This section is for those who wish to know the technical details of MIDI (including General MIDI, the MIDI file format, MIDI Sample Dump Standard, and MIDI Time Code), as well as obtain some programming information.

The MIDI Specification

MIDI (ie, Musical Instrument Digital Interface) consists of both a simple hardware interface, and a more elaborate transmission protocol.

For a simple, layman's explanation of what MIDI is, read What is MIDI?

The MIDI Specification is published by the MIDI Manufacturer's Association, ie, MMA (although this online document gives you the same information for free, in easier-to-understand language, and in many cases, with even more detail than the official document).

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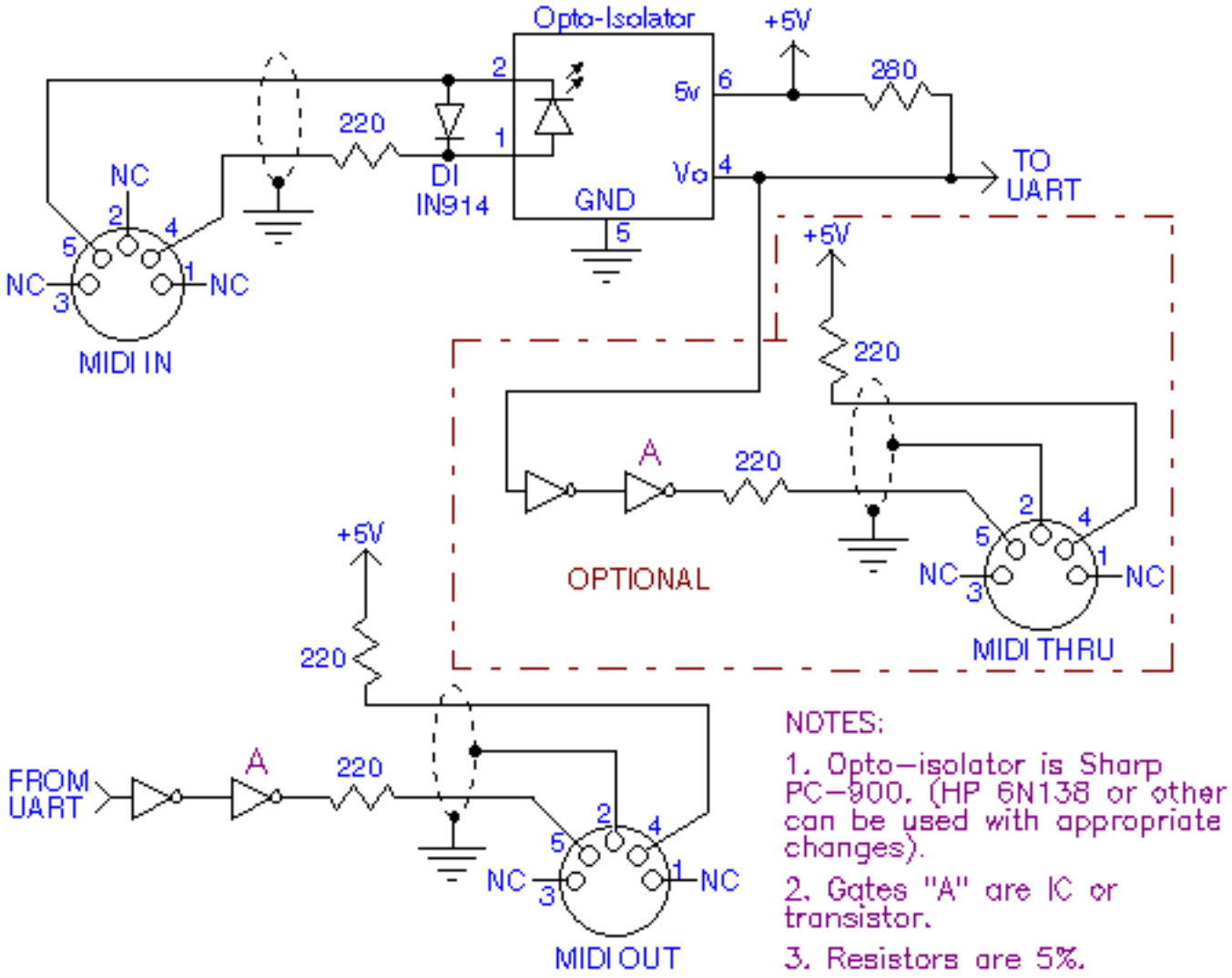
MIDI is an asynchronous serial interface. The baud rate is 31.25 Kbaud (+/- 1%). There is 1 start bit, 8 data bits, and 1 stop bit (ie, 10 bits total), for a period of 320 microseconds per serial byte.

The MIDI circuit is current loop, 5 mA. Logic 0 is current ON. One output drives one (and only one) input. To avoid grounding loops and subsequent data errors, the input is opto-isolated. It requires less than 5 mA to turn on. The Sharp PC-900 and HP 6N138 optoisolators are satisfactory devices. Rise and fall time for the optoisolator should be less than 2 microseconds.

The standard connector used for MIDI is a 5 pin DIN. Separate jacks (and cable runs) are used for input and output, clearly marked on a given device (ie, the MIDI IN and OUT are two separate DIN female panel mount jacks). 50 feet is the recommended maximum cable length. Cables are shielded twisted pair, with the shield connecting pin 2 at both ends. The pair is pins 4 and 5. Pins 1 and 3 are not used, and should be left unconnected.

A device may also be equipped with a **MIDI THRU** jack which is used to pass the MIDI IN signal to another device. The MIDI THRU transmission may not be performed correctly due to the delay time (caused by the response time of the opto-isolator) between the rising and falling edges of the square wave. These timing errors will tend to add in the "wrong direction" as more devices are daisy-chained to other device's MIDI THRU jacks. The result is that there is a limit to the number of devices that can be daisy-chained.

Schematic



A schematic of a MIDI (IN and OUT) interface

The MIDI protocol is made up of **messages**. A message consists of a string (ie, series) of 8-bit bytes. MIDI has many such defined messages. Some messages consist of only 1 byte. Other messages have 2 bytes. Still others have 3 bytes. One type of MIDI message can even have an unlimited number of bytes. The one thing that all messages have in common is that the first byte of the message is the **Status** byte. This is a special byte because it's the only byte that has bit #7 set. Any other following bytes in that message will not have bit #7 set. So, you can always detect the start a MIDI message because that's when you receive a byte with bit #7 set. This will be a Status byte in the range 0x80 to 0xFF. The remaining bytes of the message (ie, the data bytes, if any) will be in the range 0x00 to 0x7F. (Note that I'm using the C programming language convention of prefacing a value with 0x to indicate hexadecimal).

The Status bytes of 0x80 to 0xEF are for messages that can be broadcast on any one of the 16 MIDI channels. Because of this, these are called **Voice** messages. (My own preference is to say that these messages belong in the **Voice Category**). For these Status bytes, you break up the 8-bit byte into 2 4-bit nibbles. For example, a Status byte of 0x92 can be broken up into 2 nibbles with values of 9 (high nibble) and 2 (low nibble). The high nibble tells you what *type* of MIDI message this is. Here are the possible values for the high nibble, and what type of Voice Category message each represents:

8 = Note Off
9 = Note On
A = AfterTouch (ie, key pressure)
B = Control Change
C = Program (patch) change
D = Channel Pressure
E = Pitch Wheel

So, for our example status of 0x92, we see that its message type is **Note On** (ie, the high nibble is 9). What's the low nibble of 2 mean? This means that the message is on MIDI channel 2. There are 16 possible (logical) MIDI channels, with 0 being the first. So, this message is a Note On on channel 2. What status byte would specify a **Program Change** on channel 0? The high nibble would

need to be C for a Program Change type of message, and the low nibble would need to be 0 for channel 0. Thus, the status byte would be 0xC0. How about a Program Change on channel 15 (ie, the last MIDI channel). Again, the high nibble would be C, but the low nibble would be F (ie, the hexademical digit for 15).Thus, the status would be 0xCF.

