.8 Synthesizer Components

6.1.8.1 Presets

Now let's take a closer look at synthesizers. In this section, we're referring to synthesizers in the strict sense of the word – those that can be programmed to create sounds dynamically, as opposed to using recorded samples of real instruments. Synthesizer programming entails selecting an initial patch or waveform, filtering it, amplifying it, applying envelopes, applying low frequency oscillators to shape the amplitude or frequency changes, and so forth, as we'll describe below.

There are many different forms of sound synthesis, but they all use the same basic tools to generate the sounds. The difference is how the tools are used and connected together. In most cases, the software synthesizer comes with a large library of pre-built patches that configure the synthesizer to make various sounds. In your own work, you'll probably use the presets as a starting point and modify the patches to your liking. Once you learn to master the tools, you can start building your own patches from scratch to create any sound you can imagine.

6.1.8.2 Sound Generator

The first object in the audio path of any synthesizer is the **sound generator**. Regardless of the synthesis method being used, you have to start by creating some sort of sound that is then shaped into the specific sound you're looking for. In most cases, the sound generator is made up of one or more oscillators that create simple sounds like sine, sawtooth, triangle, and square waves. The sound generator might also consist of a noise generator that plays pink noise or white noise. You might also see a **wavetable oscillator** that can play a pre-recorded complex shape. If your synthesizer has multiple sound generators, there is also some sort of mixer that merges all the sounds together. Depending on the synthesis method being used, you may also have an option to decide how the sounds are combined (i.e. through addition, multiplication, modulation, etc.).

Because synthesizers are most commonly used as musical instruments, there typically is a control on the oscillator that adjusts the frequency of the sound that is generated. This frequency can usually be changed remotely over time but typically, you choose some sort of starting point and any pitch changes are applied relative to the starting frequency.

Figure 6.20 shows an example of a sound generator. In this case we have two oscillators and a noise generator. For the oscillators you can select the type of waveform to be generated. Instead of your being allowed to control the pitch of the oscillator in actual frequency values, the default frequency is defined by the note A (according to the manual). You get to choose which octave you want the A to start in and can further tune up or down from there in semitones and cents.

An option included in a number of synthesizer components is **keyboard tracking**, which allows you to control how a parameter is set or a feature is applied depending on which key on the keyboard is pressed. The keyboard tracking (Kbd. Track) button in our example sound generator defines whether you want the oscillator's frequency to change relative to the MIDI note number coming in from the MIDI controller. If this button is off, the synthesizer plays the

same frequency regardless of the note played on the MIDI controller. The Phase, FM, Mix, and Mode controls determine the way these two oscillators interact with each other.

Figure 6.21 Example of a sound generator in a synthesizer

6.1.8.3 Filters

A **filter** is another object that is often found in the audio path. A filter is an object that modifies the amplitude of specified frequencies in the audio signal. There are several types of filters. In this section, we describe the basic features of filters most commonly found in synthesizers. For more detailed information on filters, see Chapter 7.

Low-pass filters attempt to remove all frequencies above a certain point defined by the filter **cutoff frequency**. There is always a slope to the filter that defines the rate at which the frequencies are attenuated above the cutoff frequency. This is often called the **filter order**. A first order filter attenuates frequencies above the cutoff frequency at the rate of 6 dB per octave. If your cutoff frequency is 1 kHz, a first order filter attenuates 2 kHz by –6dB below the cutoff frequency, 4 kHz by –12 dB, 8 kHz by –18 dB, etc. A second order filter attenuates 12 dB per octave, a third order filter is 18 dB per octave, and a fourth order is 24 dB per octave. In some cases, the filter order is fixed, but more sophisticated filters allow you to choose the filter order that is the best fit for the sound you're looking for. The cutoff frequency is typically the frequency that has been attenuated –6 dB from the level of the frequencies that are unaffected by the filter. The space between the cutoff frequency and frequencies that are not affected by the filter is called the filter **typography.** The typography can be shaped by the filter's **resonance** control. Increasing the filter resonance creates a boost in the frequencies near the cutoff frequency.

High-pass filters are the opposite of low-pass. Instead of removing all the frequencies above a certain point, a high-pass filter removes all the frequencies below a certain point. A high-pass filter has a cutoff frequency, filter order, and resonance control just like the low-pass filter.

Bandpass filters are a combination of a high-pass and low-pass filter. A bandpass filter has a low cutoff frequency and a high cutoff frequency with filter order and resonance controls for each. In some cases, a bandpass filter is implemented with a fixed bandwidth or range of frequencies between the two cutoff frequencies. This simplifies the number of controls needed because you simply need to define a center frequency that positions the bandpass at the desired location in the frequency spectrum.

Bandstop filters (also called **notch** filters) creates a boost or cut of a defined range of frequencies. In this case the filter frequency defines the center of the notch. You might also have a bandwidth control that adjusts the range of frequencies to be boosted or cut. Finally, you have a control that adjusts the amount of change applied to the center frequency.

Figure 6.21 shows the filter controls in our example synthesizer. In this case we have two filters. Filter 1 has a frequency and resonance control and allows you to select the type of filter. The filter type selected in the example is a low-pass second order (12 dB per octave) filter. This filter also has a keyboard tracking knob where you can define the extent to which the filter cutoff frequency is changed relative to different frequencies. When you set the filter cutoff frequency using a specific key on the keyboard, the filter is affecting harmonic frequencies relative to the fundamental frequency of the key you pressed. If you play a key one octave higher, the new fundamental frequency generated by the oscillator is the same as the first harmonic of the key you were pressing when you set the filter. Consequently, the timbre of the sound changes as you move to higher and lower frequencies because the filter frequency is not changing when the oscillator frequency changes. The filter keyboard tracking allows you to change the cutoff frequency of the filter relative to the key being pressed on the keyboard. As you move to lower notes, the cutoff frequency also lowers. The knob allows you to decide how dramatically the cutoff frequency gets shifted relative to the note being pressed. The second filter is a fixed filter type (second order low-pass) with its own frequency and resonance controls and has no keyboard tracking option.

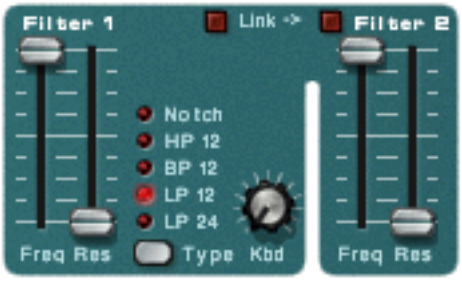


Figure 6.22 Example of filter settings in a synthesizer

We'll discuss the mathematics of filters in Chapter 7.

6.1.8.4 Signal Amplifier

The last object in the audio path of a synthesizer is a signal amplifier. The amplifier typically has a master volume control that sets the final output level for the sound. In the analog days this was a **VCA (Voltage Controlled Amplifier)** that allowed the amplitude of the synthesized sound to be controlled externally over time. This is still possible in the digital world, and it is common to have the amplifier controlled by several external modulators to help shape the amplitude of the sound as it is played. For example, you could control the amplifier in a way that lets the sound fade in slowly instead of cutting in quickly.

Figure 6.23 Master volume controller for the signal amplifier in a synthesizer

6.1.8.5 Modulation

Modulation is the process of changing a shape of a waveform over time. This is done by continuously changing one of the parameters that defines the waveform by multiplying it by some coefficient. All the major parameters that define a waveform can be modulated, including its frequency, amplitude, and phase. A graph of the coefficients by which the waveform is modified shows us the shape of the modulation over time. This graph is sometimes referred to as an **envelope** that is imposed over the chosen parameter, giving it a continuously changing shape. The graph might correspond to a continuous function, like a sine, triangle, square, or sawtooth. Alternative, the graph might represent a more complex function, like the ADSR envelope illustrated in Figure 6.25 illustrates a particular type of envelope, called ADSR.

We'll see look at mathematics of amplitude, phase, and frequency modulation in Section 3. For now, we'll focus on LFOs and ADSR envelopes, commonly-used tools in synthesizers.

6.1.8.6 LFO

LFO stands for **low frequency oscillator**. An LFO is simply an oscillator just like the ones found in the sound generator section of the synthesizer. The difference here is that the LFO is not part of the audio path of the synthesizer. In other words, you can't hear the frequency generated by the LFO. Even if the LFO was put into the audio path, it oscillates at frequencies well below the range of human hearing so it isn't heard anyway. A LFO oscillates anywhere from 10 Hz down to a fraction of a Hertz.

LFO's are used like envelopes to modulate parameters of the synthesizer over time. Typically you can choose from several different waveforms. For example, you can use an LFO with a sinusoidal shape to change the pitch of the oscillator over time, creating a vibrato effect. As the wave moves up and down, the pitch of the oscillator follows. You can also use an LFO to control the sound amplitude over time to create a pulsing effect.

Figure 6.25 shows the LFO controls on a synthesizer. The Waveform button toggles the LFO between one of six different waveforms. The Dest button toggles through a list of destination parameters for the LFO. Currently, the LFO is set to create a triangle wave and apply it to the pitch of Oscillators 1 and 2. The Rate knob defines the frequency of the LFO and the Amount knob defines the amplitude of the wave or the amount of modulation that is applied. A higher amount creates a more dramatic change to the destination parameter. When the Sync button is engaged, the LFO frequency is synchronized to the incoming tempo for your song based on a division defined by the Rate knob such as a quarter note or a half note.

Figure 6.24 LFO controls on a synthesizer

6.1.8.7 Envelopes

Most synthesizers have at least one envelope object. An envelope is an object that controls a synthesizer parameter over time. The most common application of an envelope is an **amplitude envelope**. An amplitude envelope gets applied to the signal amplifier for the synthesizer. Envelopes have four parameters: attack time, decay time, sustain level, and release time. The sustain level defines the amplitude of the sound while the note is held down on the keyboard. If the sustain level is at the maximum value, the sound is played at the amplitude defined by the master volume controller. Consequently, the sustain level is typically an attenuator that reduces rather than amplifies the level. If the other three envelope parameters are set to zero time, the sound is simply played at the amplitude defined by the sustain level relative to the master volume level.

The attack and decay values control how the sound begins. If the attack is set to a positive value, the sound fades in to the level defined by the master volume level over the period of time indicated in the attack. When the attack fade-in time completes, the amplitude moves to the sustain level. The decay value defines how quickly that move happens. If the decay is set to the lowest level, the sound jumps instantly to the sustain level once the attack completes. If the decay time has a positive value, the sound slowly fades down to the sustain level over the period of time defined by the decay after the attack completes.

The release time defines the amount of time it takes for the sound level to drop to silence after the note is released. You might also call this a fade-out time. Figure 6.23 is a graph showing these parameters relative to amplitude and time. Figure 6.24 shows the amplitude envelope controls on a synthesizer. In this case, the envelope is bypassed because the sustain is set to the highest level and everything else is at the lowest value.

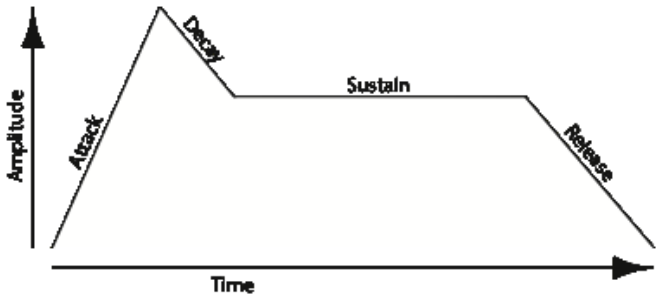


Figure 6.25 Graph of ADSR envelope

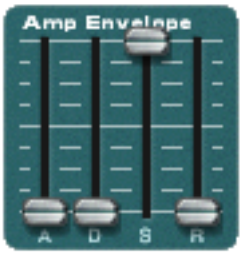


Figure 6.26 Envelope controls on a synthesizer

Envelopes can be used to control almost any synthesizer parameter over time. You might use an envelope to change the cutoff frequency of a filter or the pitch of the oscillator over time. Generally speaking, if you can change a parameter with a slider or a knob, you can modulate it over time with an envelope.

6.1.8.8 MIDI Modulation

You can also use incoming MIDI commands to modulate parameters on the synthesizer. Most synthesizers have a pre-defined set of MIDI commands it can respond to. More powerful synthesizers allow you to define any MIDI command and apply it to any synthesizer parameter. Using MIDI commands to modulate the synthesizer puts more power in the hands of the performer.

Here's an example of how MIDI modulation can work. Piano players are used to getting a different sound from the piano depending on how hard they press the key. To recreate this touch sensitivity, most MIDI keyboards change the velocity value of the Note On command depending on how hard the key is pressed. However, MIDI messages can be interpreted in whatever way the receiver chooses. Figure 6.26 shows how you might use velocity to modulate the sound in the synthesizer. In most cases, you would expect for the sound to get louder when the key is pressed harder. If you increase the Amp knob in the velocity section of the synthesizer, the signal amplifier level increases and decreases with the incoming velocity information. In some cases, you might also expect to hear more harmonics with the sound if the key is pressed harder. Increasing the value for the F.Env knob adjusts the depth at which the filter envelope is applied to the filter cutoff frequency. A higher velocity means that the filter envelope makes a more dramatic change to the filter cutoff frequency over time.

Figure 6.27Velocity modulation controls on a synthesizer

Some MIDI keyboards can send After Touch or Channel Pressure commands if the pressure at which the key is held down changes. You can use this pressure information to modulate a synthesizer parameter. For example, if you have a LFO applied to the pitch of the oscillator to create a vibrato effect, you can apply incoming key pressure data to adjust the LFO amount. This way the vibrato is only applied when the performer desires it by increasing the

pressure at which he or she is holding down the keys. Figure 6.27 shows some controls on a synthesizer to apply After Touch and other incoming MIDI data to four different synthesizer parameters.

Figure 6.28 After Touch modulation controls on a synthesizer

6.2 Applications

6.2.1 Linking Controllers, Sequencers, and Synthesizers

In this section, we'll look at how MIDI is handled in practice.

First, let's consider a very simple scenario where you're generating electronic music in a live performance. In this situation, you need only a MIDI controller and a synthesizer. The controller collects the performance information from the musician and transmits that data to the synthesizer. The synthesizer in turn generates a sound based on the incoming control data. This all happens in real-time, the assumption being that there is no need to record the performance.

Now suppose you want also want to capture the musician's performance. In this situation, you have two options. The first option involves setting up a microphone and making an audio recording of the sounds produced by the synthesizer during the performance. This option is fine assuming you don't ever need to change the performance, and you have the resources to deal with the large file size of the digital audio recording.

The second option is simply to capture the MIDI performance data coming from the controller. The advantage here is that the MIDI control messages constitute much less data than the data that would be generated if a synthesizer were to transform the performance into digital audio. Another advantage to storing in MIDI format is that you can go back later and easily change the MIDI messages, which generally is a much easier process than digital audio processing. If the musician played a wrong note, all you need to do is change the data byte representing that note number, and when the stored MIDI control data is played back into the synthesizer, the synthesizer generates the correct sound. In contrast, there's no easy way to change individual notes in a digital audio recording. Pitch correction plug-ins can be applied to digital audio, but they potentially distort your sound, and sometimes can't fix the error at all.

So let's say you go with option two. For this, you need a MIDI sequencer between the controller and the synthesizer. The sequencer captures the MIDI data from the controller and sends it on to the synthesizer. This MIDI data is stored in the computer. Later, the sequencer can recall the stored MIDI data and send it again to the synthesizer, thereby perfectly recreating the original performance.

The next questions to consider are these: Which parts of this setup are hardware and which are software? And how do these components communicate with each other? Four different configurations for linking controllers, sequencers, and synthesizers are diagrammed in Figure 6.28. We'll describe each of these in turn.

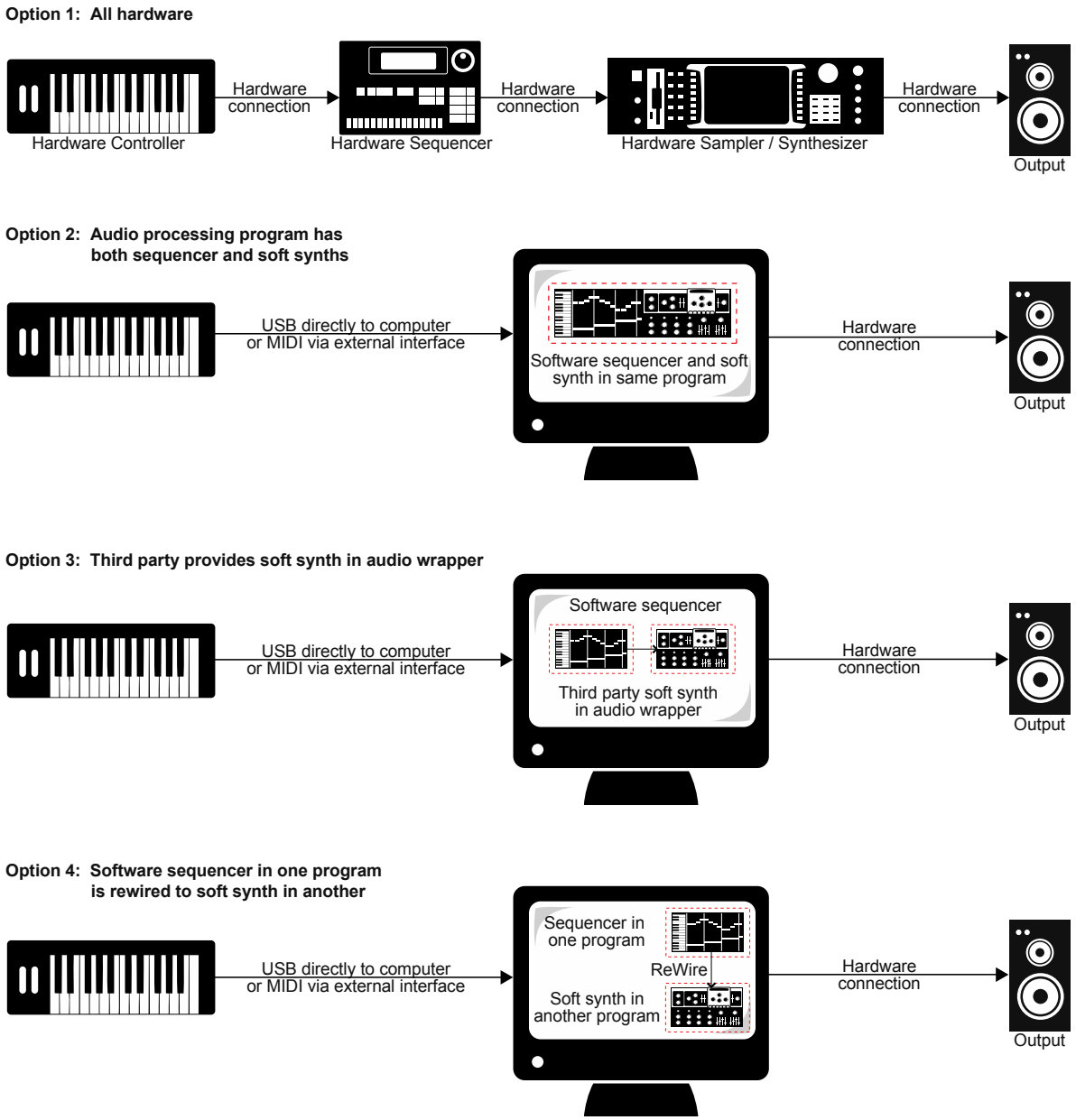


Figure 6.29 Configurations linking controllers, sequencers, and synthesizers

In the early days of MIDI, hardware synthesizers were the norm, and dedicated hardware sequencers also existed, like the one shown in Figure 6.29. Thus, an entire MIDI setup could be accomplished through hardware, as diagrammed in Option 1 of Figure 6.28.