NOTE: Although the MIDI Status byte counts the 16 MIDI channels as numbers 0 to F (ie, 15), all MIDI gear (including computer software) displays a channel number to the musician as 1 to 16. So, a Status byte sent on MIDI channel 0 is considered to be on "channel 1" as far as the musician is concerned. This discrepancy between the status byte's channel number, and what channel the musician "believes" that a MIDI message is on, is accepted because most humans start counting things from 1, rather than 0.

The Status bytes of 0xF0 to 0xFF are for messages that aren't on any particular channel (and therefore all daisy-chained MIDI devices always can "hear" and choose to act upon these messages. Contrast this with the Voice Category messages, where a MIDI device can be set to respond to those MIDI messages only on a specified channel). These status bytes are used for messages that carry information of interest to all MIDI devices, such as syncronizing all playback devices to a particular time. (By contrast, Voice Category messages deal with the individual musical parts that each instrument might play, so the channel nibble scheme allows a device to respond to its own MIDI channel while ignoring the Voice Category messages intended for another device on another channel).

These status bytes are further divided into two categories. Status bytes of 0xF0 to 0xF7 are called **System Common** messages. Status bytes of 0xF8 to 0xFF are called **System Realtime** messages. The implications of such will be discussed later.

Actually, certain Status bytes within this range are not defined by the MIDI spec to date, and are reserved for future use. For example, Status bytes of 0xF4, 0xF5, 0xF9, and 0xFD are not used. If a MIDI device ever receives such a Status, it should ignore that message. See Ignoring MIDI Messages.

What follows is a description of each message type. The description tells what the message does, what its status byte is, and whether it has any subsequent data bytes and what information those carry. Generally, these descriptions take the view of a device receiving such messages (ie, what the device would typically be expected to do when receiving particular messages). When applicable, remarks about a device that transmits such messages may be made.

Note Off

Category: Voice

Purpose

Indicates that a particular note should be released. Essentially, this means that the note stops sounding, but some patches might have a long VCA release time that needs to slowly fade the sound out. Additionally, the device's Hold Pedal controller may be on, in which case the note's release is postponed until the Hold Pedal is released. In any event, this message either causes the VCA to move into the release stage, or if the Hold Pedal is on, indicates that the note should be released (by the device automatically) when the Hold Pedal is turned off. If the device is a MultiTimbral unit, then each one of its Parts may respond to Note Offs on its own channel. The Part that responds to a particular Note Off message is the one assigned to the message's MIDI channel.

Status

0x80 to 0x8F where the low nibble is the MIDI channel.

Data

Two data bytes follow the Status.

The first data is the note number. There are 128 possible notes on a MIDI device, numbered 0 to 127 (where Middle C is note number 60). This indicates which note should be released.

The second data byte is the velocity, a value from 0 to 127. This indicates how quickly the note should be released (where 127 is the fastest). It's up to a MIDI device how it uses velocity information. Often velocity will be used to tailor the VCA release time. MIDI devices that can generate Note Off messages, but don't implement velocity features, will transmit Note Off messages with a preset velocity of 64.

Errata

An All Notes Off controller message can be used to turn off all notes for which a device received **Note On** messages (without having received respective Note Off messages).

Note On

Category: Voice

Purpose

Indicates that a particular note should be played. Essentially, this means that the note starts sounding, but some patches might have a long VCA attack time that needs to slowly fade the sound in. In any case, this message indicates that a particular note should start playing (unless the velocity is 0, in which case, you really have a Note Off). If the device is a MultiTimbral unit, then each one of its Parts may sound Note Ons on its own channel. The Part that sounds a particular Note On message is the one assigned to the message's MIDI channel.

Status

0x90 to 0x9F where the low nibble is the MIDI channel.

Data

Two data bytes follow the Status.

The first data is the note number. There are 128 possible notes on a MIDI device, numbered 0 to 127 (where Middle C is note number 60). This indicates which note should be played.

The second data byte is the velocity, a value from 0 to 127. This indicates with how much force the note should be played (where 127 is the most force). It's up to a MIDI device how it uses velocity information. Often velocity is be used to tailor the VCA attack time and/or attack level (and therefore the overall volume of the note). MIDI devices that can generate Note On messages, but don't implement velocity features, will transmit Note On messages with a preset velocity of 64.

A Note On message that has a velocity of 0 is considered to actually be a Note Off message, and the respective note is therefore released. See the Note Off entry for a description of such. This "trick" was created in order to take advantage of running status.

A device that recognizes MIDI Note On messages must be able to recognize both a real Note Off as well as a Note On with 0 velocity (as a Note Off). There are many devices that generate real Note Offs, and many other devices that use Note On with 0 velocity as a substitute.

Errata

In theory, every Note On should eventually be followed by a respective Note Off message (ie, when it's time to stop the note from sounding). Even if the note's sound fades out (due to some VCA envelope decay) before a Note Off for this note is received, at some later point a Note Off should be received. For example, if a MIDI device receives the following Note On:

0x90 0x3C 0x40 Note On/chan 0, Middle C, velocity could be anything except 0

Then, a respective Note Off should subsequently be received at some time, as so:

0x80 0x3C 0x40 Note Off/chan 0, Middle C, velocity could be anything

Instead of the above Note Off, a Note On with 0 velocity could be substituted as so:

0x90 0x3C 0x00 Really a Note Off/chan 0, Middle C, velocity must be 0

If a device receives a Note On for a note (number) that is already playing (ie, hasn't been turned off yet), it the device's decision whether to layer another "voice" playing the same pitch, or cut off the voice playing the preceding note of that same pitch in order to "retrigger" that note.

Aftertouch

Category: Voice

Purpose

While a particular note is playing, pressure can be applied to it. Many electronic keyboards have pressure sensing circuitry that can detect with how much force a musician is holding down a key. The musician can then vary this pressure, even while he continues to hold down the key (and the note continues sounding). The Aftertouch message conveys the amount of pressure on a key at a given point. Since the musician can be continually varying his pressure, devices that generate Aftertouch typically send out many such messages while the musician is varying his pressure. Upon receiving Aftertouch, many devices typically use the message to vary a note's VCA and/or VCF envelope sustain level, or control LFO amount and/or rate being applied to the note's sound generation circuitry. But, it's up to the device how it chooses to respond to received Aftertouch (if at all). If the device is a MultiTimbral unit, then each one of its Parts may respond differently (or not at all) to Aftertouch. The Part affected by a particular Aftertouch message is the one assigned to the message's MIDI channel.

It is recommended that Aftertouch default to controlling the LFO amount (ie, a vibrato effect).

Status

0xA0 to 0xAF where the low nibble is the MIDI channel.

Data

Two data bytes follow the Status.

The first data is the note number. There are 128 possible notes on a MIDI device, numbered 0 to 127 (where Middle C is note number 60). This indicates to which note the pressure is being applied.

The second data byte is the pressure amount, a value from 0 to 127 (where 127 is the most pressure).

Errata

See the remarks under Channel Pressure.

Controller

Category: Voice

Purpose

Sets a particular controller's value. A controller is any switch, slider, knob, etc, that implements some function (usually) other than sounding or stopping notes (ie, which are the jobs of the Note On and Note Off messages respectively). There are 128 possible controllers on a MIDI device. These are numbered from 0 to 127. Some of these controller numbers are assigned to particular hardware controls on a MIDI device. For example, controller 1 is the **Modulation Wheel**. Other controller numbers are free to be arbitrarily interpreted by a MIDI device. For example, a drum box may have a slider controlling Tempo which it arbitrarily assigns to one of these free numbers. Then, when the drum box receives a Controller message with that controller number, it can adjust its tempo. A MIDI device need not have an actual physical control on it in order to respond to a particular controller. For example, even though a rack-mount sound module may not have a **Mod Wheel** on it, the module will likely still respond to and utilize **Modulation controller** messages to modify its sound. If the device is a MultiTimbral unit, then each one of its Parts may respond differently (or not at all) to various controller numbers. The Part affected by a particular controller message is the one assigned to the message's MIDI channel.

Status

0xB0 to 0xBF where the low nibble is the MIDI channel.

Data

Two data bytes follow the Status.

The first data is the controller number (0 to 127). This indicates which controller is affected by the received MIDI message.

The second data byte is the value to which the controller should be set, a value

from 0 to 127.

Errata

An All Controllers Off controller message can be used to reset all controllers (that a MIDI device implements) to default values. For example, the **Mod Wheel** is reset to its "off" position upon receipt of this message.

See the list of Defined Controller Numbers for more information about particular controllers.

Controller Numbers

A **Controller** message has a Status byte of 0xB0 to 0xBF depending upon the MIDI channel. There are two more data bytes.

The first data byte is the **Controller Number**. There are 128 possible controller numbers (ie, 0 to 127). Some numbers are defined for specific purposes. Others are undefined, and reserved for future use.

The second byte is the "value" that the controller is to be set to.

Most controllers implement an effect even while the MIDI device is generating sound, and the effect will be immediately noticeable. In other words, MIDI controller messages are meant to implement various effects by a musician while he's operating the device.

If the device is a MultiTimbral module, then each one of its Parts may respond differently (or not at all) to a particular controller number. Each Part usually has its own setting for every controller number, and the Part responds only to controller messages on the same channel as that to which the Part is assigned. So, controller messages for one Part do not affect the sound of another Part even while that other Part is playing.

Some controllers are **continuous** controllers, which simply means that their value can be set to any value within the range from 0 to 16,384 (for 14-bit coarse/fine resolution) or 0 to 127 (for 7-bit, coarse resolution). Other controllers are **switches** whose state may be either ***on*** or ***off***. Such controllers will usually generate only one of two values; 0 for off, and 127 for on. But, a device should be able to respond to any received switch value from 0 to 127. If the device implements only an "on" and "off" state, then it should regard values of 0 to 63 as off, and any value of 64 to 127 as on.

Many (continuous) controller numbers are **coarse** adjustments, and have a respective **fine** adjustment controller number. For example, controller #1 is the coarse adjustment for Modulation Wheel. Using this controller number in a message, a device's Modulation Wheel can be adjusted in large (coarse)

increments (ie, 128 steps). If finer adjustment (from a coarse setting) needs to be made, then controller #33 is the fine adjust for Modulation Wheel. For controllers that have coarse/fine pairs of numbers, there is thus a 14-bit resolution to the range. In other words, the Modulation Wheel can be set from 0x0000 to 0x3FFF (ie, one of 16,384 values). For this 14-bit value, bits 7 to 13 are the coarse adjust, and bits 0 to 6 are the fine adjust. For example, to set the Modulation Wheel to 0x2005, first you have to break it up into 2 bytes (as is done with Pitch Wheel messages). Take bits 0 to 6 and put them in a byte that is the fine adjust. Take bits 7 to 13 and put them right-justified in a byte that is the coarse adjust. Assuming a MIDI channel of 0, here's the coarse and fine Mod Wheel controller messages that a device would receive (coarse adjust first):

```
0xB0 0x01 0x40
Controller on chan 0, Mod Wheel coarse, bits 7 to 13 of 14-bit
value right-justified (with high bit clear).

0xB0 0x33 0x05
Controller on chan 0, Mod Wheel fine, bits 0 to 6 of 14-bit
value (with high bit clear).
```

Some devices do not implement fine adjust counterparts to coarse controllers. For example, some devices do not implement controller #33 for Mod Wheel fine adjust. Instead the device only recognizes and responds to the Mod Wheel coarse controller number (#1). It is perfectly acceptable for devices to only respond to the coarse adjustment for a controller if the device desires 7-bit (rather than 14-bit) resolution. The device should ignore that controller's respective fine adjust message. By the same token, if it's only desirable to make fine adjustments to the Mod Wheel without changing its current coarse setting (or vice versa), a device can be sent only a controller #33 message without a preceding controller #1 message (or vice versa). Thus, if a device can respond to both coarse and fine adjustments for a particular controller (ie, implements the full 14-bit resolution), it should be able to deal with either the coarse or fine controller message being sent without its counterpart following. The same holds true for other continuous (ie, coarse/fine pairs of) controllers.

Note: In most MIDI literature, the coarse adjust is referred to with the designation "MSB" and the fine adjust is referred to with the designation

"LSB". I prefer the terms "coarse" and "fine".

Here's a list of the defined controllers. To the left is the controller number (ie, how the MIDI Controller message refers to a particular controller), and on the right is its name (ie, how a human might refer to the controller). To get more information about what a particular controller does, click on its controller name to bring up a description. Each description shows the controller name and number, what the range is for the third byte of the message (ie, the "value" data byte), and what the controller does. For controllers that have separate coarse and fine settings, both controller numbers are shown.

MIDI devices should use these controller numbers for their defined purposes, as much as possible. For example, if the device is able to respond to **Volume controller** (coarse adjustment), then it should expect that to be controller number 7. It should not use **Portamento Time controller** messages to adjust volume. That wouldn't make any sense. Other controllers, such as **Foot Pedal**, are more general purpose. That pedal could be controlling the tempo on a drum box, for example. But generally, the Foot Pedal shouldn't be used for purposes that other controllers already are dedicated to, such as adjusting **Pan position**. If there is not a defined controller number for a particular, needed purpose, a device can use the General Purpose Sliders and Buttons, or NRPN for device specific purposes. The device should use controller numbers 0 to 31 for coarse adjustments, and controller numbers 32 to 63 for the respective fine adjustments.

Defined Controllers

13 Effect Control 2 (coarse)
16 General Purpose Slider 1
17 General Purpose Slider 2
18 General Purpose Slider 3
19 General Purpose Slider 4
32 Bank Select (fine)
33 Modulation Wheel (fine)
34 Breath controller (fine)
36 Foot Pedal (fine)
37 Portamento Time (fine)
38 Data Entry (fine)
39 Volume (fine)
40 Balance (fine)
42 Pan position (fine)
43 Expression (fine)
44 Effect Control 1 (fine)
45 Effect Control 2 (fine)
64 Hold Pedal (on/off)
65 Portamento (on/off)
66 Sustenuto Pedal (on/off)
67 Soft Pedal (on/off)
68 Legato Pedal (on/off)
69 Hold 2 Pedal (on/off)
70 Sound Variation
71 Sound Timbre
72 Sound Release Time
73 Sound Attack Time
74 Sound Brightness
75 Sound Control 6
76 Sound Control 7
77 Sound Control 8
78 Sound Control 9
79 Sound Control 10
80 General Purpose Button 1 (on/off)
81 General Purpose Button 2 (on/off)
82 General Purpose Button 3 (on/off)
83 General Purpose Button 4 (on/off)
91 Effects Level
92 Tremulo Level
93 Chorus Level
94 Celeste Level

95 Phaser Level
96 Data Button increment
97 Data Button decrement
98 Non-registered Parameter (fine)
99 Non-registered Parameter (coarse)
100 Registered Parameter (fine)
101 Registered Parameter (coarse)
120 All Sound Off
121 All Controllers Off
122 Local Keyboard (on/off)
123 All Notes Off
124 Omni Mode Off
125 Omni Mode On
126 Mono Operation
127 Poly Operation

Bank Select

Number: 0 (coarse) 32 (fine)

Affects:

Some MIDI devices have more than 128 Programs (ie, Patches, Instruments, Preset, etc). A MIDI Program Change message supports switching between only 128 programs. So, Bank Select Controller (sometimes called Bank Switch) is sometimes used to allow switching between groups of 128 programs. For example, let's say that a device has 512 Programs. It may divide these into 4 banks of 128 programs apiece. So, if you want program #129, that would actually be the first program within the second bank. You would send a Bank Select Controller to switch to the second bank, and then follow with a Program Change to select the first Program in this bank. If a MultiTimbral device, then each Part usually can be set to its own Bank/Program.