Figure 6.30 A dedicated MIDI sequencer

Now that personal computers have ample memory and large external drives, software solutions are more common. A standard setup is to have a MIDI controller keyboard connected to your computer via USB or through a USB MIDI interface like the one shown in Figure 6.30. Software on the computer serves the role of sequencer and synthesizer. Sometimes one program can serve both roles, as diagrammed in Option 2 of Figure 6.28. This is the case, for example, with both Cakewalk Sonar and Apple Logic, which provide a sequencer and built-in soft synths. Sonar's sample-based soft synth is called the TTS, shown in Figure 6.31. Because samplers and synthesizers are often made by third party companies and then incorporated into software sequencers, they can be referred to as **plug-ins**. Logic and Sonar have numerous plug-ins that are automatically installed – for example, the EXS24 sampler (Figure 6.32) and the EFM1 FM synthesizer (Figure 6.33).

Figure 6.31 A USB MIDI interface for a personal computer

Figure 6.32 TTS soft synth in Cakewalk Sonar

Figure 6.33 EXS24 sampler in Logic

Figure 6.34 EFM1 synthesizer in Logic

Some third-party vendor samplers and synthesizers are not automatically installed with a software sequencer, but they can be added by means of a **software wrapper**. The software wrapper makes it possible for the plug-in to run natively inside the sequencer software. This way you can use the sequencer's native audio and MIDI engine and avoid the problem of having several programs running at once and having to save your work in multiple formats. Typically what happens is a developer creates a standalone soft synth like the one shown in Figure 6.34. He can then create an Audio Unit wrapper that allows his program to be inserted as an Audio Unit instrument, as shown for Logic in Figure 6.35. He can also create a VSTi wrapper for his synthesizer that allows the program to be inserted as a VSTi instrument in a program like Cakewalk, an MAS wrapper for MOTU Digital Performer, and so forth. A setup like this is shown in Option 3 of Figure 6.28.

Aside: As you work with MIDI and digital audio, you'll develop a large vocabulary of abbreviations and acronyms. In the area of plug-ins, the abbreviations relate to standardized formats that allow various software components to communicate with each other. **VSTi** stands for **virtual studio technology instrument**, created and licensed by Steinberg. This is one of the most widely used formats. **Dxi** is a plug-in format based on Microsoft Direct X, and is a Windows-based format. **AU**, standing for **audio unit**, is a Mac-based format. **MAS** refers to plug-ins that work with Digital Performer, an audio/MIDI processing system created by the MOTU company. **RTAS** (**Real-Time AudioSuite**) is the protocol developed by Digidesin for Pro Tools. You need to know which formats are compatible on which platforms. You can find the most recent information through the documentation of your software or through on-line sources.

Figure 6.35 Soft synth running as a standalone application

Figure 6.36 Soft synth running in Logic through an Audio Unit instrument wrapper

An alternative to built-in synths or installed plug-ins is to have more than one program running on your computer, each serving a different function to create the music. An example of such a configuration would be to use Sonar or Logic as your sequencer, and then use Reason to provide a wide array of samplers and synthesizers. This setup introduces a new question: How do the different software programs communicate with each other?

One strategy is to create little software objects that pretend to be MIDI or audio inputs and outputs on the computer. Instead of linking directly to input and output hardware on the computer, you use these software objects as virtual cables. That is, the output from the MIDI

sequencer program goes to the input of the software object, and the output of the software object goes to the input of the MIDI synthesis program. The software object functions as a virtual wire between the sequencer and synthesizer. The audio signal output by the soft synth can be routed directly to a physical audio output on your hardware audio interface, to a separate audio recording program, or back into the sequencer program to be stored as sampled audio data. This configuration is diagrammed in Option 4 of Figure 6.28.

An example of this virtual wire strategy is the **Rewire** technology developed by Propellerhead and used with its sampler/synthesizer program, Reason. Figure 6.36 shows how a track in Sonar can be rewired to connect to a sampler in Reason. The track labeled "MIDI to Reason" has the MIDI controller as its input and Reason as its output. The NN-XT sampler in Reason translates the MIDI commands into digital audio and sends the audio back to the track labeled "Reason to audio." This track sends the audio output to the sound card.

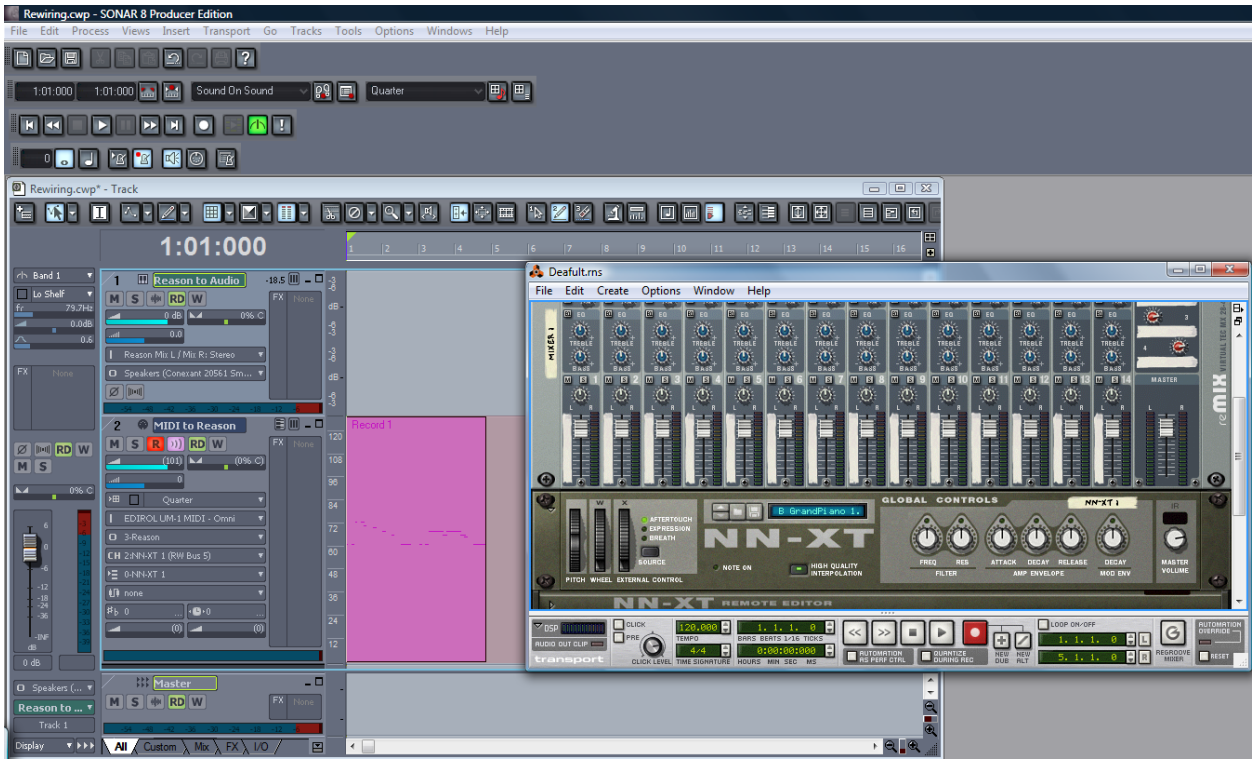


Figure 6.37 Rewiring between Sonar and Reason

Other virtual wiring technologies are available. **Soundflower** is another program for Mac OS X developed by Cycling '74 that creates virtual audio wires that can be routed between programs. **CoreMIDI Virtual Ports** are integrated into Apple's CoreMIDI framework on Mac OS X. A similar technology called **MIDI Yoke** (developed by a third party) works in the Windows operating systems. **Jack** is an open source tool that runs on Windows, Mac OS X, and various UNIX platforms to create virtual MIDI and audio objects.

6.2.2 Creating Your Own Synthesizer Sounds

Section 6.1.8 covered the various components of a synthesizer. Now that you've read about the common objects and parameters available on a synthesizer you should have an idea of what can be done with one. So how do you know which knobs to turn and when? There's not an easy

answer to that question. The thing to remember is that there are no rules. Use your imagination and don't be afraid to experiment. In time, you'll develop an instinct for programming the sounds you can hear in your head. Even if you don't feel like you can create a new sound from scratch, you can easily modify existing patches to your liking, learning to use the controls along the way. For example, if you load up a synthesizer patch and you think the notes cut off too quickly, just increase the release value on the amplitude envelope until it sounds right.

Practical Exercise: Subtractive Synthesis

Most synthesizers use obscure values for the various controls, in the sense that it isn't easy to relate numerical settings to real-world units or phenomena. Reason uses control values from 0 to 127. While this nicely relates to MIDI data values, it doesn't tell you much about the actual parameter. For example, how long is an attack time of 87? The answer is, it doesn't really matter. What matters is what it sounds like. Does an attack time of 87 sound too short or too long? While it's useful to understand what the controller affects, don't get too caught up in trying to figure out what exact value you're dialing in when you adjust a certain parameter. Just listen to the sound that comes out of the synthesizer. If it sounds good, it doesn't matter what number is hiding under the surface. Just remember to save the settings so you don't lose them.

Max Demo: Subtractonaut Synthesizer

6.2.3 Making and Loading Your Own Samples

Sometimes you may find that you want a certain sound that isn't available in your sampler. In that case, you may want to create your own sample

Practical Exercise: Programming Sampler Instruments

If you want to create a sampler patch that sounds like a real instrument, the first thing to do is find someone who has the instrument you're interested in and get them to play different notes one at a time while you record them. To make sure you don't have to stretch the pitch too far for any one sample, make sure you get a recording for at least three notes per octave within the instrument's range.

Some instruments can sound different depending on how they are played. For example, a trumpet sounds very different with a mute inserted on the horn. If you want your sampler to be able to create the muted sound, you might be able to mimic it using filters in the sampler, but you'll get better results by just recording the real trumpet with the mute inserted. Then you can program the sampler to play the muted samples instead of the unmuted ones when it receives a certain MIDI command.

Video Tutorial: Guitar Sampler Demo

Keep in mind that the more samples you have, the more RAM space the sampler requires. If you have 500 MB worth of recorded samples and you want to use them all, the sampler is going to use up 500 MB of RAM on your computer. The trick is finding the right balance between having enough samples so that none of them get stretched unnaturally, but not so many that you use up all the RAM in your computer. As long as you have a real person and a real instrument to record, go ahead and get as many samples as you can. It's much easier to delete the ones you don't need than to schedule another recording session to get the two notes you forgot to record.

Once you have all your samples recorded, you need to edit them and add all the metadata required by the sampler. In order for the sampler to do what it needs to do with the audio files, the files need to be in an uncompressed file format. Usually this is WAV or AIF format. Some samplers have a limit on the sampling rate they can work with. Make sure you convert the samples to the rate required by the sampler before you try to use them.

The first bit of metadata you need to add to each sample is a loop start and loop end marker. Because you're working with prerecorded sounds, the sound doesn't necessarily keep playing just because you're still holding the key down on the keyboard. You could just record your samples so they hold on for a long time, but that would use up an unnecessary amount of RAM. Instead, you can tell the sampler to play the file from the beginning and stop playing the file when the key on the keyboard is released. If the key is still down when the sampler reaches the end of the file, the sampler can start playing a small portion of the sample over and over in an endless loop until the note is released. The challenge here is finding a portion of the sample that loops naturally without any clicks or other swells in amplitude or harmonics.

Video Tutorial: Recorder Sampler Demo

Figure 6.37 shows a loop defined in a sample editing program. On the left side of the screen you can see the overall waveform of the sample, in this case a violin. In the time ruler above the waveform you can see a green bar labeled Sustaining Loop. This is the portion of the sample that is looped. On the right side of the screen you can see a close up view of the loop point. The left half of the wave is the end of the sample, and the right part of the wave is the start of the loop point. The trick here is to line up the loop points so the two parts intersect with the zero amplitude cross point. This way you avoid any clicks or pops that might be introduced when the sampler starts looping the playback.

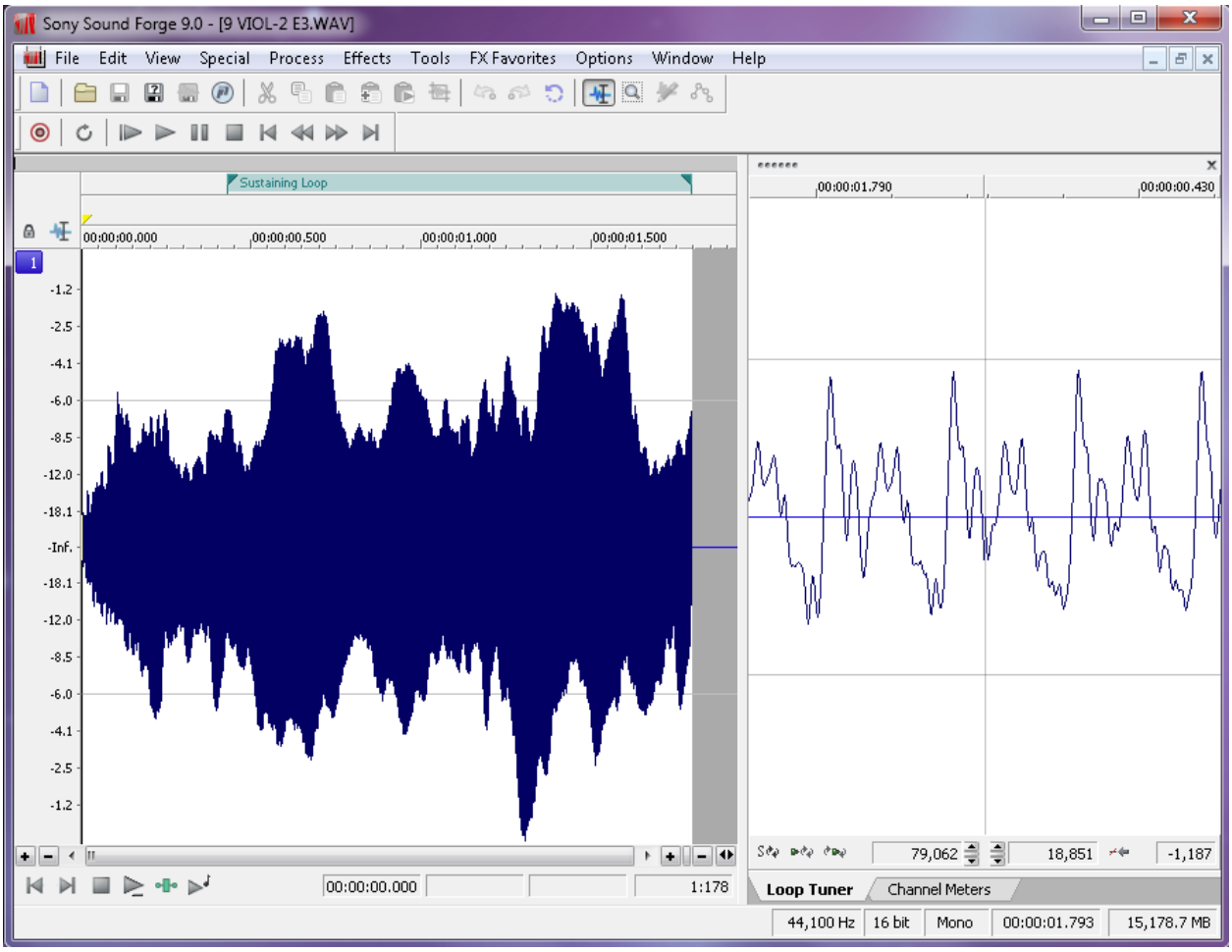


Figure 6.38 Editing sample loop points

In some sample editors you can also add other metadata that saves you programming time later. For WAV and AIF files, you can add information about the root pitch of the sample and the range of notes this sample should cover. You can also add information about the loop behavior. For example, do you want the sample to continue to loop during the release of the amplitude envelope, or do you want it to start playing through to the end of the file? You could set the sample not to loop at all and instead play as a "one shot" sample. This means the sample ignores Note Off events and play the sample from beginning to end every time. Some samplers can read that metadata and do some pre-programming for you on the sampler when you load the sample.

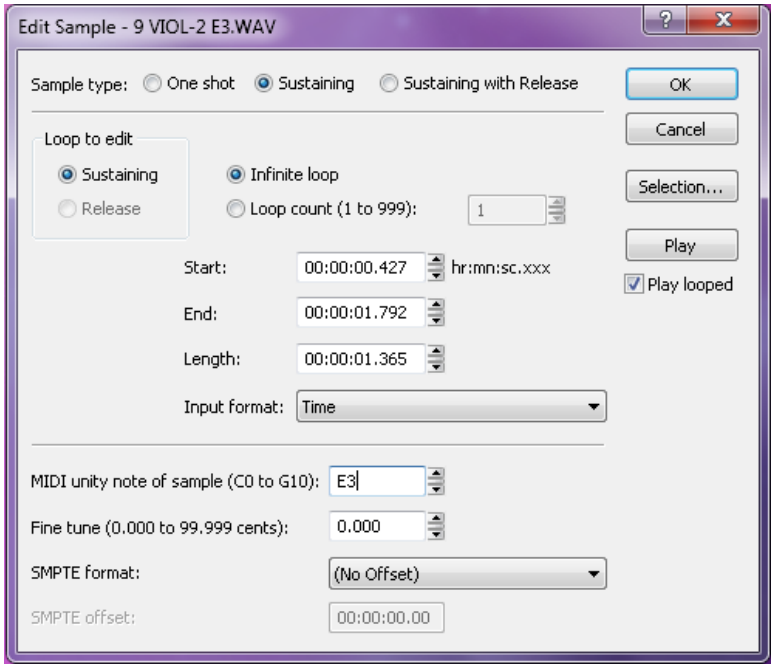


Figure 6.39 Adding sample metadata such as root pitch and loop type

Once the samples are ready, you can load them into the sampler. If you weren't able to add the metadata about root key and loop type, you'll have to add that manually into the sampler for each sample. You'll also need to decide which notes trigger each sample and which velocities each sample responds to. This process of assigning root keys and key ranges to all your samples is a time consuming but essential process. Figure 6.39 shows a list of sample WAV files loaded in a software sampler. In the figure we have the sample "PianoC43.wav" selected. You can see in the center of the screen the span of keys that have been assigned to that sample. Along the bottom row of the screen you can see the root key, loop, and velocity assignments for that sample.