On MultiTimbral devices that have a Drum Part, the Bank Select is sometimes used to switch between "Drum Kits".

Note: When a Bank Select is received, the MIDI module doesn't actually change to a patch in the new bank. Rather, the Bank Select value is simply stored by the MIDI module without changing the current patch. Whenever a subsequent Program Change is received, the stored Bank Select is then utilized to switch to the specified patch in the new bank. For this reason, Bank Select must be sent before a Program Change, when you desire changing to a patch in a different bank. (Of course, if you simply wish to change to another patch in the same bank, there is no need to send a Bank Select first).

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF. Various manufacturers may number their banks differently. Consult the documentation for a MIDI device to determine what values to use with the Bank Select Coarse and Bank Select Fine messages to select particular banks.

Note: Most devices use the Coarse adjust (#0) alone to switch banks since most devices don't have more than 128 banks (of 128 Patches each).

MOD Wheel

Number: 1 (coarse) 33 (fine)

Affects:

Sets the **MOD Wheel** to a particular value. Usually, MOD Wheel introduces some sort of (LFO) vibrato effect. If a MultiTimbral device, then each Part usually has its own MOD Wheel setting.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is no modulation effect.

Breath Controller

Number: 2 (coarse) 34 (fine)

Affects:

Whatever the musician sets this controller to affect. Often, this is used to control a parameter such as what Aftertouch can. After all, breath control is a wind player's version of how to vary pressure. If a MultiTimbral device, then each Part usually has its own Breath Controller setting.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum breath pressure.

Foot Pedal

Number: 4 (coarse) 36 (fine)

Affects:

Whatever the musician sets this controller to affect. Often, this is used to control a parameter such as what Aftertouch may control. This foot pedal is a continuous controller (ie, potentiometer). If a MultiTimbral device, then each Part usually has its own Foot Pedal value.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum effect.

Portamento Time

Number: 5 (coarse) 37 (fine)

Affects:

The rate at which portamento slides the pitch between 2 notes. If a MultiTimbral device, then each Part usually has its own Portamento Time.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is slowest rate.

Data Entry Slider

Number: 6 (coarse) 38 (fine)

Affects:

The value of some Registered or Non-Registered Parameter. Which parameter is affected depends upon a preceding RPN or NRPN message (which itself identifies the parameter's number).

On some devices, this slider may not be used in conjunction with RPN or NRPN messages. Instead, the musician can set the slider to control a single parameter directly, often a parameter such as what Aftertouch can control.

If a MultiTimbral device, then each Part usually has its own RPN and NRPN settings, and Data Entry slider setting.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum effect.

Volume

Number: 7 (coarse) 39 (fine)

Affects:

The volume level for one MIDI channel. If a MultiTimbral device, then each Part has its own volume.

Note: A device's master volume may be controlled by another method such as the Univeral SysEx Master Volume message, or take its volume from one of the Parts, or be controlled by a General Purpose Slider controller.

Expression Controller also may affect the volume.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is no volume at all.

Note: Most all devices ignore the Fine adjust (#39) for Volume, and just implement Coarse adjust (#7) because 14-bit resolution isn't needed for this. In this case, maximum is 127 and off is 0.

It is recommended that a device use the volume value in a logarithmic manner, as specified by the following formula if only the coarse value is used:

$$40 \log (Volume/127)$$

If both the coarse and fine values are used (and combined into a 14-bit volume), then use the following formula:

$$40 \log (Volume/127^2)$$

Note: In the above formula, "Volume" may be equal to Channel Volume * Expression.

Balance

Number: 8 (coarse) 40 (fine)

Affects:

The device's stereo balance (assuming that the device has stereo audio outputs). If a MultiTimbral device, then each Part usually has its own Balance. This is generally when Balance becomes useful, because then you can use Pan, Volume, and Balance controllers to internally mix all of the Parts to the device's stereo outputs. Typically, Balance would be used on a Part that had stereo elements (where you wish to adjust the volume of the stereo elements without changing their pan positions), whereas Pan is more appropriate for a Part that is strictly a "mono instrument".

Value Range:

14-bit coarse/fine resolution. 16,384 possible setting, 0x0000 to 0x3FFF where 0x2000 is center balance, 0x0000 emphasizes the left elements mostly, and 0x3FFF emphasizes the right elements mostly. Some devices only respond to coarse adjust (128 settings) where 64 is center, 0 is leftmost emphsis, and 127 is rightmost emphasis.

Note: Most all devices ignore the Fine adjust (#40) for Balance, and just implement Coarse adjust (#8) because 14-bit resolution isn't needed for this.

Pan

Number: 10 (coarse) 42 (fine)

Affects:

Where within the stereo field the device's sound will be placed (assuming that it has stereo audio outputs). If a MultiTimbral device, then each Part usually has its own pan position. This is generally when Pan becomes useful, because then you can use Pan, Volume, and Balance controllers to internally mix all of the Parts to the device's stereo outputs.

Pan should effect all notes on the channel, including notes that were triggered prior to the pan message being received, and are still sustaining.

Value Range:

14-bit coarse/fine resolution. 16,384 possible positions, 0x0000 to 0x3FFF where 0x2000 is center position, 0x0000 is hard left, and 0x3FFF is hard right. Some devices only respond to coarse adjust (128 positions) where 64 is center, 0 is hard left, and 127 is hard right.

Note: Most all devices ignore the Fine adjust (#42) for Pan, and just implement Coarse adjust (#10) because 14-bit resolution isn't needed for this.

Expression

Number: 11 (coarse) 43 (fine)

Affects:

This is a percentage of Volume (ie, as set by Volume Controller). In other words, Expression divides the current volume into 16,384 steps (or 128 if 8-bit instead of 14-bit resolution is used). Volume Controller is used to set the overall volume of the entire musical part (on a given channel), whereas Expression is used for doing crescendos and decrescendos. By having both a master Volume and sub-Volume (ie, Expression), it makes possible to do crescendos and decrescendos without having to do algebraic calculations to maintain the relative balance between instruments. When Expression is at 100% (ie, the maximum of 0x3FFF), then the volume represents the true setting of Volume Controller. Lower values of Expression begin to subtract from the volume. When Expression is 0% (ie, 0x0000), then volume is off. When Expression is 50% (ie, 0x1FFF), then the volume is cut in half.

Here's how Expression is typically used. Let's assume only the coarse adjust is used (ie, #11) and therefore only 128 steps are possible. Set the Expression for every MIDI channel to one initial value, for example 100. This gives you some leeway to increase the expression percentage (ie, up to 127 which is 100%) or decrease it. Now, set the channel (ie, instrument) "mix" using Volume Controllers. Maybe you'll want the drums louder than the piano, so the former has a Volume Controller value of 110 whereas the latter has a value of 90, for example. Now if, at some point, you want to drop the volumes of both instruments to half of their current Main Volumes, then send Expression values of 64 (ie, 64 represents a 50% volume percentage since 64 is half of 128 steps). This would result in the drums now having an effective volume of 55 and the piano having an effective volume of 45. If you wanted to drop the volumes to 25% of their current Main Volumes, then send Expression values of 32. This would result in the drums now having an effective volume of approximately 27 and the piano having an effective volume of approximately 22. And yet, you haven't had to change their Volume settings, and therefore still maintain that relative mix between the two instruments. So think of Volume Controllers as

being the individual faders upon a mixing console. You set up the instrumental balance (ie, mix) using these values. Then you use Expression Controllers as "group faders", whereby you can increase or decrease the volumes of one or more tracks without upsetting the relative balance between them.

If a MultiTimbral device, then each Part usually has its own Expression level.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum effect.

Note: Most all devices ignore the Fine adjust (#43) for Expression, and just implement Coarse adjust (#11) because 14-bit resolution isn't needed for this.