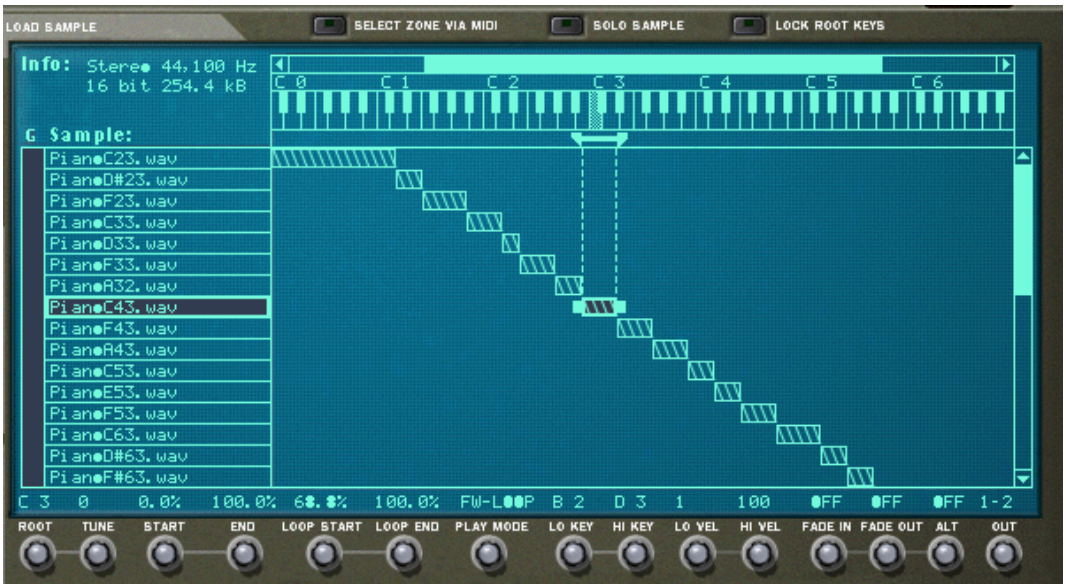


Figure 6.40 Loading and assigning samples in a software sampler

Once you have all your samples loaded and assigned, each sample can be passed through a filter and amplifier which can in turn be modulated using envelopes, LFO, and MIDI controller commands. Most samplers let you group samples together into zones or keygroups allowing you to apply a single set of filters, envelopes, etc. This feature can save a lot of time in programming. Imagine programming all of those settings on each of 100 samples without losing track of how far you are in the process. Figure 6.40 shows all the common synthesizer objects being applied to the "PianoC43.wav" sample.

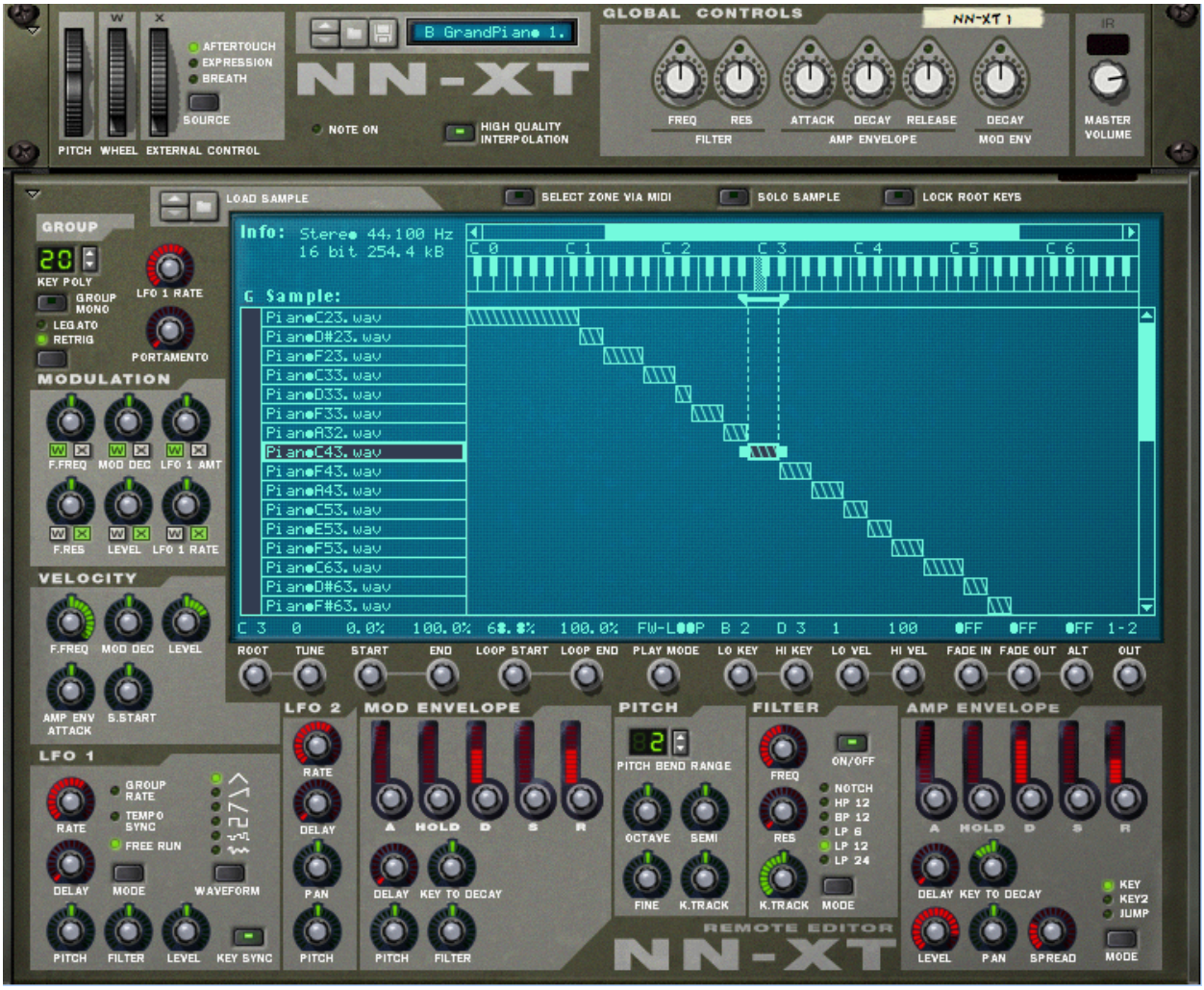


Figure 6.41 Synthesizer functions that can be applied to samples once they have been loaded and assigned

6.2.4 Data Flow and Performance Issues in Audio/MIDI Recording

Combined audio/MIDI recording can place high demands on a system, requiring a fast CPU and hard drive, an appropriate choice of audio driver, and a careful setting of the audio buffer size. These components affect your ability to record, process, and play sound, particularly in real time.

In Chapter 5, we introduced the subject of latency in digital audio systems. The problem of latency is compounded when MIDI data is added to the mix. A common frustration in MIDI recording sessions is that there can be an audible difference between the moment when you press a key on a MIDI controller keyboard and the moment when you hear the sound coming out of the headphones or monitors. In this case, the latency is the result of your buffer size. The MIDI signal generated by the key press must be transformed into digital audio by a synthesizer or sampler, and the digital data is then placed in the output buffer. This sound is not heard until the buffer is filled up. When the buffer is full, it undergoes ADC and is set sent to the headphones or monitors. Playback latency results when the buffer is too large. As discussed in Chapter 5, you can reduce the playback latency by using a low latency audio driver like ASIO or reducing the buffer size if this option is available in your driver. However, if you make the buffer size too low, you'll have breaks in the sound when the CPU cannot keep up with the number of times it has to empty the buffer.

Another potential bottleneck in digital audio playback is the hard drive. Fast hard drives are a must when working with digital audio, and it is also important to use a dedicated hard drive for your audio files. If you're storing your audio files on the same hard drive as your operating system, you'll need a larger playback buffer to accommodate all the times the hard drive is busy delivering system data instead of your audio. If you get a second hard drive and use it only for audio files, you can usually get away with a much smaller playback buffer, thereby reducing the playback latency.

When you use software instruments, there are other system resources besides the hard drive that also become a factor to consider. Software samplers require a lot of system RAM because all the audio samples have to be loaded completely in RAM in order for them to be instantly accessible. On the other hand, software synthesizers that generate the sound dynamically can be particularly hard on the CPU. The CPU has to mathematically create the audio stream in real time, which is a more computationally intense process than simply playing an audio stream that already exists. Synthesizers with multiple oscillators can be particularly problematic. Some programs let you offload individual audio or instrument tracks to another CPU. This could be a networked computer running a processing node program or some sort of dedicated processing hardware connected to the host computer. If you're having problems with playback dropouts due to CPU overload and you can't add more CPU power, another strategy is to render the instrument audio signal to an audio file that is played back instead of generated live (often called "freezing" a track). However, this effectively disables the MIDI signal and the software instrument so if you need to make any changes to the rendered track, you need to go back to the MIDI data and re-render the audio.

6.2.5 Non-Musical Applications for MIDI

6.2.5.1 MIDI Show Control

MIDI is not limited to use for digital music. There have been many additions to the MIDI specification to allow MIDI to be used in other areas. One addition is the MIDI Show Control Specification. MIDI Show Control (MSC) is used in live entertainment to control sound, lighting, projection, and other automated features in theatre, theme parks, concerts, and more.

MSC is a subset of the MIDI Systems Exclusive (SysEx) status byte. MIDI SysEx commands are typically specific to each manufacturer of MIDI devices. Each manufacturer gets

a SysEx ID number. MSC has a SysEx sub-ID number of 0x02. The syntax, in hexadecimal, for a MSC message is:

```
F0 7F <device_ID> 02 <command_format> <command> <data> F7
```

- `F0` is the status byte indicating the start of a SysEx message.
- `7F` indicates the use of a SysEx sub-ID. This is technically a manufacturer ID that has been reserved to indicate extensions to the MIDI specification.
- `<device_ID>` can be any number between 0x00 and 0x7F indicating the device ID of the thing you want to control. These device ID numbers have to be set on the receiving end as well so each device knows which messages to respond to and which messages to ignore. 0x7F is a universal device ID. All devices respond to this ID regardless of their individual ID numbers.
- `02` is the sub-ID number for MIDI Show Control. This tells the receiving device that the bytes that follow are MIDI Show Control syntax as opposed to MIDI Machine Control, MIDI Time Code, or other commands.
- `<command_format>` is a number indicating the type of device being controlled. For example, 0x01 indicates a lighting device, 0x40 indicates a projection device, and 0x60 indicates a pyrotechnics device. A complete list of command format numbers can be found in the MIDI Show Control 1.1 specification.
- `<command>` is a number indicating the type of command being sent. For example, 0x01 is "GO", 0x02 is "STOP", 0x03 is "RESUME", 0x0A is "RESET".
- `<data>` represents a variable number of data bytes that are required for the type of command being sent. A "GO" command might need some data bytes to indicate the cue number that needs to be executed from a list of cues on the device. If no cue number is specified, the device simply executes the next cue in the list. Two data bytes (0x00, 0x00) are still needed in this case as delimiters for the cue number syntax. Some MSC devices are able to interpret the message without these delimiters, but they're technically required in the MSC specification.
- `F7` is the End of Systems Exclusive byte indicating the end of the message.

Most lighting consoles, projection systems, and computer sound playback systems used in live entertainment are able to generate and respond to MSC messages. Typically, you don't have to create the commands manually in hex. You have a graphical interface that lets you choose the command you're want from a list of menus. Figure 6.41 shows some MSC FIRE commands for a lighting console generated as part of a list of sound cues. In this case, the sound effect of a firecracker is synchronized with the flash of a strobe light using MSC.

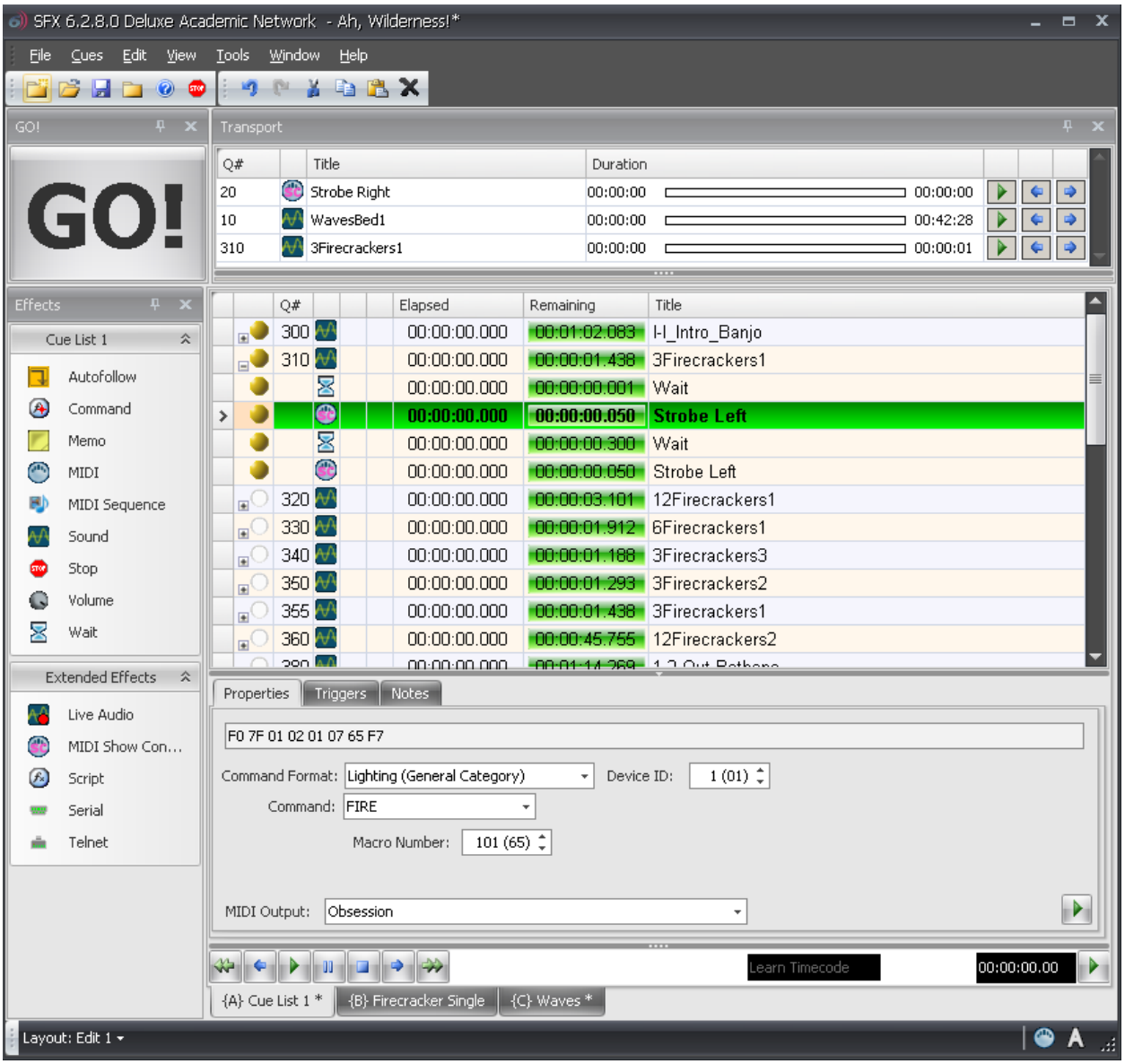


Figure 6.42 A list of sound and MIDI Show Control cues in a sound playback system for live entertainment

6.2.5.2 MIDI Relays and Footswitches

You may not always have a device that knows how to respond to MIDI commands. For example, although there is a MIDI Show Control command format for motorized scenery, the motor that moves a unit on or off the stage doesn't understand MIDI commands. However, it does understand a switch. There are many options available for MIDI controlled relays that respond to a MIDI command by making or breaking an electrical contact closure. This makes it possible for you to connect the wires for the control switch of a motor to the relay output, and then when you send the appropriate MIDI command to the relay, it closes the connection and the motor starts turning.

Another possibility is that you may want to use MIDI commands without a traditional MIDI controller. For example, maybe you want a sound effect to play each time a door is opened

or closed. Using a MIDI footswitch device, you could wire up a magnetic door sensor to the footswitch input and have a MIDI command sent to the computer each time the door is opened or closed. The computer could then be programmed to respond by playing a sound file. Figure 6.42 shows an example of a MIDI relay device.

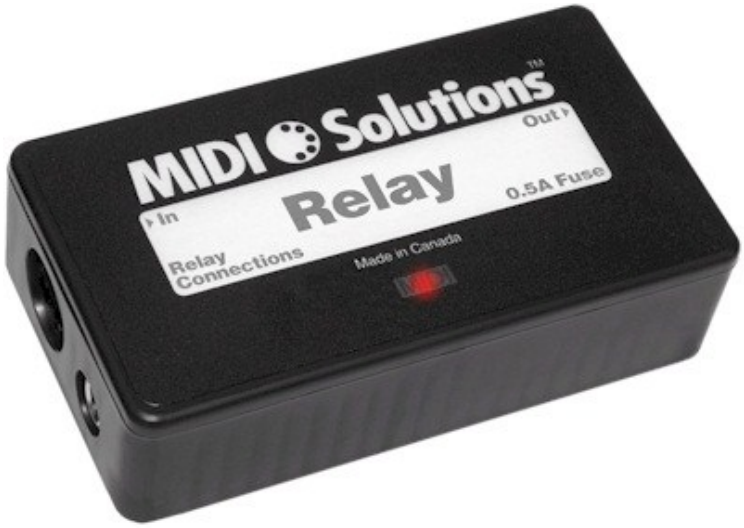


Figure 6.43 A MIDI controllable relay

6.2.5.3 MIDI Time Code

MIDI sequencers, audio editors, and audio playback systems often need to synchronize their timeline with other systems. Synchronization could be needed for working with sound for video, lighting systems in live performance, and even automated theme part attractions. The world of filmmaking has been dealing with synchronization issues for decades, and the Society of Motion Picture and Television Engineers (SMPTE) has developed a standard format for a time code that can be used to keep video and audio in sync. Typically this is accomplished using an audio signal called Linear Time Code that has the synchronization data encoded in SMPTE format. The format is Hours:Minutes:Seconds:Frames. The number of frames per second varies, but the standard allows for 24, 25, 29.97 (also known as 30-Drop), and 30 frames per second.

MIDI Time Code (MTC) uses this same time code format, but instead of being encoded into an analog audio signal, the time information is transmitted in digital format via MIDI. A full MIDI Time Code message has the following syntax:

```
F0 7F <device_ID> <sub-ID 1> <sub-ID 2> <hr> <mn> <sc> <fr> F7
```

- `F0` is the status byte indicating the start of a SysEx message.
- `7F` indicates the use of a SysEx sub-ID. This is technically a manufacturer ID that has been reserved to indicate extensions to the MIDI specification.
- `<device_ID>` can be any number between 0x00 and 0x7F indicating the device ID of the thing you want to control. These device ID numbers have to be set on the receiving end as well so each device knows which messages to respond to and which messages to ignore. 0x7F is a universal device ID. All devices respond to this ID regardless of their individual ID numbers.

- `<sub-ID 1>` is the sub-ID number for MIDI Time Code. This tells the receiving device that the bytes that follow are MIDI Time Code syntax as opposed to MIDI Machine Control, MIDI Show Control, or other commands. There are a few different MIDI Time Code sub ID numbers. 01 is used for full SMPTE messages and for SMPTE user bits messages. 04 is used for a MIDI Cueing message that includes a SMPTE time along with values for markers such as Punch In/Out and Start/Stop points. 05 is used for real-time cuing messages. These messages have all the marker values but use the quarter-frame format for the time code.
- `<sub-ID 2>` is a number used to define the type of message within the sub-ID 1 families. For example, there are two types of Full Messages that use 01 for sub-ID 1. A value of 01 for sub-ID 2 in a Full Message would indicate a Full Time Code Message whereas 02 would indicate a User Bits Message.
- `<hr>` is a byte that carries both the hours value as well as the frame rate. Since there are only 24 hours in a day, the hours value can be encoded using the five least significant bits, while the next two greater significant bits are used for the frame rate. With those two bits you can indicate four different frame rates (24, 25, 29.97, 30). As this is a data byte, the most significant bit stays at 0.
- `<mn>` represents the minutes value 00–59.
- `<sc>` represents the seconds value 00–59.
- <fr> represents the frame value 00–29.
- `F7` is the End of Systems Exclusive byte indicating the end of the message.

For example, to send a message that sets the current timeline location of every device in the system to 01:30:35:20 in 30 frames/second format, the hexadecimal message would be:

```
F0 7F 7F 01 01 61 1E 23 14 F7
```

Once the full message has been transmitted, the time code is sent in quarter-frame format. Quarter-frame messages are much smaller than full messages. This is less demanding of the system since only two bytes rather than ten bytes have to be parsed at a time. Quarter frame messages use the 0xF1 status byte and one data byte with the high nibble indicating the message type and the low nibble indicating the value of the given time field. There are eight message types, two for each time field. Each field is separated into a least significant nibble and a most significant nibble:

0 = Frame count LS

1 = Frame count MS

2 = Seconds count LS

3 = Seconds count MS

4 = Minutes count LS

5 = Minutes count MS

6 = Hours count LS

7 = Hours count MS and SMPTE frame rate

As the name indicates, quarter frame messages are transmitted in increments of four messages per frame. Messages are transmitted in the order listed above. Consequently, the entire SMPTE time is completed every two frames. Let's break down the same 01:30:35:20 time value into quarter frame messages in hexadecimal. This time value would be broken up into eight messages:

`F1 04` (Frame LS)

`F1 11` (Frame MS)

`F1 23` (Seconds LS)

`F1 32` (Seconds MS)

`F1 4E` (Minutes LS)

`F1 51` (Minutes MS)

`F1 61` (Hours LS)

`F1 76` (Hours MS and Frame Rate)

When synchronizing systems with MIDI Time Code, one device needs to be the master time code generator and all other devices need to be configured to follow the time code from this master clock. Figure 6.43 shows Logic Pro configured to synchronize to an incoming MIDI Time Code signal. If you have a mix of devices in your system that follow LTC or MTC, you can also put a dedicated time code device in your system that collects the time code signal in LTC or MTC format and then relay that time code to all the devices in your system in the various formats required, as shown in Figure 6.44.

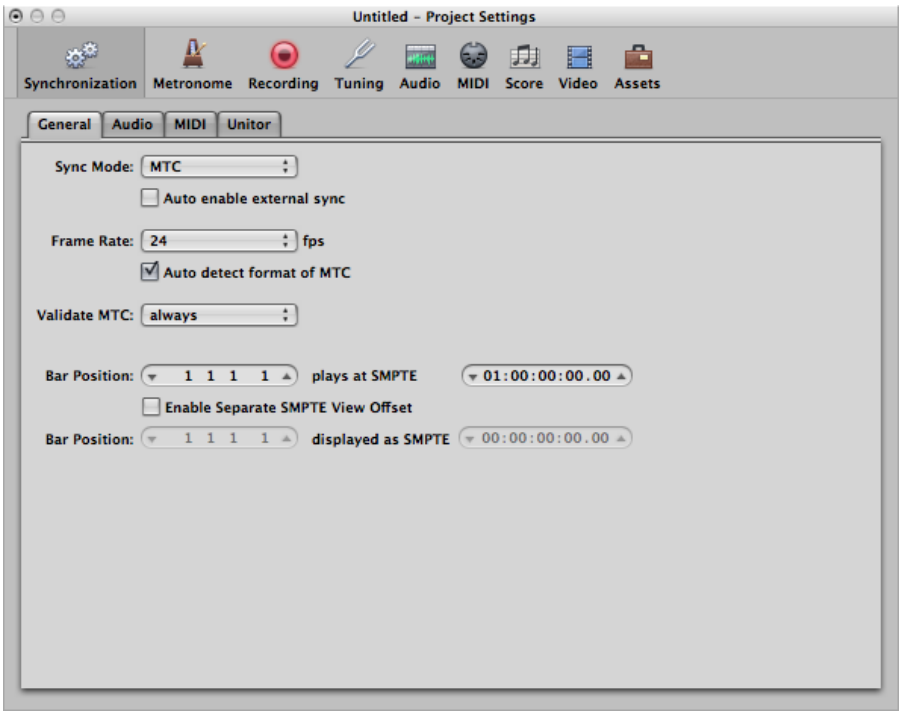


Figure 6.44 MIDI Time Code synchronization for Logic Pro

Figure 6.45 A dedicated SMPTE synchronization device capable of distributing Linear Time Code and MIDI Time Code

6.2.5.4 MIDI Machine Control

Another subset of the MIDI specification is a set of commands that can control the transport system of various recording and playback systems. Transport controls are things like play, stop, rewind, record, etc. This command set is called MIDI Machine Control. The specification is quite comprehensive and includes options for SMPTE time code values, as well as confirmation response messages from the devices being controlled. A simple MMC message has the following syntax:

```
F0 7F <device_ID> 06 <command> F7
```

- `F0` is the status byte indicating the start of a SysEx message.
- `7F` indicates the use of a SysEx sub-ID. This is technically a manufacturer ID that has been reserved to indicate extensions to the MIDI specification.
- `<device_ID>` can be any number between 0x00 and 0x7F indicating the device ID of the thing you want to control. These device ID numbers have to be set on the receiving end as well so each device knows which messages to respond to and which messages to

ignore. 0x7F is a universal device ID. All devices respond to this ID regardless of their individual ID numbers.

- `06` is the sub-ID number for MIDI Machine Control. This tells the receiving device that the bytes that follow are MIDI Machine Control syntax as opposed to MIDI Show Control, MIDI Time Code, or other commands.
- `<command>` can be a set of bytes as small as one byte and can include several bytes communicating various commands in great detail. A simple play (0x02) or stop (0x01) command only requires a single data byte.
- `F7` is the End of Systems Exclusive byte indicating the end of the message.

A MMC command for PLAY would look like this:

```
F0 7F 7F 06 02 F7
```

MIDI Machine Control was really a necessity when most studio recording systems were made up of several different magnetic tape-based systems or dedicated hard disc recorders. In these situations, a single transport control would make sure all the devices were doing the same thing. In today's software-based systems, MIDI Machine Control is primarily used with MIDI control surfaces, like the one shown in Figure 6.45, that connect to a computer so you can control the transport of your DAW software without having to manipulate a mouse.

Figure 6.46 A dedicated MIDI Machine Control device

6.3 Science, Mathematics, and Algorithms

6.3.1 MIDI SMF Files

If you'd like to dig into the MIDI specification more deeply – perhaps writing a program that can either generate a MIDI file in the correct format or interpret one and turn it into digital audio – you need to know more about how the files are formatted.

Standard MIDI Files (**SMF**), with the *.mid* or *.smf* suffix, encode MIDI data in a prescribed format for the header, timing information, and events. Format 0 files are for single

tracks, Format1 files are for multiple tracks, and Format 2 files are for multiple tracks where a separate song performance can be represented. (Format 2 is not widely used.) SMF files are platform-independent, interpretable on PCs, MAC, and Linux machines.

Blocks of information in SMF files are called **chunks**. The first chunk is the **header chunk**, followed by one or more **data chunks**. The header and data chunks begin with four bytes (a four-character string) identifying the type of chunk they are. The header chunk then has four bytes giving the length of the remaining fields of the chunk (which is always 6 for the header), two bytes telling the format (MIDI 0, 1, or 2), two bytes telling the number of tracks, and two bytes with information about how timing is handled.

Data are stored in **track chunks.** A track chunk also begins with four bytes telling the type of chunk followed by four bytes telling how much data is in the chunk. The data then follow. The bytes which constitute the data are track events: either regular MIDI events like Note On; meta-events like changes of tempo, key, or time signature; or sys-ex events. The events begin with a timestamp telling when they are to happen. Then the rest of the data are MIDI events with the format described in Section 1. The structure of an SMF file is illustrated in Figure 6.47.

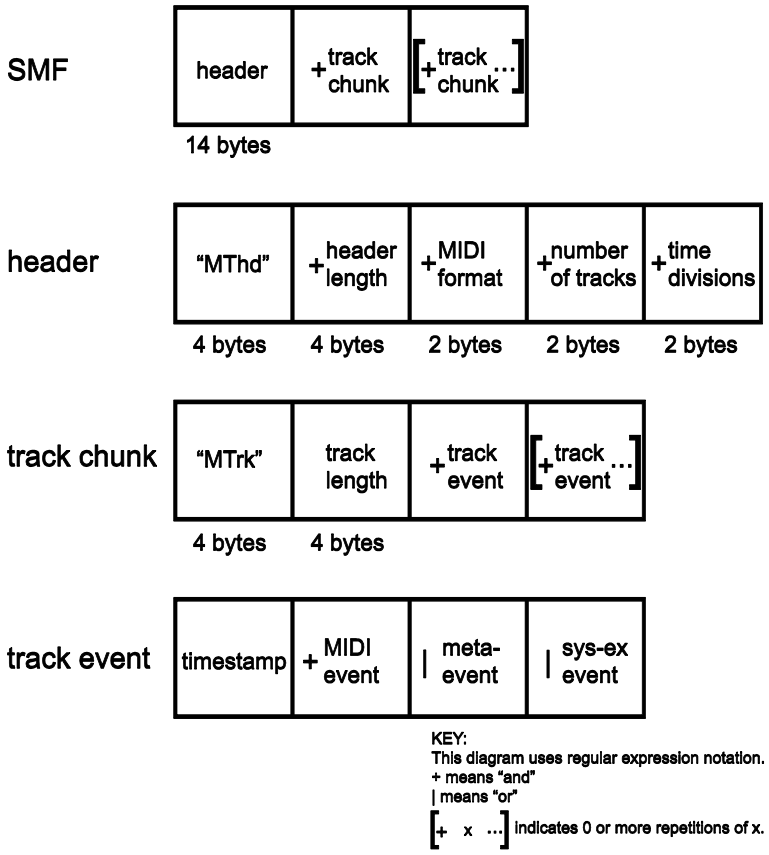


Figure 6.47 SMF file structure

MIDI event **timestamps** tell the change in time between the previous event and the current one, using the tick-per-beat, frame rate in frames/s, and tick/frame defined in the header. The timestamp itself is given in a variable-length field. To accomplish this, the first bit of each byte in the timestamp indicates how many bytes are to follow in the timestamp. If the bit is a 0, then

the value in the following seven bits of the byte make up the full value of the timestamp. If the bit is a 1, then the next byte is also to be considered part of the timestamp value. This ultimately saves space. It would be wasteful to dedicate four bytes to the timestamp just to take care of the few cases where there is a long pause between one MIDI event and the next one.

An important consideration in writing or reading SMF files is the issue of whether bytes are stored in little-endian or big-endian format. In **big-endian format**, the most significant byte is stored first in a sequence of bytes that make up one value. In **little-endian**, the least significant byte is stored first. SMF files store bytes in big-endian format. If you're writing an SMF-interpreting program, you need to check the endian-ness of the processor and operating system on which you'll be running the program. A PC/Windows combination is generally little-endian, so a program running on that platform has to swap the byte-order when determining the value of a multiple-byte timestamp.

More details about SMF files can be found at www.midi.org. To see the full MIDI specification, you have to order and pay for the documentation. (Messick 1998) is a good source to help you write a C++ program that reads and interprets SMF files.

6.3.2 Shaping Synthesizer Parameters with Envelopes and LFOs

Let's make a sharp turn now from MIDI specifications to the mathematics and algorithms under the hood of synthesizers.

In Section 6.1.8.7, envelopes were discussed as a way of modifying the parameters of some synthesizer function – for example, the cutoff frequency of a low or high pass filter or the amplitude of a waveform. The mathematics of envelopes is easy to understand. The graph of the envelope shows time on the horizontal axis and a "multiplier" or coefficient on the vertical axis. The parameter is question is simply multiplied by the coefficient over time.

Envelopes can be generated by simple or complex functions. The envelope could be a simple sinusoidal, triangle, square, or sawtooth function that causes the parameter to go up and down in this regular pattern. In such cases, the envelope is called an *oscillator*. The term low-frequency oscillator (LFO) is used in synthesizers because the rate at which the parameter is caused to change is low compared to audible frequencies.

An ADSR envelope has a shape like the one shown in Figure 6.25. Such an envelope can be defined by the attack, decay, sustain, and release points, between which straight (or evenly curved) lines are drawn. Again, the values in the graph represent multipliers to be applied to a chosen parameter.

The exercise associated with this section invites you to modulate one or more of the parameters of an audio signal with an LFO and also with an ASDR envelope that you define yourself.

6.3.3 Type of Synthesis

6.3.3.1 Table-Lookup Oscillators and Wavetable Synthesis

We have seen how single-frequency sound waves are easily generated by means of sinusoidal functions. In our example exercises, we've done this through computation, evaluating sine functions over time. In contrast, **table-lookup oscillators** generate waveforms by means of a set

of look-up wavetables stored in contiguous memory locations. Each **wavetable** contains a list of sample values constituting one cycle of a sinusoidal wave, as illustrated in Figure 6.48. Multiple wavetables are stored so that waveforms of a wide range of frequencies can be generated.

1	0.0490676743274180
2	0.098017140329561
3	0.146730474455362
…	
128	-2.449293598294706e-16

wavetable

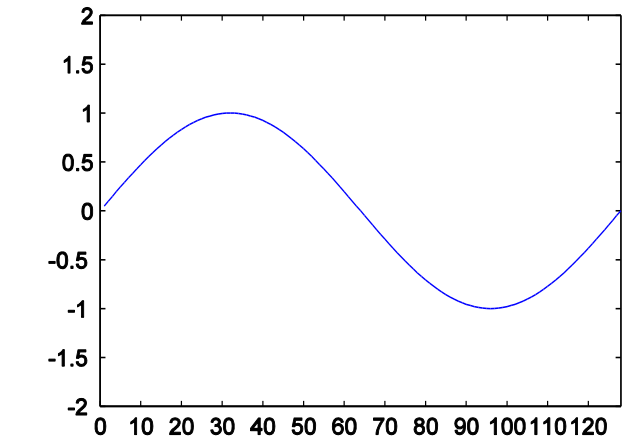


Figure 6.48 A wavetable in a table-lookup oscillator

With a table-lookup oscillator, a waveform is created by advancing a pointer through a wavetable, reading the values, cycling back to the beginning of the table as necessary, and outputting the sound wave accordingly.

With a table of *N* samples representing one cycle of a waveform and an assumed sampling rate of *r* samples/s, you can generate a fundamental frequency of *r/N Hz* simply by reading the values out of the table at the sampling rate. This entails stepping through the indexes of the consecutive memory locations of the table. The wavetable in Figure 6.48 corresponds to a fundamental frequency of $\frac{48000}{128} = 375\ Hz$.

Harmonics of the fundamental frequency of the wavetable can be created by skipping values or inserting extra values in between those in the table. For example, you can output a waveform with twice the frequency of the fundamental by reading out every other value in the table. You can output a waveform with ½ the frequency of the fundamental by reading each value twice, or by inserting values in between those in the table by interpolation.

The phase of the waveform can be varied by starting at an offset from the beginning of the wavetable. To start at a phase offset of $p\pi$ radians, you would start reading at index $\frac{pN}{2}$. For example, to start at an offset of $\pi/2$ in the wavetable of Figure 6.48, you would start at index $\frac{\frac{1}{2}*128}{2} = 32$.

To generate a waveform that is not a harmonic of the fundamental frequency, it's necessary to add an increment to the consecutive indexes that are read out from the table. This increment *i* depends on the desired frequency *f*, the table length *N*, and the sampling rate *r*, defined by $i = \frac{f*N}{r}$. For example, to generate a waveform with frequency 750 Hz using the wavetable of Figure 6.48 and assuming a sampling rate of 48000 Hz, you would need an increment of $i = \frac{750*128}{48000} = 2$. We've chosen an example where the increment is an integer, which is good because the indexes into the table have to be integers.

What if you wanted a frequency of 390 Hz? Then the increment would be $i = \frac{390*128}{48000} = 1.04$, which is not an integer. In cases where the increment is not an integer, interpolation must be used. For example, if you want to go an increment of 1.04 from index 1, that would take you to index 2.04. Assuming that our wavetable is called *table*, you want a value equal to $table[2] + 0.04 * (table[3] - table[2])$. This is a rough way to do interpolation. Cubic spline interpolation can also be used as a better way of shaping the curve of the waveform. The exercise associated with this section suggests that you experiment with table-lookup oscillators in MATLAB.

An extension of the use of table-lookup oscillators is wavetable synthesis. Wavetable synthesis was introduced in digital synthesizers in the 1970s by Wolfgang Palm in Germany. This was the era when the transition was being made from the analog to the digital realm. **Wavetable synthesis** uses multiple wavetables, combining them with additive synthesis and crossfading and shaping them with modulators, filters, and amplitude envelopes. The wavetables don't necessarily have to represent simple sinusoidals but can be more complex waveforms. Wavetable synthesis was innovative in the 1970s in allowing for the creation of sounds not realizable with by solely analog means. This synthesis method has now evolved to the NWave-Waldorf synthesizer for the iPad.

> **Aside:** The term "wavetable" is sometimes used to refer a memory bank of samples used by sound cards for MIDI sound generation. This can be misleading terminology, as wavetable synthesis is a different thing entirely.

6.3.3.2 Additive Synthesis

In Chapter 2, we introduced the concept of frequency components of complex waves. This is one of the most fundamental concepts in audio processing, dating back to the groundbreaking work of Jean-Baptiste Fourier in the early 1800s. Fourier was able to prove that any periodic waveform is composed of an infinite sum of single-frequency waveforms of varying frequencies and amplitudes. The single-frequency waveforms that are summed to make the more complex one are called the **frequency components**.