Effect Control 1

Number: 12 (coarse) 44 (fine)

Affects:

This can control any parameter relating to an effects device, such as the Reverb Decay Time for a reverb unit built into a GM sound module.

Note: There are separate controllers for setting the Levels (ie, volumes) of Reverb, Chorus, Phase Shift, and other effects. This controller would be used for some effects parameter for which there is not already another defined controller.

If a MultiTimbral device, then each Part usually has its own Effect Control 1.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum effect.

Effect Control 2

Number: 13 (coarse) 45 (fine)

Affects:

This can control any parameter relating to an effects device, such as the Reverb Decay Time for a reverb unit built into a GM sound module.

Note: There are separate controllers for setting the Levels (ie, volumes) of Reverb, Chorus, Phase Shift, and other effects. This controller would be used for some effects parameter for which there is not already another defined controller.

If a MultiTimbral device, then each Part usually has its own Effect Control 2.

Value Range:

14-bit coarse/fine resolution. 0x0000 to 0x3FFF where 0 is minimum effect.

General Purpose Slider

Number: 16, 17, 18, 19

Affects:

Whatever the musician sets this controller to affect. There are 4 General Purpose Sliders, with the above controller numbers. Often, these are used to control parameters such as what Aftertouch can. If a MultiTimbral device, then each Part usually has its own responses to the 4 General Purpose Sliders. Note that these sliders don't have a fine adjustment.

Value Range:

0x00 to 0x7F where 0 is minimum effect.

Hold Pedal

Number: 64

Affects:

When on, this holds (ie, sustains) notes that are playing, even if the musician releases the notes. (ie, The Note Off effect is postponed until the musician switches the Hold Pedal off). If a MultiTimbral device, then each Part usually has its own Hold Pedal setting.

Note: When on, this also postpones any All Notes Off controller message on the same channel.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Portamento (On/Off)

Number: 65

Affects:

Whether the portamento effect is on or off. If a MultiTimbral device, then each Part usually has its own portamento on/off setting.

Note: There is another controller to set the Portamento Time.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Sustenuto

Number: 66

Affects:

Like the Hold Pedal controller, except this only sustains notes that are already on (ie, the device has received Note On messages, but the respective Note Offs haven't yet arrived) when the pedal is turned on. After the pedal is on, it continues to hold these initial notes all of the while that the pedal is on, but during that time, all other arriving Note Ons are not held. So, this pedal implements a "chord hold" for the notes that are sounding when this pedal is turned on. If a MultiTimbral device, then each Part usually has its own Sustenuto setting.

Note: When on, this also postpones any All Notes Off controller message on the same channel for those notes being held.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Soft Pedal

Number: 67

Affects:

When on, this lowers the volume of any notes played. If a MultiTimbral device, then each Part usually has its own Soft Pedal setting.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Legato Pedal

Number: 68

Affects:

When on, this causes a legato effect between notes, which is usually achieved by skipping the attack portion of the VCA's envelope. Use of this controller allows a keyboard player to better simulate the phrasing of wind and brass players, who often play several notes with a single tonguing, or simulate guitar pull-offs and hammer-ons (ie, where secondary notes are not picked). If a MultiTimbral device, then each Part usually has its own Legato Pedal setting.

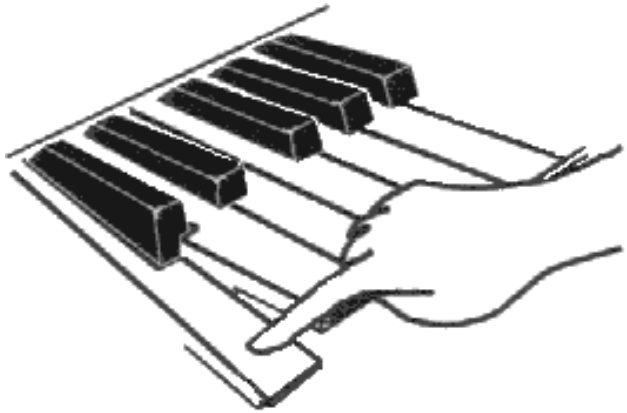
Value Range:

0 (to 63) is off. 127 (to 64) is on.

Preface

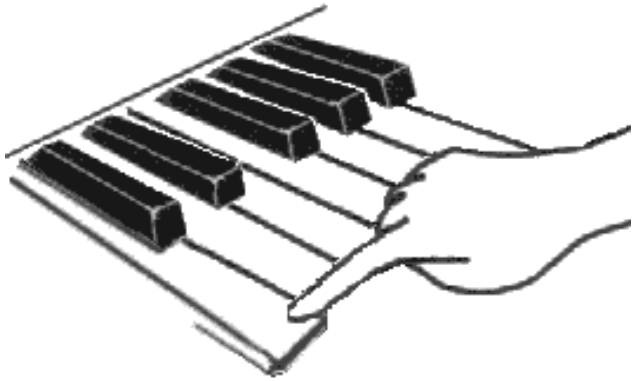
Before discussing what MIDI is, it is important to understand some basic principles about musical instruments.

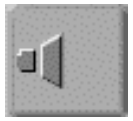
There is one thing that all musical instruments do. All musical instruments make a sound under the control of a musician. In other words, at any time, the musician can cause an instrument to start making a sound. For example, a musician can push down a key on a piano to start a sound. Or, he can begin dragging a bow across a violin string to start a sound. Or he can fret and pick a guitar string to start a sound. Let's refer to the action of starting a sound as a "Note On".



A musician pushes down (and holds down) a key on a keyboard. This sounds some musical note (which continues to sound while the musician continues holding down the key). This single gesture of the musician is known as a Note-On to MIDI.

Most instruments also allow the musician to stop the sound at any given time. For example, the musician can release that piano key, thus stopping the sound. Or, he can stop dragging a bow across the violin string. Or he can release his finger from the guitar fret. Let's refer to the action of stopping a sound as a "Note Off".





The musician releases the key (that he was holding down) on a keyboard. This stops the musical note from sounding. This single gesture of the musician is known as a Note-Off to MIDI.

Of course, many instruments can play distinct pitches (ie, a musical scale). For example, an acoustic piano has 88 keys, or 88 distinct pitches/notes.

There are other things that many musical instruments may have in common, for example, most instruments can make sounds at various volumes. (ie, They can sound notes at volumes ranging from very soft to very loud). For example, if the pianist pushes down a key with great force, the resulting note will be louder than if he were to gently press down the key.

Introduction

Musicians often want to be able to control electronic instruments remotely or automatically.

Remote control is when a musician plays one musical instrument, and that instrument controls (one or more) other musical instruments.

For example, musicians sometimes find it desirable to combine the sounds of several instruments playing in perfect unison to "thicken" or layer a musical part. The musician wants to blend certain patches upon those instruments. Perhaps he wishes to blend the sax patches upon 5 different instruments to create a more authentic-sounding sax section in a big band. But, since a musician has only two hands and feet, it's not possible to play 5 instruments at once unless he has some method of remote control.

Or, sometimes a musician wants to use only one physical keyboard to control several, separate sound modules. In the old days, every single musical instrument manufactured had its own built-in method of controlling it. For example, an electronic organ, an electronic piano, a string ensemble, a synthesizer, etc, each had its own built-in keyboard. This got to be rather expensive, as the physical keyboard is one of the more expensive parts of an instrument. Also, all of those keyboards tend to take up a lot of space, which is a problem for a gigging musician. So musicians thought *"Wouldn't it be great if I could buy a small box that made organ sounds into which I could plug a physical keyboard? And wouldn't it be great if I could buy other boxes that made piano, string, synth, etc, sounds, into which I could plug that same keyboard? And wouldn't it be great if I could attach them all together simultaneously, and switch the keyboard between playing any of them? I could save money and space. All I need is a standard for remotely controlling all of those boxes with that one keyboard."*

Automatic control is when the musician uses some other device to play a musical instrument as if another musician were playing it. (Such a device is referred to as a Sequencer).