The implications of Fourier's discovery are far reaching. It means that, theoretically, we can build whatever complex sounds we want just by adding sine waves. This is the basis of additive synthesis. We demonstrated how it worked in Chapter 2, illustrated by the production of square, sawtooth, and triangle waveforms. Additive synthesis of each of these waveforms begins with a sine wave of some fundamental frequency, *f*. As you recall, a square wave is constructed from an infinite sum of odd-numbered harmonics of *f* of diminishing amplitude, as in

$$A\sin(2\pi ft) + \frac{A}{3}sin(6\pi ft) + \frac{A}{5}sin(10\pi ft) + \frac{A}{7}sin(14\pi ft) + \frac{A}{9}sin(18\pi ft) + \cdots$$

A sawtooth waveform can be constructed from an infinite sum of all harmonics of *f* of diminishing amplitude, as in

$$\frac{2}{\pi}\left(A\sin(2\pi ft) + \frac{A}{2}sin(4\pi ft) + \frac{A}{3}sin(6\pi ft) + \frac{A}{4}sin(8\pi ft) + \frac{A}{5}sin(10\pi ft) + \cdots\right)$$

A triangle waveform can be constructed from an infinite sum of odd-numbered harmonics of *f* that diminish in amplitude and vary in their sign, as in

$$\frac{8}{\pi^2}\left(\mathrm{A}\sin(2\pi f) - \frac{A}{3^2}sin(6\pi ft) + \frac{A}{5^2}sin(10\pi ft) - \frac{A}{7^2}sin(14\pi ft) + \frac{A}{9^2}sin(18\pi ft) - \frac{A}{11^2}sin(22\pi ft) + \cdots\right)$$

These basic waveforms turn out to be very important in subtractive synthesis, as they serve as a starting point from which other more complex sounds can be created.

To be able to create a sound by additive synthesis, you need to know the frequency components to add together. It's usually difficult to get the sound you want by adding waveforms from the ground up. In turns out that subtractive synthesis is often an easier way to proceed.

6.3.3.3 Subtractive Synthesis

The first synthesizers, including the Moog and Buchla's Music Box, were analog synthesizers that made distinctive electronic sounds different from what is produced by traditional instruments. This was part of the fascination that listeners had for them. They did this by **subtractive synthesis**, a process that begins with a basic sound and then selectively removes frequency components. The first digital synthesizers imitated their analog precursors. Thus, when people speak of "analog synthesizers" today, they often mean digital subtractive synthesizers. The Subtractor Polyphonic Synthesizer shown in Figure 6.19 is an example of one of these.

The development of subtractive synthesis arose from an analysis of musical instruments and the way they create their sound, the human voice being among those instruments. Such sounds can be divided into two components: a source of excitation and a resonator. For a violin, the source is the bow being drawn across the string, and the resonator is the body of the violin. For the human voice, the source results from air movement and muscle contractions of the vocal chords, and the resonator is the mouth. In a subtractive synthesizer, the source could be a pulse, sawtooth, or triangle wave or random noise of different colors (colors corresponding to how the noise is spread out over the frequency spectrum). Frequently, preset patches are provided, which are basic waveforms with certain settings like amplitude envelopes already applied (another usage of the term *patch*). Filters are provided that allow you to remove selected frequency components. For example, you could start with a sawtooth wave and filter out some of the higher harmonic frequencies, creating something that sounds fairly similar to a stringed instrument. An amplitude envelope could be applied also to shape the attack, decay, sustain, and release of the sound.

The exercise suggests that you experiment with subtractive synthesis in C++ by beginning with a waveform, subtracting some of its frequency components, and applying an envelope.

6.3.3.4 Amplitude Modulation (AM)

Amplitude, phase, and frequency modulation are three types of modulation that can be applied to synthesize sounds in a digital synthesizer. We explain the mathematical operations below. In Section 0, we defined **modulation** as the process of changing the shape of a waveform over time. Modulation has long been used in analog telecommunication systems as a way to transmit a signal on a fixed frequency channel. The frequency on which a television or radio station is

broadcast is referred to as the **carrier signal** and the message "written on" the carrier is called the **modulator signal**. The message can be encoded on the carrier signal in one of three ways: **AM** (**amplitude modulation**), **PM** (**phase modulation**), or **FM** (**frequency modulation**).

Amplitude modulation (**AM**) is commonly used in radio transmissions. It entails sending a message by modulating the amplitude of a carrier signal with a modulator signal.

In the realm of digital sound as created by synthesizers, AM can be used to generate a digital audio signal of *N* samples by application of the following equation:

$$a(n) = \sin(\omega_c n/r) * \left(1.0 + A\cos(\omega_m n/r)\right)$$
$$for\ 0 \leq n \leq N-1$$
$$where\ N\ is\ the\ number\ of\ samples,$$
$$\omega_c\ is\ the\ angular\ frequency\ of\ the\ carrier\ signal,$$
$$\omega_m\ is\ the\ angular\ frequency\ of\ the\ modulator\ signal,$$
$$r\ is\ the\ sampling\ rate$$
$$and\ A\ is\ amplitude$$

Equation 6.1 Amplitude modulation for digital synthesis

The process is expressed algorithmically in Algorithm 6.8. The algorithm shows that the AM synthesis equation must be applied to generate each of the samples for $1 \leq t \leq N$.

```
algorithm amplitude_modulation
/*
Input:
      f_c, the frequency of the carrier signal
      f_m, the frequency of a low frequency modulator signal
      N, the number of samples you want to create
      r, the sampling rate
      A, to adjust the amplitude of the
Output:
      y, an array of audio samples where the carrier has been amplitude
modulated by the modulator */
{
      for (n = 1 to N)
            y[n] = sin(2*pi*f_c*n/r) * (1.0 + A*cos(2*pi*f_m*n/r));
}
```

Algorithm 6.1 Amplitude modulation for digital synthesis

Algorithm 6.1can be executed at the MATLAB command line with the statements below, generating the graphs is Figure 6.49. Because MATLAB executes the statements as vector operations, a loop is not required. (Alternatively, a MATLAB program could be written using a loop.) For simplicity, we'll assume $A = 1$ in what follows.

```
N = 44100;
r = 44100;
n = [1:N];
f_m = 10;
f_c = 440;
m = cos(2*pi*f_m*n/r);
c = sin(2*pi*f_c*n/r);
figure;
```

```
AM = c.*(1.0 + m);
plot(m(1:10000));
axis([0 10000 -2 2]);
figure;
plot(c(1:10000));
axis([0 10000 -2 2]);
figure;
plot(AM(1:10000));
axis([0 10000 -2 2]);
wavplay(c, 44100);
wavplay(m, 44100);
wavplay(AM, 44100);
```

This yields the following graphs:

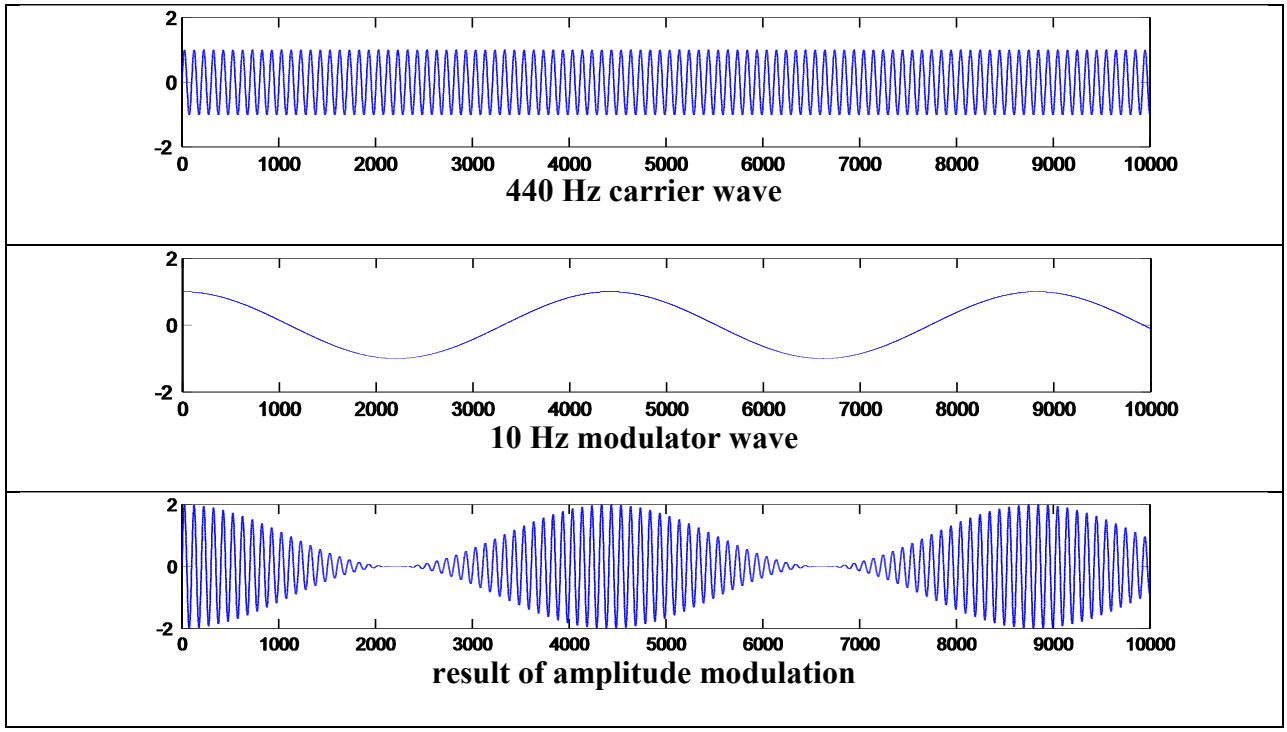


Figure 6.49 Amplitude modulation using two sinusoidals

If you listen to the result, you'll see that amplitude modulation creates a kind of tremolo effect.

The same process can be accomplished with more complex waveforms. *HornsE04.wav* is a 132,300-sample audio clip of horns playing, at a sampling rate of 44,100 Hz. Below, we shape it with a 440 Hz cosine wave (Figure 6.50).

```
N = 132300;
r = 44100;
c = wavread('HornsE04.wav');
n = [1:N];
m = sin(2*pi*10*n/r);
m = transpose(m);
AM2 = c .* (1.0 + m);
figure;
plot(c);
```

```
axis([0 132300 -2 2]);
figure;
plot(m);
axis([0 132300 -2 2]);
figure;
plot(AM2);
axis([0 132300 -2 2]);
wavplay(c, 44100);
wavplay(m, 44100);
wavplay(AM2, 44100);
```

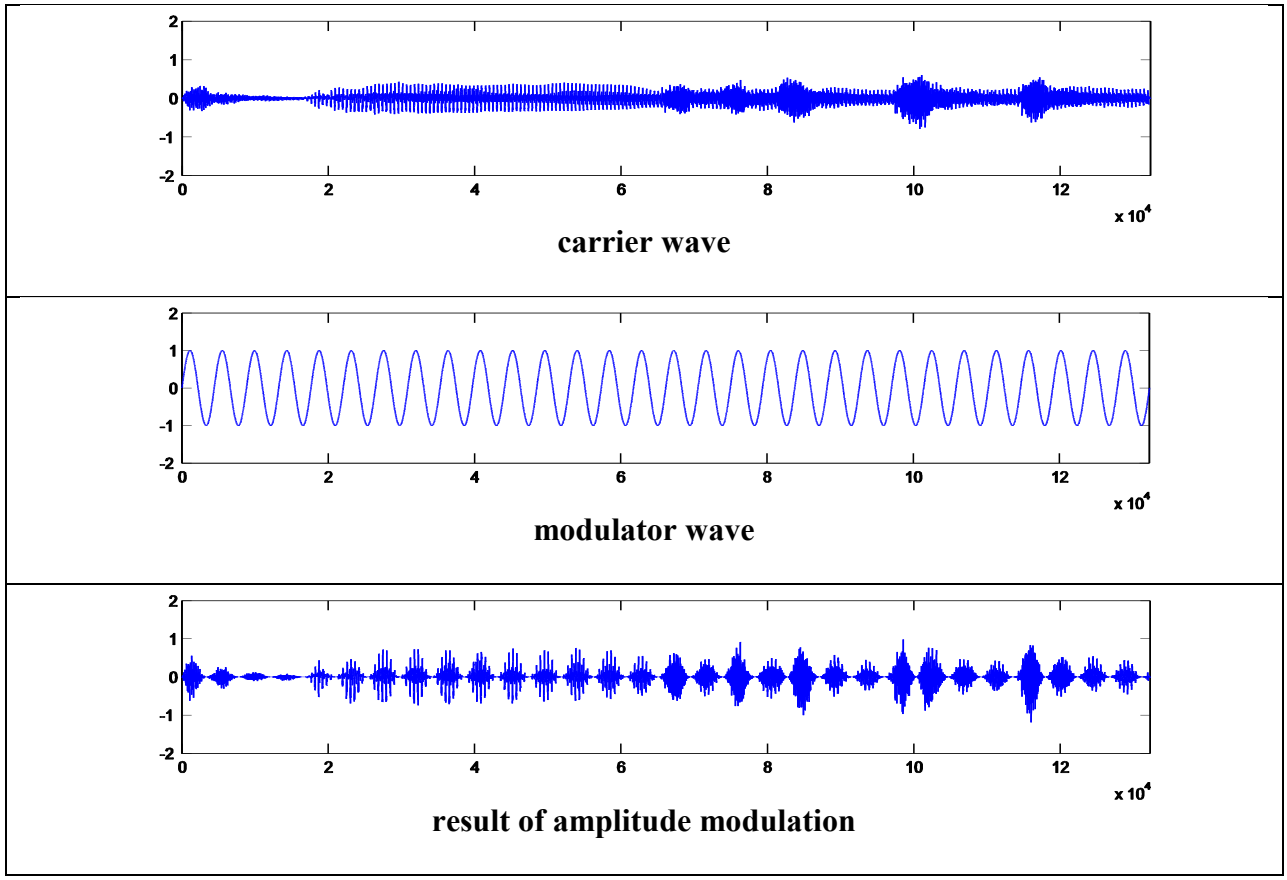


Figure 6.50 Amplitude modulation of a complex audio signal

The audio effect is different, depending on which signal is chosen as the carrier and which as the modulator. Below, the carrier and modulator are reversed from the previous example, generating the graphs in Figure 6.51.

```
N = 132300;
r = 44100;
m = wavread('HornsE04.wav');
n = [1:N];
c = sin(2*pi*10*n/r);
c = transpose(c);
AM3 = c .* (1.0 + m);
figure;
plot(c);
```

```
axis([0 132300 -2 2]);
figure;
plot(m);
axis([0 132300 -2 2]);
figure;
plot(AM3);
axis([0 132300 -2 2]);
wavplay(c, 44100);
wavplay(m, 44100);
wavplay(AM3, 44100);
```

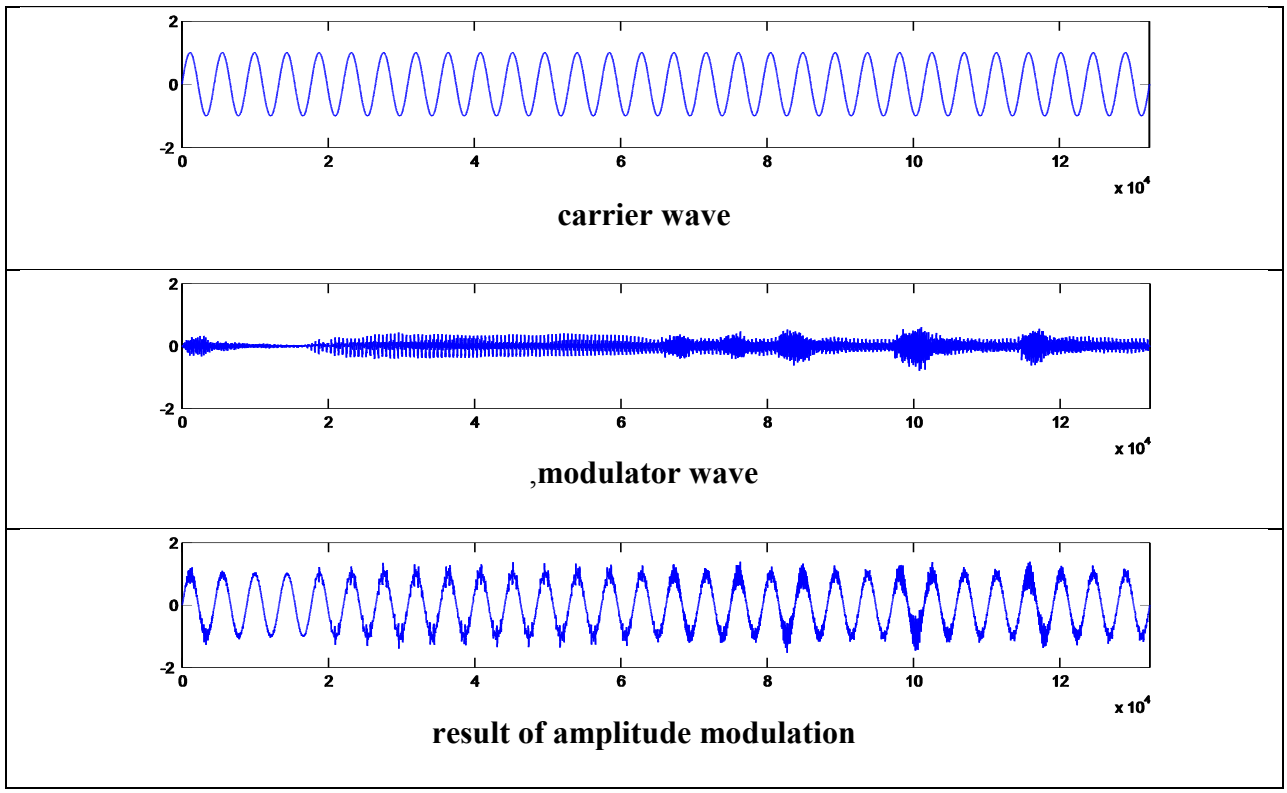


Figure 6.51 Amplitude modulation of a complex audio signal with carrier and modulator waves reversed

Amplitude modulation produces new frequency components in the resulting waveform at $f_c + f_m$ and $f_c - f_m$, where f_c is the frequency of the carrier and f_m is the frequency of the modulator. These are called **sidebands**. You can verify that the sidebands are at $f_c + f_m$ and $f_c - f_m$ with a little math based on the properties of sines and cosines.

$$\cos(2\pi f_c n)\,(1.0 + \cos(2\pi f_m n)) =$$
$$\cos(2\pi f_c n) + \cos(2\pi f_c n)\cos(2\pi f_m n) =$$
$$\cos(2\pi f_c n) + \frac{1}{2}\cos(2\pi(f_c + f_m)\,n) + \frac{1}{2}\cos(2\pi(f_c - f_m)\,n)$$

(The third step comes from the cosine product rule.) This derivation shows that there are three frequency components: one with frequency f_c, a second with frequency $f_c + f_m$, and a third with frequency $f_c - f_m$.