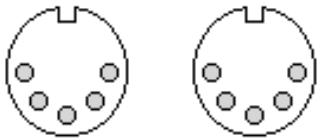
For example, some musicians want to be able to have "backing tracks" in live performance, but they found it too cumbersome, unreliable, and limiting to use prerecorded tapes. They wanted a method that allowed more flexibility, perhaps to do things such as subtlely alter the arrangement live. To achieve this, rather

than playing pre-recorded backing tracks, they wanted a method to automatically control their instruments during the performance using a device that could "intelligently" manipulate the arrangement (such as a computer).

So, musicians had a need to remotely or automatically control their musical instruments, and they wanted a method that wasn't tied to one particular manufacturer's product, nor one particular type of instrument. (ie, They wanted a method that worked as well with an electronic piano as it did with a drum box, for example). They wanted a standard that could be useful in controlling any electronic musical device. To satisfy this need, a few music manufacturers got together in mid 1983 and created **MIDI**, which **stands for Musical Instrument Digital Interface**. (For more information about the history of MIDI's development, see The beginnings of MIDI).

Hardware/Connections

The visible **MIDI connectors** on an instrument **are female 5-pin DIN jacks**. There are separate jacks for incoming MIDI signals (received from another instrument that is sending MIDI signals), and outgoing MIDI signals (ie, MIDI signals that the instrument creates and sends to another device). The jacks look like these:



Midi In Midi Out

You use **MIDI cables** (with male DIN connectors) to **connect the MIDI jacks of various instruments together**, so that those instruments can pass MIDI signals to each other. You connect the MIDI OUT of one instrument to the MIDI IN of another instrument, and vice versa. For example, the following diagram shows the connection between a computer's MIDI interface and a MIDI keyboard that has built-in sounds.

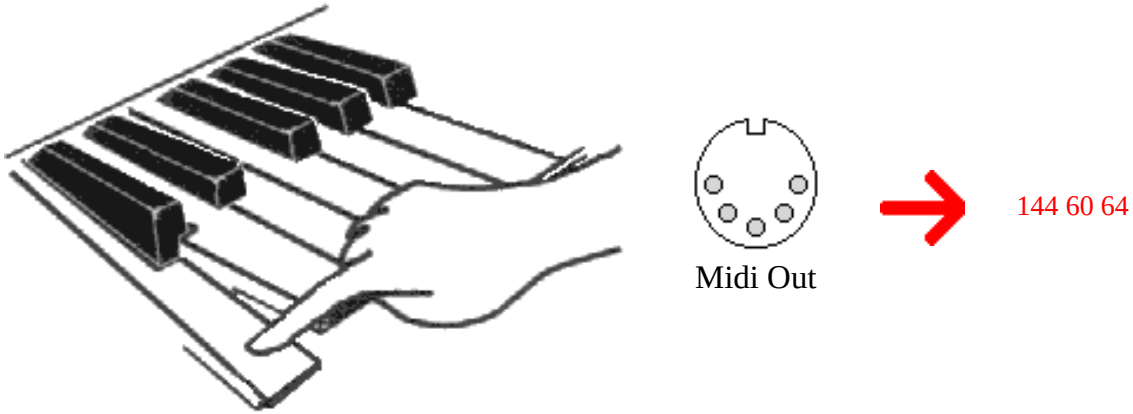
Some instruments have a third MIDI jack labeled "Thru". This is used as if it were an OUT jack, and therefore you attach a THRU jack only to another instrument's IN jack. In fact, the THRU jack is exactly like the OUT jack with one important difference. Any signals that the instrument itself creates (or modifies) are sent out its MIDI OUT jack but not the MIDI THRU jack. Think of the THRU jack as a stream-lined, unprocessed MIDI OUT jack.

MIDI messages

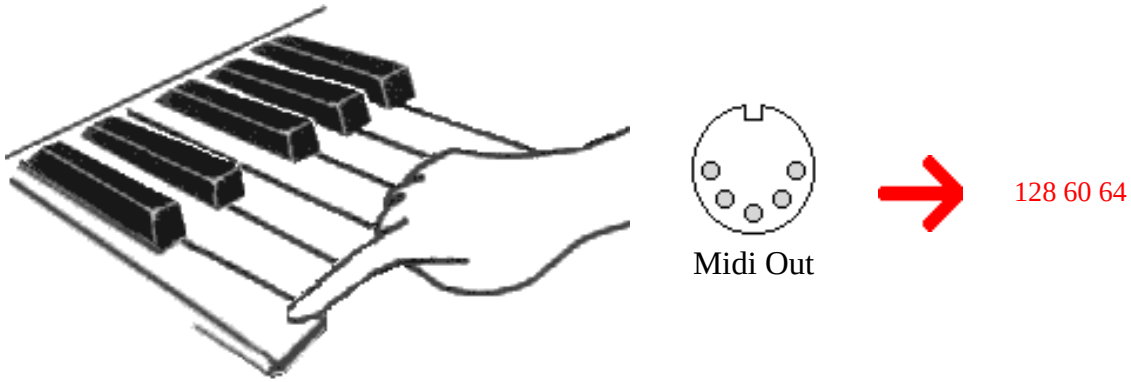
But MIDI is much more than just some jacks on an electronic instrument. In fact, **MIDI is a lot more than just hardware. Mostly, MIDI is an extensive set of "musical commands" which electronic instruments use to control each other**. The MIDI instruments pass these commands to each other over the cables connecting their MIDI jacks together. (ie, Those MIDI signals that I referred to above are these commands).

So, what is a MIDI command? **A MIDI command consists of a few (usually 2 or 3) "data bytes"** (like the data bytes within files that you have on your computer's hard drive). **These data bytes are merely a series of numbers. We refer to one of these groups of numbers as a "message"** (rather than a command). There are many different MIDI messages, and **each one correlates to a specific musical action**. For example, there is a certain group of numbers that tells an instrument to make a sound. (This would be that "Note On" message which I mentioned earlier). There is a different group of numbers that tells an instrument to stop making a sound. (This is the "Note Off" message). One of the numbers within that "Note On" or "Note Off" message tells the instrument which one of its "keys" (ie, notes) to start or stop sounding. (Remember that a piano has 88 notes. MIDI instruments can have a maximum of 128 different notes, although some instruments respond to only messages limited to a smaller range, say 72 notes).

Many electronic instruments not only respond to MIDI messages that they receive (at their MIDI IN jack), they also automatically generate MIDI messages while the musician plays the instrument (and send those messages out their MIDI OUT jacks).



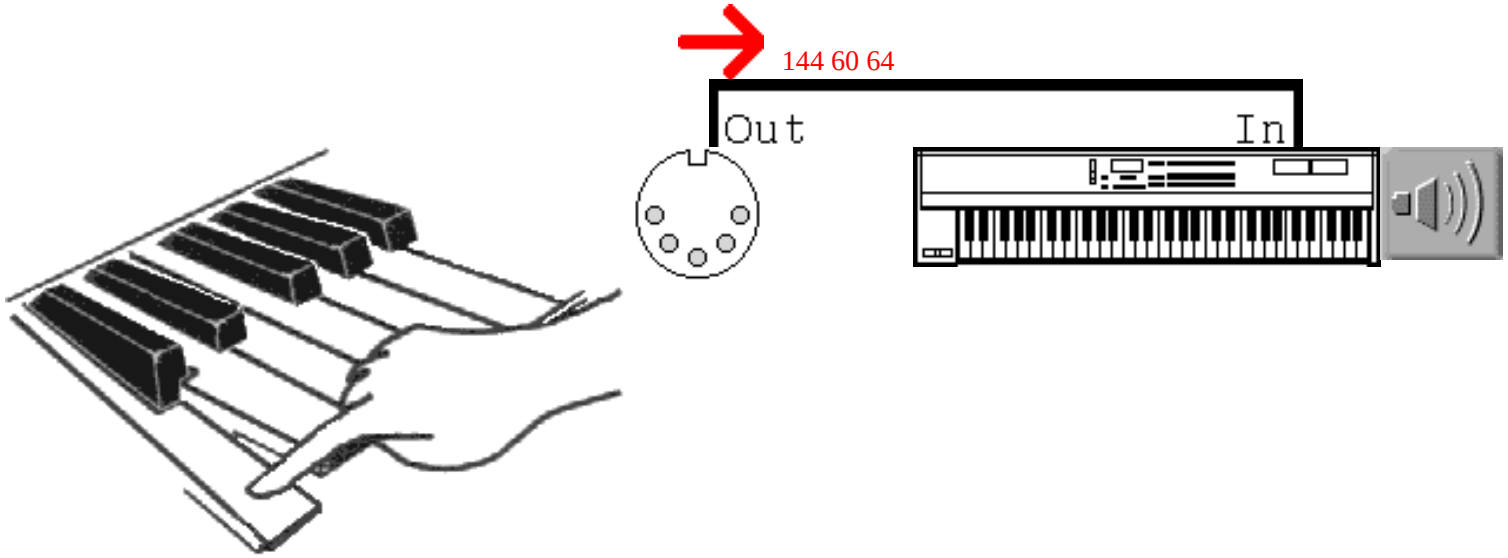
A musician pushes down (and holds down) the middle C key on a keyboard. Not only does this sound a musical note, it also causes a MIDI Note-On message to be sent out of the keyboard's MIDI OUT jack. That message consists of 3 numeric values as shown above.



The musician now releases that middle C key. Not only does this stop sounding the musical note, it also causes another message -- a MIDI Note-Off message -- to be sent out of the keyboard's MIDI OUT jack. That message consists of 3 numeric values as shown above. Note that one of the values is different than the Note-On message.

You saw above that when the musician pushed down that middle C note, the instrument sent a MIDI Note On message for middle C out of its MIDI OUT jack. If you were to connect a second instrument's MIDI IN jack to the first instrument's MIDI OUT, then the second instrument would "hear" this MIDI message and sound its middle C too. When the musician released that middle C note, the first instrument would send out a MIDI Note Off message for that middle C to

the second instrument. And then the second instrument would stop sounding its middle C note.



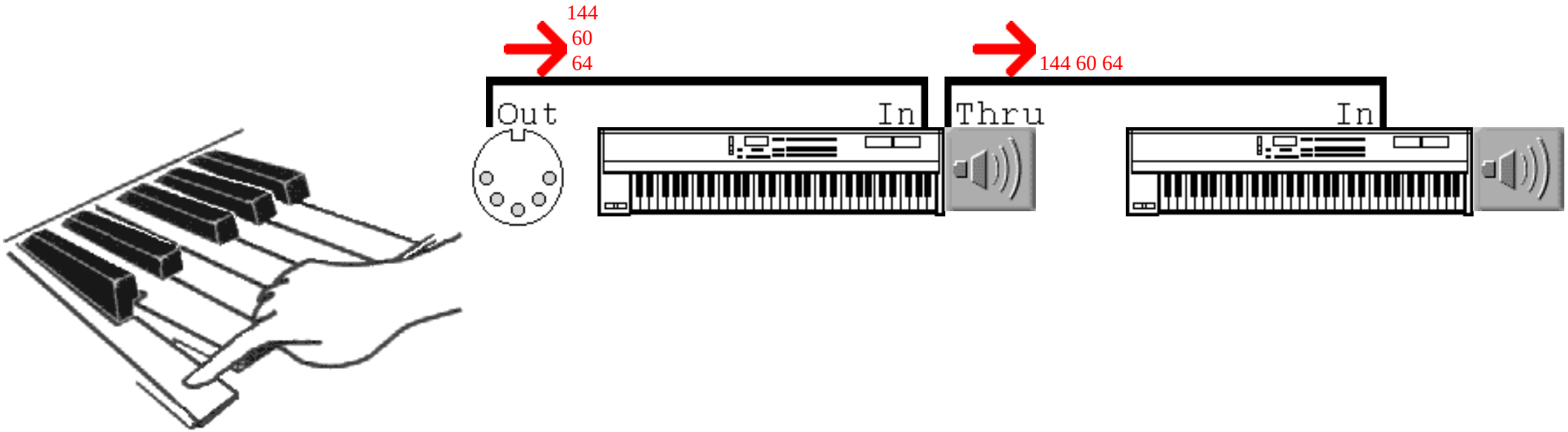
A musician pushes down (and holds down) the middle C key on a keyboard. This causes a MIDI Note-On message to be sent out of the keyboard's MIDI OUT jack. That message is received by the second instrument which sounds its middle C in unison.

But MIDI is more than just "Note On" and "Note Off" messages. There are lots more messages. There's a message that tells an instrument to move its pitch wheel and by how much. There's a message that tells the instrument to press or release its sustain pedal. There's a message that tells the instrument to change its volume and by how much. There's a message that tells the instrument to change its patch (ie, maybe from an organ sound to a guitar sound). And of course, these are only a few of the many available messages in the MIDI command set.

And just like with Note On and Note Off messages, these other messages are automatically generated when a musician plays the instrument. For example, if the musician moves the pitch wheel, a pitch wheel MIDI message is sent out of the instrument's MIDI OUT jack. (Of course, the pitch wheel message is a different group of numbers than either the Note On or Note Off messages). What with all of the possible MIDI messages, everything that the musician did upon the first instrument would be echoed upon the second instrument. It would be like he had two left and two right hands that worked in perfect sync.

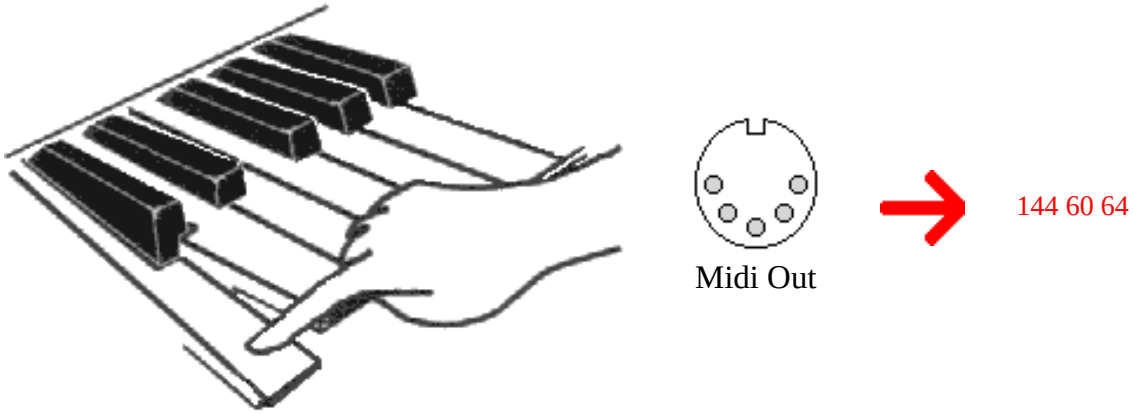
Daisy-chaining/MIDI Channels

You can attach a MIDI cable from the second instrument's MIDI THRU to a third instrument's MIDI IN, and the second instrument will pass onto the third instrument those messages that the first instrument sent. Now, all 3 instruments can play in unison. You could add a fourth, fifth, sixth, etc, instrument. **We call this "daisy-chaining" instruments**.