To verify this with an example, you can generate a graph of the sidebands in MATLAB by doing a Fourier transfer of the waveform generated by AM and plotting the magnitudes of the frequency components. MATLAB's *fft* function does a Fourier transform of a vector of audio data, returning a vector of complex numbers. The *abs* function turns the complex numbers into a vector of magnitudes of frequency components. Then these values can be plotted with the *plot* function. We show the graph only from frequencies 1 through 600 Hz, since the only frequency components for this example lie in this range. Figure 6.52 shows the sidebands corresponding the AM performed in Figure 6.49. The sidebands are at 450 Hz and 460 Hz, as predicted.

```
figure;
fftmag = abs(fft(AM));
plot(fftmag(1:600));
```

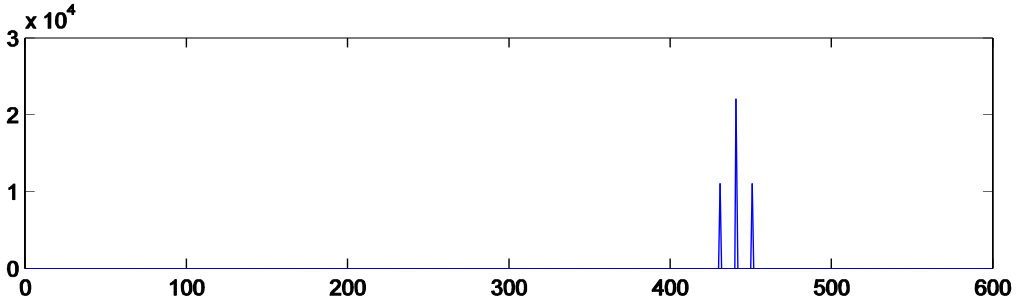


Figure 6.52 Frequency components after amplitude modulation in Figure 6.49

6.3.3.5 Ring Modulation

Ring modulation entails simply multiplying two signals. To create a digital signal using ring modulation, the Equation 6.2 can be applied.

$$r(n) = A_1 \sin(\omega_1 n/r) * A_2 sin(\omega_2 n/r)$$
$$for\ 0 \leq n \leq N-1$$
$$where\ N\ is\ the\ number\ of\ samples,$$
$$r\ is\ the\ sampling\ rate,$$
$$where\ \omega_1\ and\ \omega_2\ are\ the\ angular\ frequencies\ of\ two\ signals, and$$
$$A_1\ and\ A_2\ are\ their\ respective\ amplitudes$$

Equation 6.2 Ring modulation for digital synthesis

Since multiplication is commutative, there's no sense in which one signal is the carrier and the other the modulator. Ring modulation is illustrated with two simple sine waves in Figure 6.53. The ring modulated waveform is generated with the MATLAB commands below. Again, we set amplitudes to 1.

```
N = 44100;
r = 44100;
n = [1:N];
w1 = 440;
w2 = 10;
rm = sin(2*pi*w1*n/r) .* cos(2*pi*w2*n/r);
plot(rm(1:10000));
axis([1 10000 -2 2]);
```

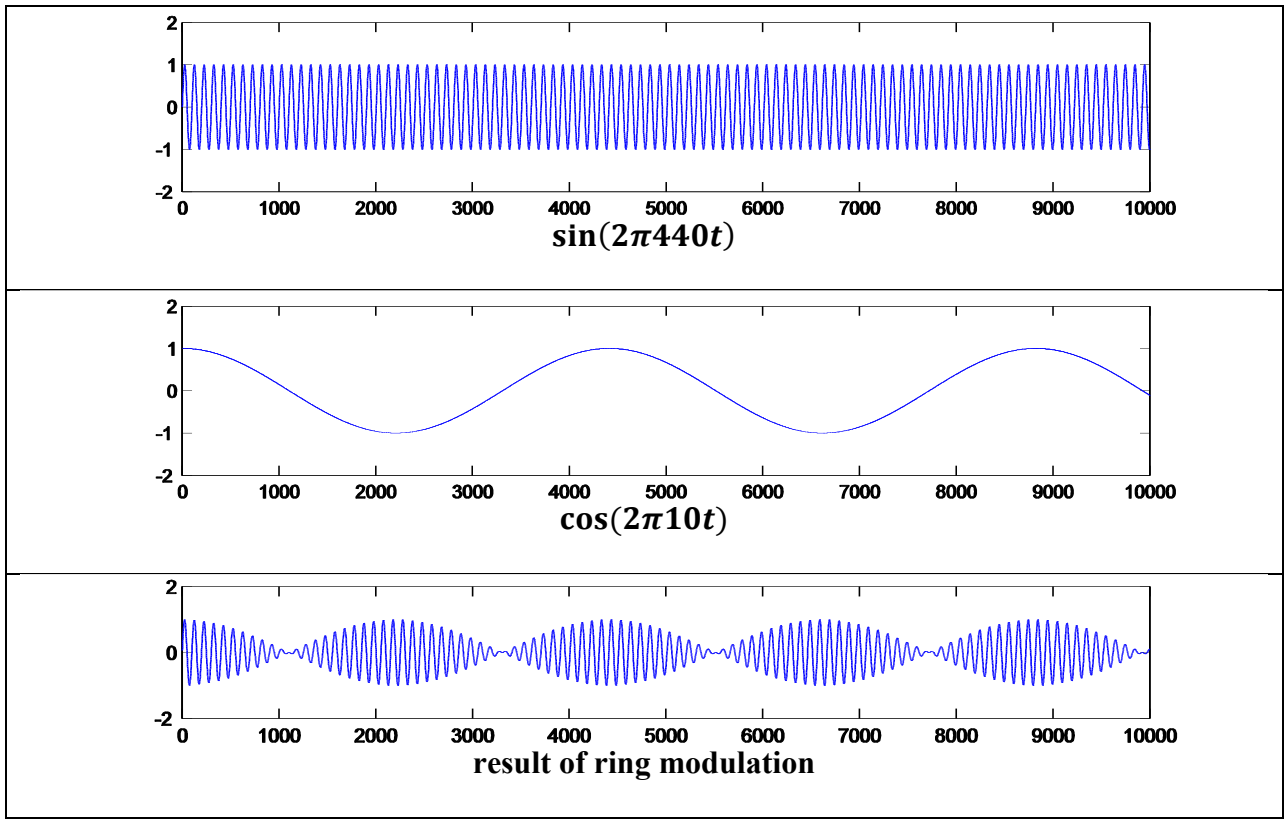


Figure 6.53 Ring modulation using two sinusoidals

6.3.3.6 Phase Modulation (PM)

Even more interesting audio effects can be created with phase (PM) and frequency modulation (FM). We're all familiar with FM radio, which is based on sending a signal by frequency modulation. Phase modulation is not used extensively in radio transmissions because it can be ambiguous to interpret at the receiving end, but it turns out to be fairly easy to implement PM in digital synthesizers. Some hardware-based synthesizers that are commonly referred to as FM actually use PM synthesis internally – the Yamaha DX series, for example.

Recall that the general equation for a cosine waveform is $Acos(2\pi fn + \phi)$ where f is the frequency and ϕ is the phase. Phase modulation involves changing the phase over time. Equation 6.3 uses phase modulation to generate a digital signal.

$$p(t) = Acos\left(\omega_c n/r + Isin(\omega_m n/r)\right)$$
$$for\ 0 \le n \le N-1$$
$$where\ N\ is\ the\ number\ of\ samples,$$
$$\omega_c\ is\ the\ angular\ frequency\ of\ the\ carrier\ signal,$$
$$\omega_m\ is\ the\ angular\ frequency\ of\ the\ modulator\ signal,$$
$$r\ is\ the\ sampling\ rate$$
$$and\ A\ is\ amplitude$$

Equation 6.3 Phase modulation for digital synthesis

Phase modulation is demonstrated in MATLAB with the following statements:

```
N = 44100;
r = 44100;
n = [1:N];
f_m = 10;
f_c = 440;
w_m = 2 * pi * f_m;
w_c = 2 * pi * f_c;
A = 1;
I = 1;
p = A*cos(w_c * n/r + I*sin(w_m * n/r));
plot(p);
axis([1 30000 -2 2]);
wavplay(p, 44100);
```

The result is graphed in Figure 6.54.

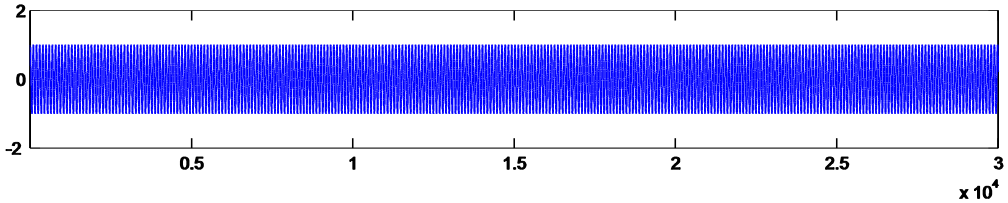


Figure 6.54 Phase modulation using two sinusoidals, where $\omega_c = 2\pi 440$ and $\omega_m = 2\pi 10$

The frequency components shown in Figure 6.55 are plotted in MATLAB with

```
fp = fft2(p);
figure;
plot(abs(fp));
axis([400 480 0 18000]);
```

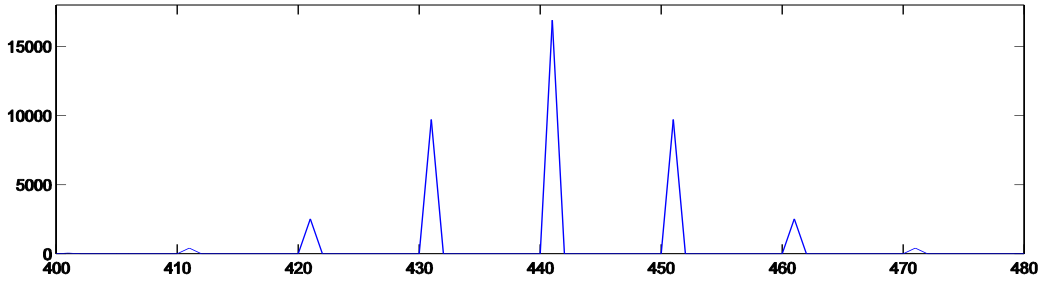


Figure 6.55 Frequency components from phase modulation in Figure 6.54

Phase modulation produces an infinite number of sidebands (many of whose amplitudes are too small to be detected). This fact is expressed in Equation 6.4.

$$cos\big(\omega_c n + Isin(\omega_m n)\big) = \sum_{k=-\infty}^{\infty} J_k(I)\cos([\omega_c + k\omega_m]n)$$

Equation 6.4 Phase modulation equivalence with additive synthesis

$J_k(I)$ gives the amplitude of the frequency component for each k^{th} component in the phase-modulated signal. These scaling functions $J_k(I)$ are called *Bessel functions of the first kind*. It's beyond the scope of the book to define these functions further. You can experiment for yourself to see that the frequency components have amplitudes that depend on *I*. If you listen to the sounds created, you'll find that the timbres of the sounds can also be caused to change over time by changing *I*. The frequencies of the components, on the other hand, depend on the ratio of ω_c/ω_m. You can try varying the MATLAB commands above to experience the wide variety of sounds that can be created with phase modulation. You should also consider the possibilities of applying additive or subtractive synthesis to multiple phase-modulated waveforms.

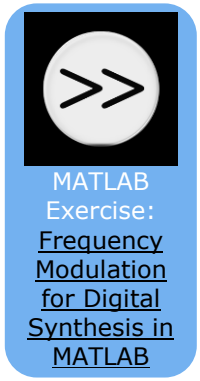
6.3.3.7 Frequency Modulation (FM)

We have seen in the previous section that phase modulation can be applied to the digital synthesis of a wide variety of waveforms. Frequency modulation is equally versatile and frequently used in digital synthesizers. Frequency modulation is defined recursively as follows:

$$f(n) = A\cos(p(n)) \text{ and}$$
$$p(n) = p(n-1) + \frac{\omega_c}{r} + \left(\frac{I\omega_m}{r} * cos\left(\frac{n*\omega_m}{r}\right)\right),$$
$$for\ 1 \le n \le N-1, \text{ and}$$
$$p(0) = \frac{\omega_c}{r} + \frac{I\omega_m}{r},$$
$$where\ N\ is\ the\ number\ of\ samples,$$
$$\omega_c\ is\ the\ angular\ frequency\ of\ the\ carrier\ signal,$$
$$\omega_m\ is\ the\ angular\ frequency\ of\ the\ modulator\ signal,$$
$$r\ is\ the\ sampling\ rate,$$
$$I\ is\ the\ index\ of\ modulation,$$
$$and\ A\ is\ amplitude$$

Equation 6.5 Frequency modulation for digital synthesis

Frequency modulation can yield results identical to phase modulation, depending on how inputs parameters are handled in the implementation. A difference between phase and frequency modulation is the perspective from which the modulation is handled. Obviously, the former is shaping a waveform by modulating the phase, while the latter is modulating the frequency. In frequency modulation, the change in the frequency can be handled by a parameter *d*, an absolute change in carrier signal frequency, which is defined by $d = If_m$. The input parameters $N = 44100$, $r = 44100$, $f_c = 880$, $f_m = 10$, $A = 1$, and $d = 100$ yield the graphs shown in Figure 6.56 and Figure 6.57. We suggest that you try to replicate these results by writing a MATLAB program based on Equation 6.5 defining frequency modulation.



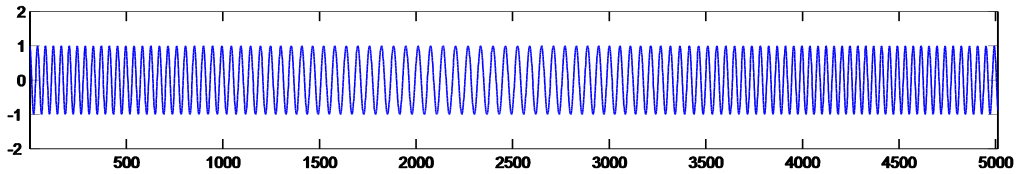


Figure 6.56 Frequency modulation using two sinusoidals,

$$\text{where } \omega_c = 2\pi 880 \text{ and } \omega_m = 2\pi 10$$

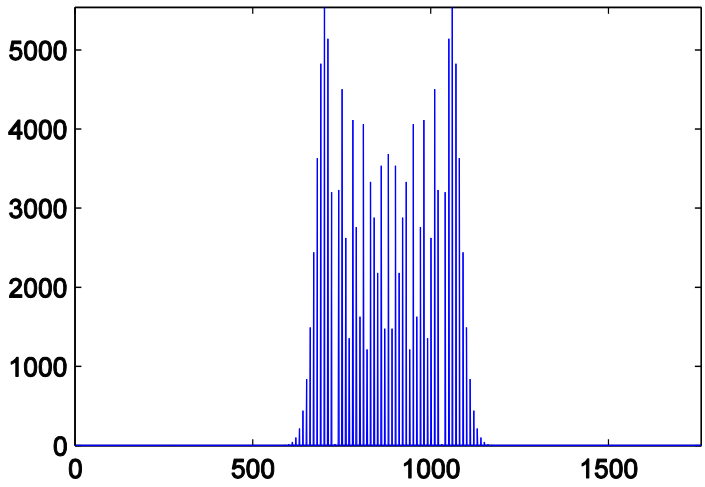


Figure 6.57 Frequency components after frequency modulation in Figure 6.56

6.3.4 Creating A Block Synthesizer in Max

Earlier in this chapter, we discussed the fundamentals of synthesizers and the types of controls and processing components you might find built in to their function. In Section 6.1 and in several of the practical exercises and demonstrations, we took you through these components and features one by one. Breaking apart a synthesizer into these individual blocks makes it easier to understand the signal flow and steps behind audio synthesis. In the book, we generally look at software synthesizers in the examples due to their flexibility and availability, most of which come with a full feature set of oscillators, amplifiers, filters, envelopes, and other synthesis objects. Back in the early days of audio synthesis, however, most of this functionality was delegated to separate, dedicated devices. The oscillator device only generated tones. If you also needed a noise signal, that might require you to find a separate piece of hardware. If you wanted to filter the signal, you would need to acquire a separate audio filtering device. These devices were all connected by a myriad of audio patch cables, which could quickly become a complete jumble of wires. The beauty of hardware synthesis was its modularity. There were no strict rules on what wire must plug into what jack. Where a software synth today might have one or two filters at specific locations in the signal path, there was nothing to stop someone back then from connecting a dozen filters in a row if they wanted to and had the available resources. While the modularity and expanse of these setups may seem daunting, it allowed for maximum flexibility and creativity, and could result in some interesting, unique, and possibly unexpected synthesized sounds.

Wouldn't it be fun to recreate this in the digital world as software? Certainly there are some advantages in this, such as reduced cost, potentially greater flexibility and control, a less cluttered tabletop, not to mention less risk of electrocuting yourself or frying a circuit board. The remainder of this section takes you through the creation of several of these synthesis blocks in MAX, which lends itself quite well to the concept of a creating and using a modular synthesizer.

At the end of this example, we'll see how we can then combine and connect these blocks in any imaginable configuration to possibly create some never-before-heard sounds.

It is assumed that you're proficient in the basics of MAX. If you're not familiar with an object we use or discuss, please refer to the MAX help and reference files for that object (Help > Open [ObjectName] Help in the file menu, or Right-click > Open [ObjectName] Help). Every object in MAX has a help file that explains and demonstrates how the object works. Additionally, if you hover the mouse over an inlet or outlet of an object, MAX often shows you a tooltip with information on its format or use. We also try to include helpful comments and hints within the solution itself.

At this point in the chapter, you should be familiar with a good number of these synthesis blocks and how they are utilized. We'll look at how you might create an audio oscillator, an amplification stage, and an envelope block to control some of our synth parameters. Before we get into the guts of the synth blocks, let's take a moment to think about how we might want to use these blocks together and consider some of the features of MAX that lend themselves to this modular approach.

It would be ideal to have a single window where you could add blocks and arrange and connect them in any way you want. The blocks themselves would need to have input and outputs for audio and control values. The blocks should have the relevant and necessary controls (knobs, sliders, etc.) and displays. We also don't necessarily need to see the inner workings of the completed blocks, and because screen space is at a premium, the blocks should be as compact and simply packaged as possible. From a programming point of view, the blocks should be easily copied and pasted, and if we had to change the internal components of one type of block it would be nice if the change reflected across all the other blocks of that type. In this way, if we wanted, say, to add another waveform type to our oscillator blocks later on, and we had already created and connected dozens of oscillator blocks, we wouldn't have to go through and reprogram each one individually.

There is an object in MAX called the *bpatcher*. It is similar in a sense to a typical *subpatcher* object, in that it provides a means to create multiple elements and a complex functionality inside of one single object. However, where the *subpatcher* physically contains the additional objects within it, the *bpatcher* essentially allows you to embed an entire other patcher file inside your current one. It also provides a "window" into the patcher file itself, showing a customizable area of the embedded patcher contents. Additionally, like the *subpatcher*, the *bpatcher* allows for connection to and from custom inlets and outlets in the patcher file. Using the *bpatcher* object, we're able to create each synth block in its own separate patcher file. We can configure the customizable viewing window to maximize display potential while keeping screen real estate to a minimum. Creating another block is as simple as duplicating the one *bpatcher* object (no multiple objects to deal with), and as the *bpatcher* links to the one external file, updating the block file updates all of the instances in our main synthesizer file. To tell a *bpatcher* which file to link, you open the *bpatcher* Inspector window (Object > Inspector in the file menu, or select the object and Right-click > Inspector) and click the Choose button on the Patcher File line to browse for the patcher file, as shown in Figure 6.46.

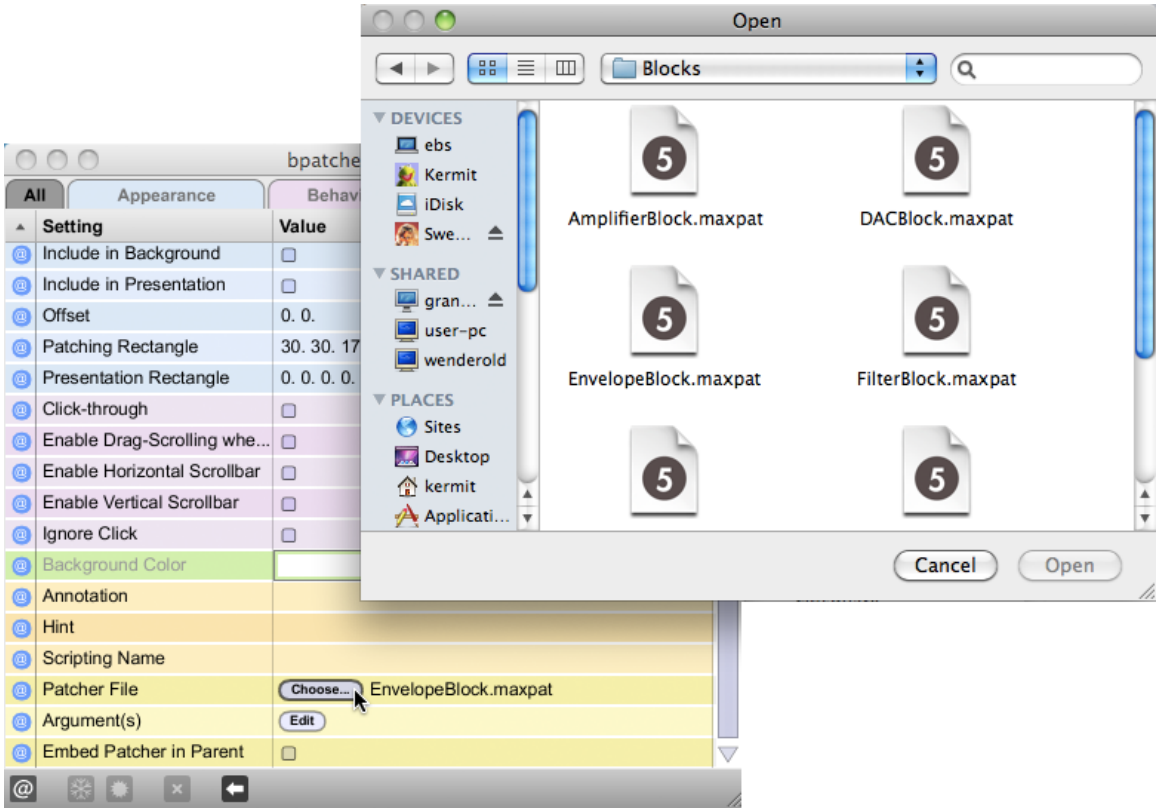


Figure 6.58 *bpatcher* object Inspector

Of course, before we can embed our synth block files, we need to actually create them. Let's start with the oscillator block as that's the initial source for our sound synthesis. Figure 6.47 shows you the oscillator block file in programming mode.

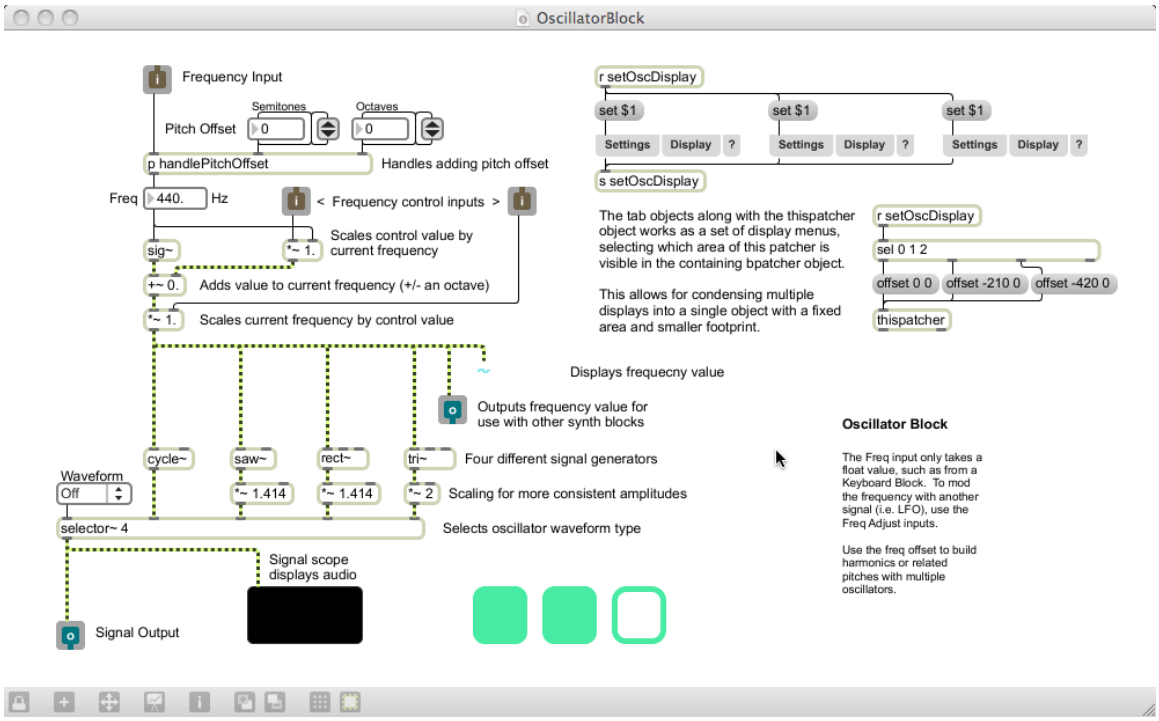


Figure 6.59 Oscillator block in programming mode

While the oscillator block may appear complex at first, there is a good bit of content that we added to improve the handling and display of the block for demonstrative and educational purposes. The objects grouped at the top right side of the patcher enable multiple display views for use within the *bpatcher* object. This is a useful and interesting feature of the *bpatcher* object, but we're not going to focus on the nuts and bolts of it in this section. There is a simple example of this kind of implementation in the built-in *bpatcher* help file (Help > Open [ObjectName] Help in the file menu, or Right-click > Open [ObjectName] Help), so feel free to explore this further on your own. There are some additional objects and connections that are added to give additional display and informational feedback, such as the *scope~* object that displays the audio waveform and a *number~* object as a numerical display of the frequency, but these are relatively straightforward and non-essential to the design of this block. If we were to strip out all the objects and connections that don't deal with audio or aren't absolutely necessary for our synth block to work, it would look more like the file in Figure 6.48.

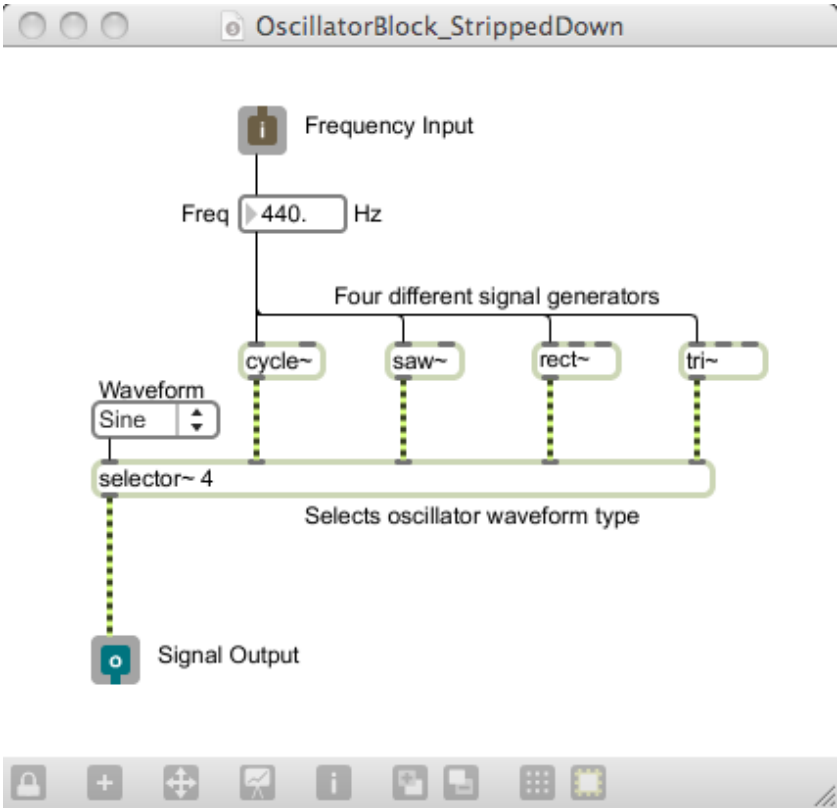


Figure 6.60 Oscillator block stripped down

In its most basic functionality, the oscillator block simply generates an audio signal of some kind. In this case, we chose to provide several different options for the audio signal, including an object for generating basic sine waves (*cycle~*), sawtooth waves (*saw~*), square waves (*rect~*) and triangle waves (*tri~*). Each of these objects takes a number in to its leftmost inlet to set its frequency. The *selector~* object is used to select between multiple signals, and the desired signal is passed through based on the waveform selected in the connected user interface dropdown menu (umenu). The inlets and outlets may seem strange when used in a standalone patcher file, and at this stage they aren't really useful. However, when the file is linked into the *bpatcher* object, these inlets and outlets appear on the *bpatcher* and allow you access to these signals and control values in the main synth patcher.

While this module alone would enable you to begin synthesizing simple sounds, one of the purposes of the modular synthesizer blocks is to be able to wire them together to control each other in interesting ways. In this respect, we may want to consider ways to control or manipulate our oscillator block. The primary parameter in the case of the oscillator is the oscillator frequency, so we should incorporate some control elements for it. Perhaps we would want to increase or decrease the oscillator frequency, or even scale it by an external factor. Consider the way some of the software or hardware synths previously discussed in this chapter have manipulated the oscillator frequency. Looking back at the programming mode for this block, you may recall seeing several inlets labeled "frequency control" and some additional connected objects. A close view of these can be seen in Figure 6.49.

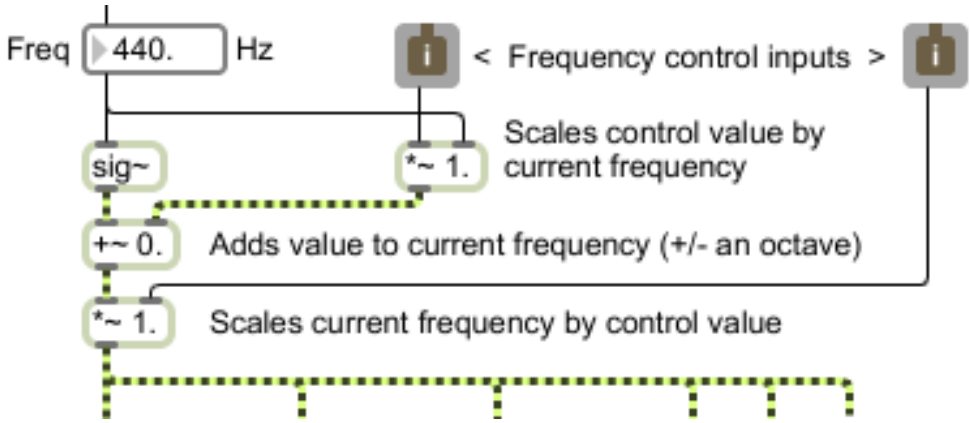


Figure 6.61 Oscillator block frequency control inputs

These control inputs are attached to several arithmetic objects, including addition and multiplication. The "~" following the mathematical operator simply means this version of the object is used for processing signal rate (audio) values. This is the sole purpose of the *sig~* object also seen here; it takes the static numerical value of the frequency (shown at 440 Hz) and outputs it as a signal rate value, meaning it is constantly streaming its output. In the case of these two control inputs, we set one of them up as an additive control, and the other a multiplicative control. The inlet on the left is the additive control. It adds to the oscillator frequency an amount equal to some portion of the current oscillator frequency value. For example, if for the frequency value shown it receives a control value of 1, it adds 440 Hz to the oscillator frequency. If it receives a control value of 0.5, it adds 220 Hz to the oscillator frequency. If it receives a control value of 0, it adds nothing to the oscillator frequency, resulting in no change.

At this point we should take a step back and ask ourselves, "What kind of control values should we be expecting?" There is really no one answer to this, but as the designers of these synth blocks we should perhaps establish a convention that is meaningful across all of our different synthesizer elements. For our purposes, a floating-point range between 0.0 and 1.0 seems simple and manageable. Of course, we can't stop someone from connecting a control value outside of this range to our block if they really wanted. That's just what makes modular synthesis so fun.

Going back to the second control input (multiplicative), you can see that it scales the output of the oscillator directly by this control value. If it receives a control value of 0, the oscillator's frequency is 0 and it outputs silence. If it receives a control value of 1, the signal is multiplied by 1, resulting in no change. Notice that the multiplication object has been set to a default value of 1 so that the initial oscillator without any control input behaves normally. The addition object has been initialized for that behavior as well. It is also very important to include the "." with these initial values, as that tells the object to expect a floating-point value rather than an integer (default). Without the decimal, the object may round or truncate the floating-point value to the nearest integer, resulting in our case to a control value of either 0 or 1, which would be rather limiting.

Let's now take a look at what goes into an amplification block. The amplification block is simply used to control the amplitude, or level, of the synthesized audio. Figure 6.50 shows you our amplifier block file in programming mode. There isn't much need to strip down the non-essentials from this block, as there's just not much to it.

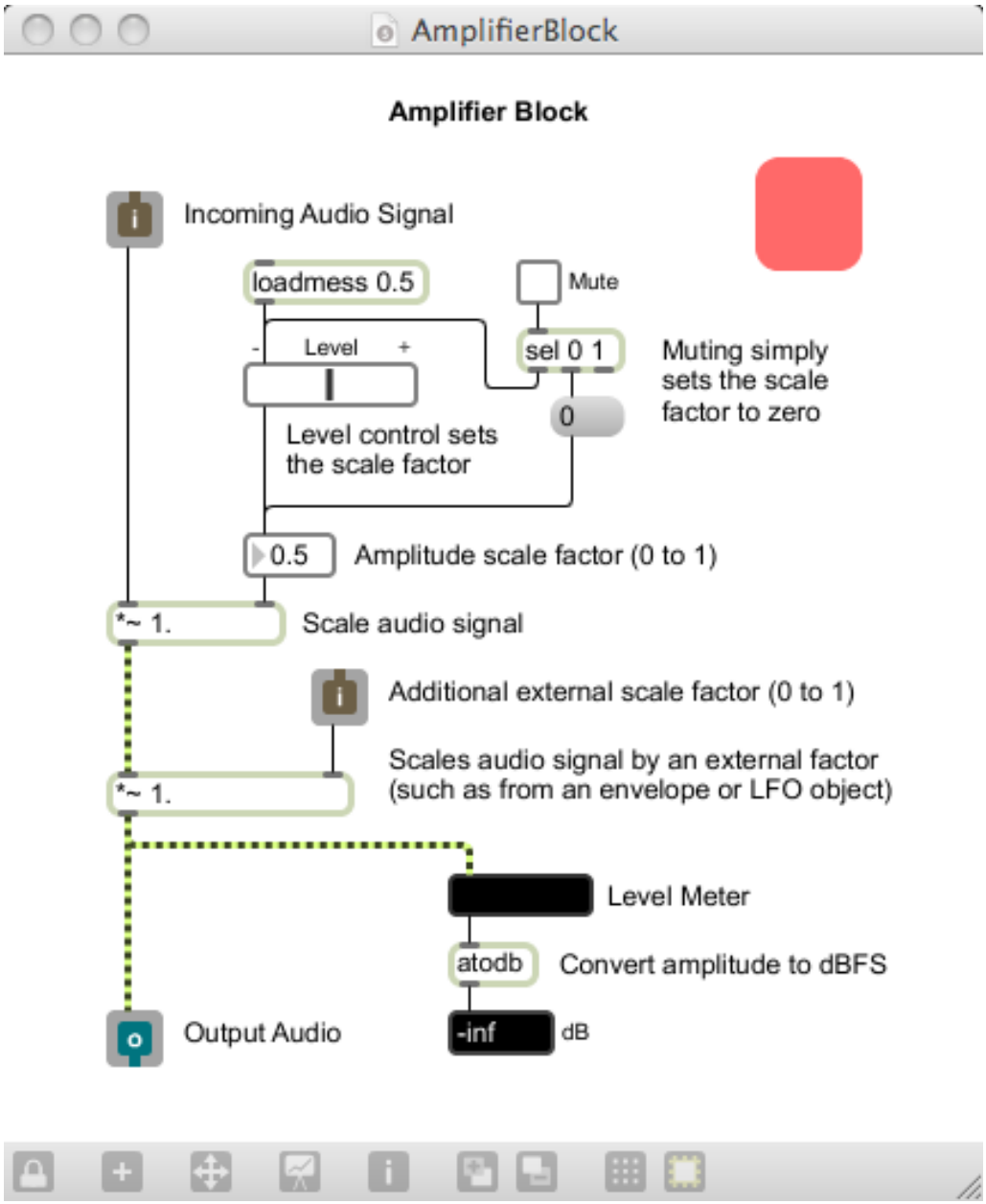


Figure 6.62 Amplifier block in programming mode

Tracing the audio signal path straight down the left side of the patcher, you can see that there are two stages of multiplication, which essentially is amplification in the digital realm. The first stage of amplification is controlled by a level slider and a mute toggle switch. These interface objects allow the overall amplification to be set graphically by the user. The second stage of amplification is set by an external control value. Again this is another good fit for the standard range we decided on of 0.0 to 1.0. A control value of 0 causes the signal amplitude to be scaled to 0, outputting silence. A control value of 1 results in no change in amplification. On the other hand, could you think of a reason we might want to increase the amplitude of the signal, and input a control value greater than 1.0? Perhaps a future synth block may have good reason to do this. Now that we have these available control inputs, we should start thinking of how we might actually control these blocks, such as with an envelope.

Figure 6.51 shows our envelope block in programming mode. Like the oscillator block, there are quite a few objects that we included to deal with additional display and presentational aspects that aren't necessary to the block's basic functionality. It too has a group of objects in the top right corner dedicated to enabling alternate views when embedded in the *bpatcher* object. If we were again to strip out all the objects and connections that don't deal with audio or aren't absolutely necessary for our synth block to work, it would look more like the file in Figure 6.52.

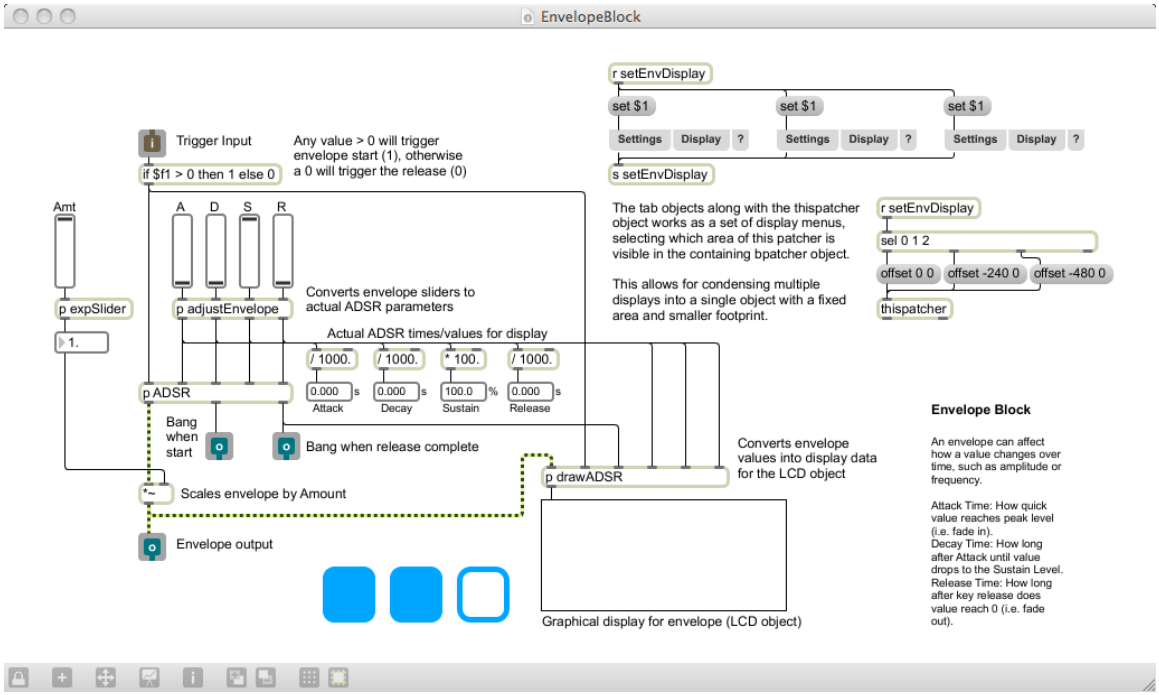


Figure 6.63 Envelope block in programming mode

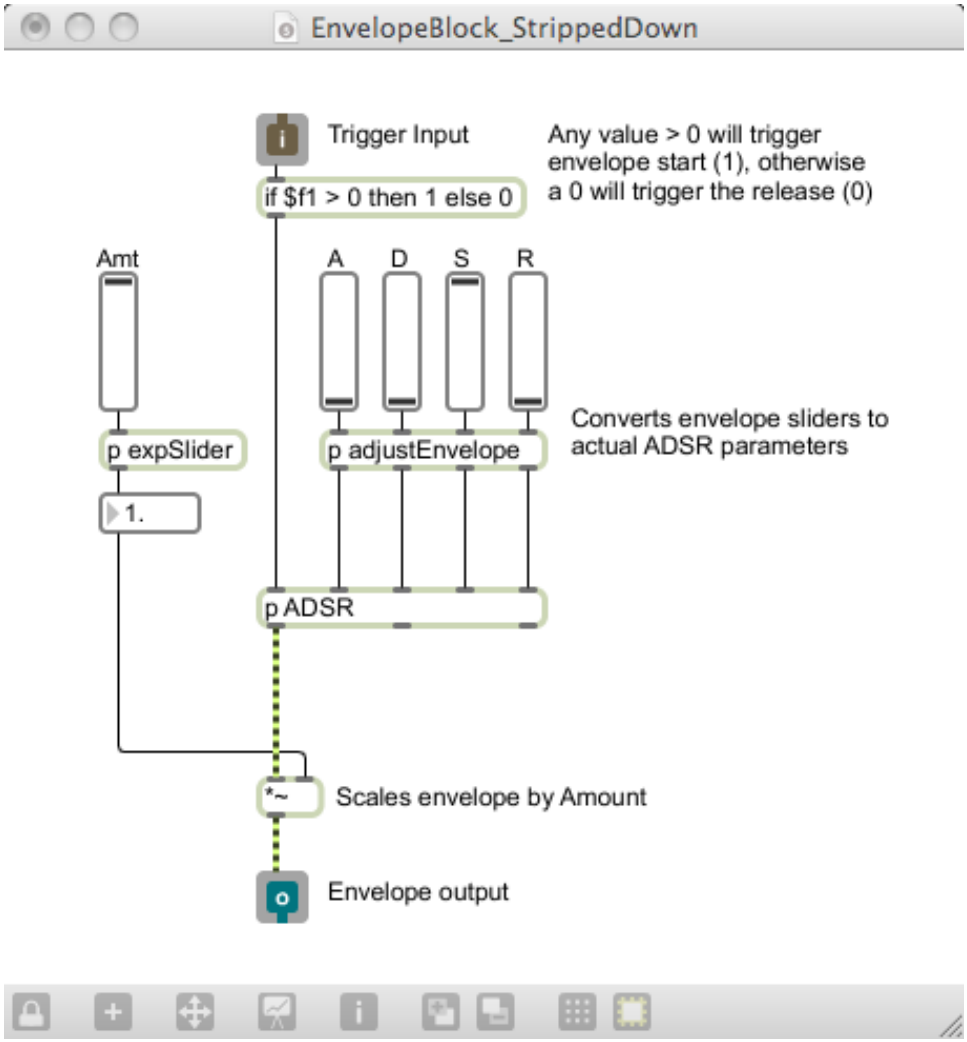


Figure 6.64 Envelope block stripped down

While it incorporates signal rate objects, the envelope doesn't actually involve the audio signal. Its output is purely used for control. Then again, it's possible for the user to connect the envelope block output to the audio signal path, but it probably won't sound very good if it even makes a sound at all. Let's start with the inlet at the top. In order for an envelope to function properly, it has to know when to start and when to stop. Typically it starts when someone triggers a note, such as by playing a key on a keyboard, and releases when the key is let go. Thinking back to MIDI and how MIDI keyboards work, we know that when a key is pressed, it sends a Note On command accompanied by a velocity value of between 1 and 127. When the key is released, it often sends a Note On command with a velocity value of zero. In our case, we don't really care what the value is between 1 and 127; we need to start the envelope regardless of how hard the key is pressed.

This velocity value lends itself to being a good way to tell when to trigger and release an envelope. It would be useful to condition the input to give us only the two cases we care about: start and release. There are certainly multiple ways to implement this in MAX, but a simple *if…then* object is an easy way to check this logic using one object. The *if…then* object checks if

the value at inlet 1 ($f1) is greater than 0 (as is the case for 1 to 127), and if so it outputs the value 1. Otherwise, (if the inlet value is less than or equal to 0) it outputs 0. Since there are no negative velocity values, we've just lumped them into the release outcome to cover our bases. Now our input value results in a 1 or a 0 relating to a start or release envelope trigger.

As is often the case, it just so happens that MAX has a built in *adsr~* object that generates a standard ADSR envelope. We opted to create our own custom ADSR *subpatcher*. However, its purpose and inlets/outlets are essentially identical, and it's meant to be easily interchangeable with the *adsr~* object. You can find out more about our custom ADSR *subpatcher* and the reason for its existence in the MAX Demo "ADSR Envelopes" back in section 6.1.8.5.1. For the moment, we'll refer to it as the standard *adsr~* object.

Now, it just so happens that the *adsr~* object's first inlet triggers the envelope start when it sees any non-zero number, and it triggers the release when it receives a 0. Our input is thus already primed to be plugged straight into this object. It may also seem that our conditioning is now somewhat moot, since any typical velocity value, being a positive number, triggers the envelope start of the *adsr~* object just as well as the number 1. It is however still important to consider that perhaps the user may (for creative reasons) connect a different type of control value to the envelope trigger input, other than note velocity. The conditioning *if…then* object thereby ensures a more widely successful operation with potentially unexpected input formats.

The *adsr~* object also takes four other values as input. These are, as you might imagine, the attack time, decay time, sustain value, and release time. For simple controls, we've provided the user with four sliders to manipulate these parameters. In between the sliders and the *adsr~* inlets, we need to condition the slider values to relate to usable ADSR values. The *subpatcher adjustEnvelope* does just that, and its contents can be seen in Figure 6.53. The four slider values (default 0 to 127) are converted to time values of appropriate scale, and in the case of sustain a factor ranging from 0.0 to 1.0. The *expSlider subpatcher*s as seen in earlier MAX demos give the sliders an exponential curve, allowing for greater precision at lower values.

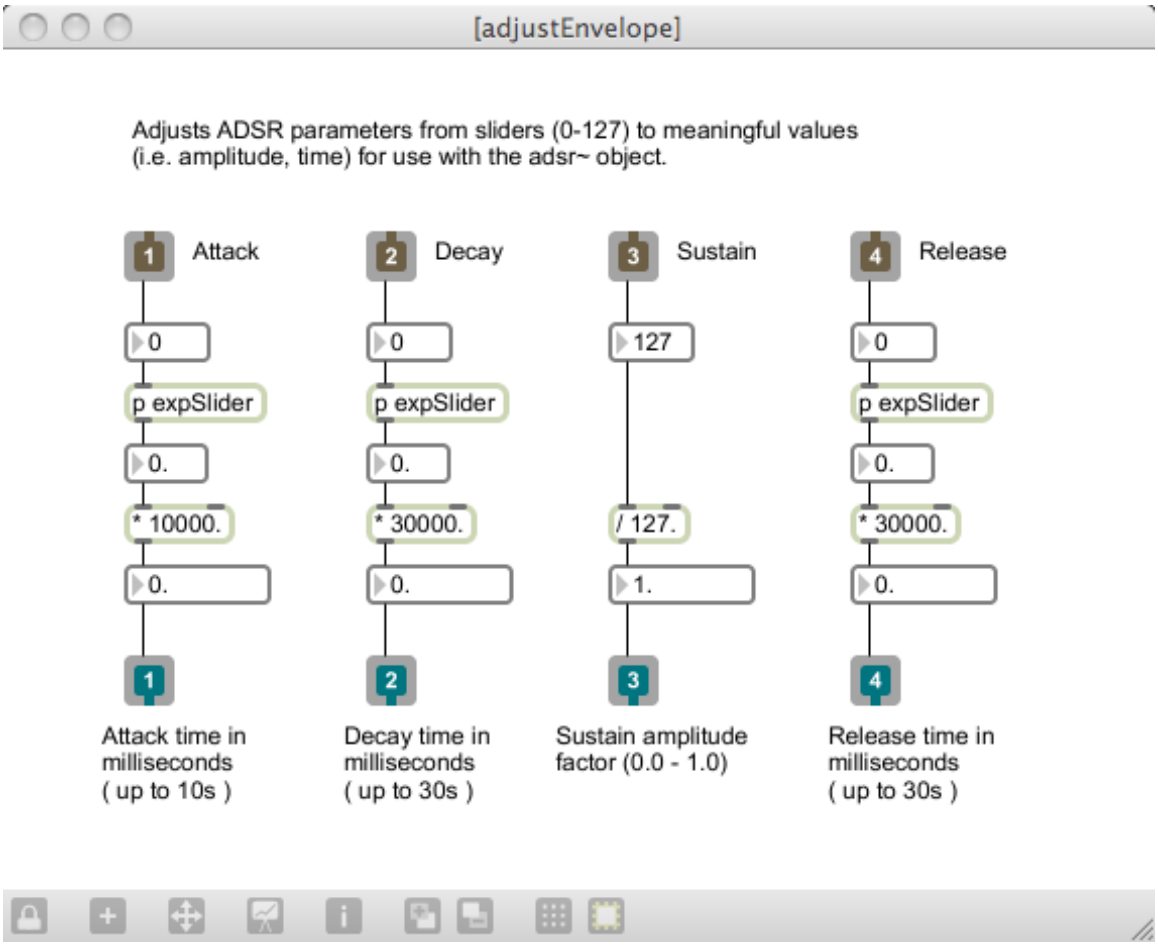


Figure 6.65 *adjustEnvelope* subpatcher contents

The generated ADSR envelope ranges from 0.0 to 1.0, rising and falling according to the ADSR input parameters. Before reaching the final outlet, the envelope is scaled by a user definable amount also ranging from 0.0 to 1.0, which essentially controls the impact the envelope has on the controllable parameter it is affecting.

It would too time-consuming to go through all of the modular synth blocks that we have created or that could potentially be created as part of this MAX example. However, these and other modular blocks can be downloaded and explored from the "Block Synthesizer" MAX demo linked at the start of this section. The blocks include the oscillator, amplifier, and envelope blocks discussed here, as well as a keyboard input block, LFO block, filter block, and a DAC block. Each individual block file is fully commented to assist you in understanding how it was put together. These blocks are all incorporated as *bpatcher* objects into a main *BlockSynth.maxpat* patcher, shown in Figure 6.54, where they can all be arranged, duplicated, and connected in various configurations to create a great number of synthesis possibilities. You can also check out the additional included *BlockSynth_Example.maxpat* and example settings files to see a few configurations and settings we came up with to create some unique synth sounds. Feel free to come up with your own synth blocks and implement them as well. Some possible ideas might be a metering block to provide additional displays and meters for the audio signal, a noise generator block for an alternative signal source or modulator, or perhaps a

polyphony block to manage several instances of oscillator blocks and allow multiple simultaneous notes to be played.

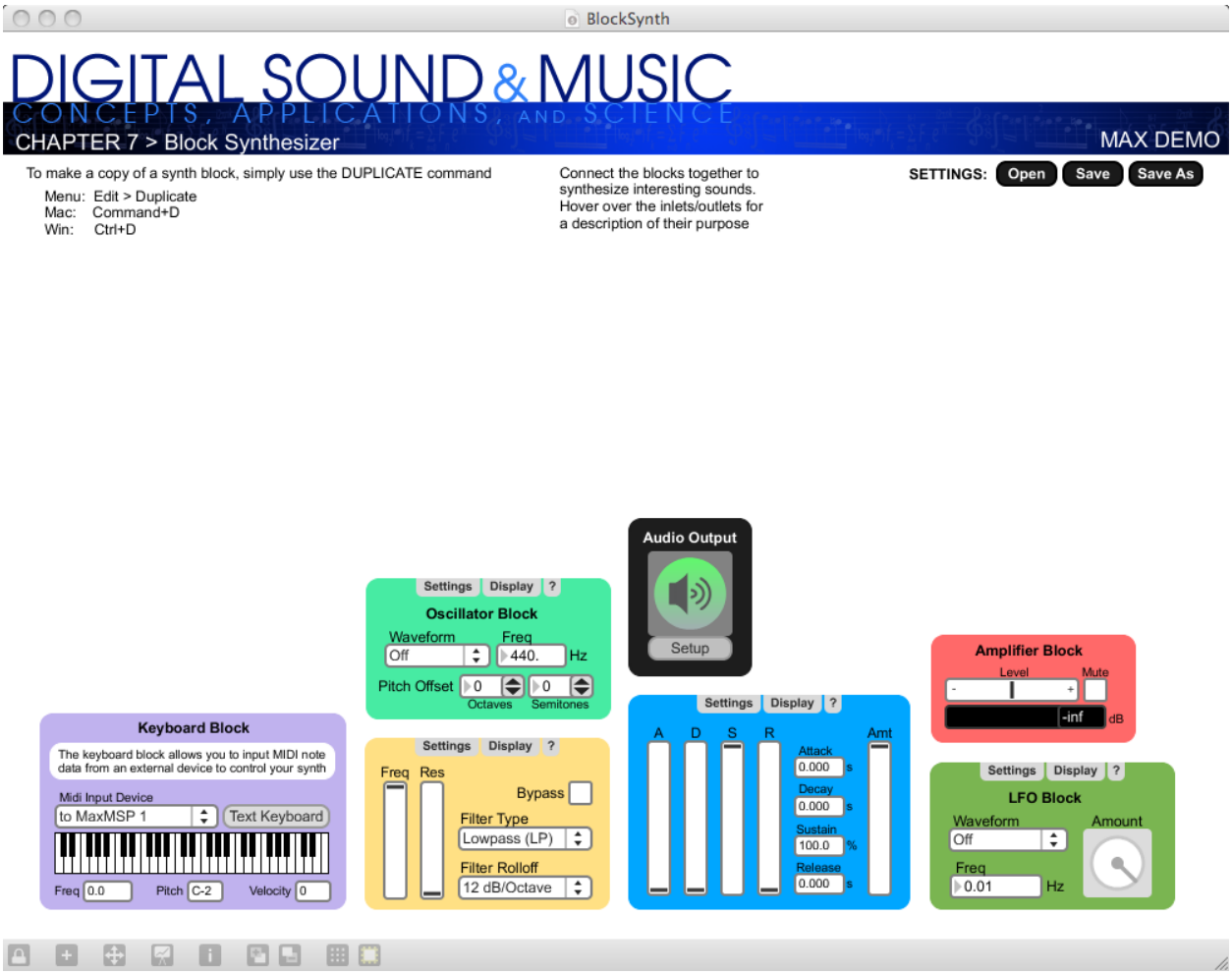
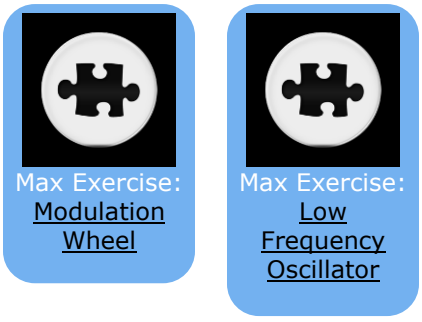


Figure 6.66 Main block synth patcher with embedded *bpatcher* blocks

Software synthesis is really the bedrock of the Max program, and you can learn a lot about Max through programming synthesizers. To that end, we have also created two Max programming exercises using our Subtractonaut synthesizer. These programming assignments challenge you to add some features to the existing synthesizer. Solution files are available that show how these features might be programmed.



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