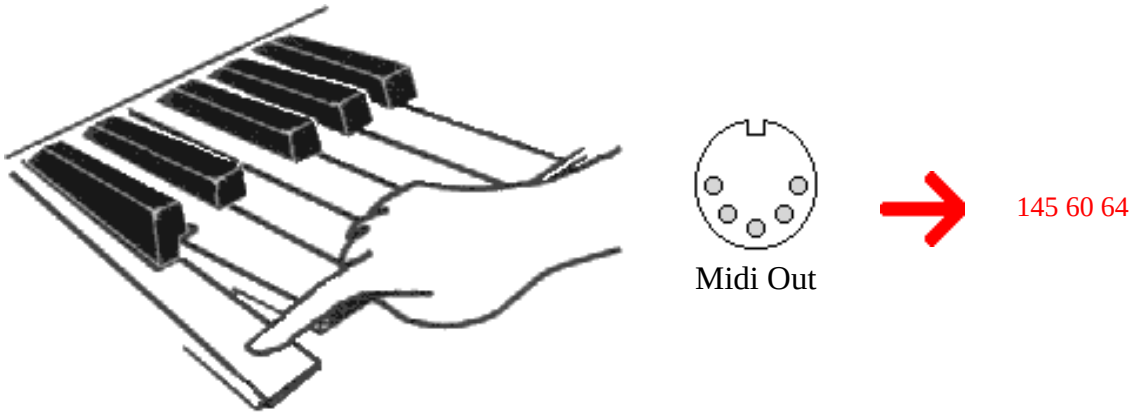


A musician pushes down (and holds down) the middle C key on a keyboard. This causes a MIDI Note-On message to be sent out of the keyboard's MIDI OUT jack. That message is received by the second instrument which sounds its middle C in unison. It is also passed from the second instrument to the third instrument which also sounds its middle C note.

But, daisy-chained instruments don't always have to play in unison either. Each can play its own, individual musical part even though all of the MIDI messages controlling those daisy-chained instruments pass through each instrument. How is this possible? **There are 16 MIDI "channels". They all exist in that one run of MIDI cables that daisy-chain 2 or more instruments** (and perhaps a computer) together. For example, that MIDI message for the middle C note can be sent on channel 1. Or, it can be sent on channel 2. Etc.



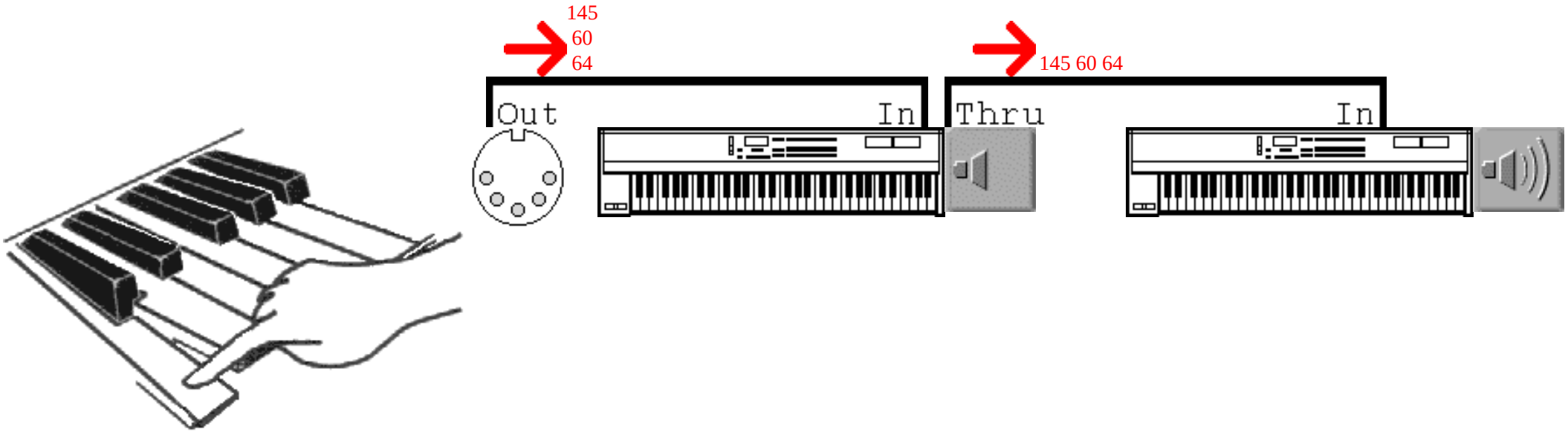
A musician sets his keyboard to send messages upon MIDI channel 1. Then, he pushes down the middle C key. Note the 3 values of the MIDI Note-On message sent out of the keyboard's MIDI OUT jack.



The musician changes the keyboard to send messages upon MIDI channel 2 instead. Then, he pushes down the middle C key. Compare the 3 values of this MIDI Note-On message to the preceding example. Note that the first value is different after the MIDI channel has been changed from 1 to 2.

Most all MIDI instruments allow the musician to select which channel(s) to respond to and which to ignore. For example, if you set an instrument to respond to MIDI messages only on channel 1, and you send a MIDI Note On on channel 2, then the instrument will not play the note. So, if a musician has several

instruments daisy-chained, he can set them all to respond to different channels and therefore have independent control over each one.



A musician sets his keyboard to send messages upon MIDI channel 2. Then, he pushes down the middle C key. This causes a MIDI Note-On message on MIDI channel 2 to be sent out of the keyboard's MIDI OUT jack. The second instrument is set to MIDI channel 1, so it ignores this MIDI message. (ie, It doesn't play its middle C note). The message is also passed on to the third instrument. This is set to MIDI channel 2, so it does play its middle C note.

Also, when the musican plays each instrument, it generates MIDI messages only on its one channel. So, it's very easy to keep the MIDI data separate for each instrument even though it all goes over one long run of cables. After all, the MIDI Note On message for middle C on channel 1 will be slightly different than the MIDI Note On message for middle C on channel 2.

In a nutshell, that's what MIDI is -- a set of commands that electronic devices digitally pass between their MIDI jacks to tell each other what to do (ie, what actions to perform).

Suffice it to say that MIDI can do everything musical to an electronic instrument that a human being can physically do, and a few things that humans can't do.

The advantages of MIDI

There are two main advantages of MIDI -- it's an easily edited/manipulated form of data, and also it's a compact form of data (ie, produces relatively small data files).

Because MIDI is a digital signal, it's very easy to interface electronic instruments to computers, and then do things with that MIDI data on the computer with software. For example, software can store MIDI messages to the computer's disk drive. Also, the software can playback MIDI messages upon all 16 channels with the same rhythms as the human who originally caused the instrument(s) to generate those messages. So, a musician can digitally record his musical performance and store it on the computer (to be played back by the computer). He does this not by digitizing the actual audio coming out of all of his electronic instruments, but rather by "recording" the MIDI OUT (ie, those MIDI messages) of all of his instruments. Remember that the MIDI messages for all of those instruments go over one run of cables, so if you put the computer at the end, it "hears" the messages from all instruments over just one incoming cable. The great advantage of MIDI is that the "notes" and other musical actions, such as moving the pitch wheel, pressing the sustain pedal, etc, are all still separated by messages on different channels. So the musician can store the messages generated by many instruments in one file, and yet the messages can be easily pulled apart on a per instrument basis because each instrument's MIDI messages are on a different MIDI channel. In other words, when using MIDI, a musician never loses control over every single individual action that he made upon each instrument, from playing a particular note at a particular point, to pushing the sustain pedal at a certain time, etc. The data is all there, but it's put together in such a way that every single musical action can be easily examined and edited.

Contrast this with digitizing the audio output of all of those electronic instruments. If you've got a system that has 16 stereo digital audio tracks, then you can keep each instrument's output separate. But, if you have only 2 digital audio tracks (typically), then you've got to mix the audio signals together before you digitize them. Those instruments' audio outputs don't produce digital signals. They're analog. Once you mix the analog signals together, it would take massive amounts of computation to later filter out separate instruments, and the process would undoubtably be far from perfect. So ultimately, you lose control over each instrument's output, and if you want to edit a certain note of one instrument's part, that's even less feasible.

Furthermore, it typically takes much more storage to digitize the audio output of an instrument than it does to record an instrument's MIDI messages. Why? Let's take an example. Say that you want to record a whole note. With MIDI, there are only 2 messages involved. There's a Note On message when you sound the note, and then the next message doesn't happen until you finally release the note (ie, a Note Off message). That's 6 bytes. In fact, you could hold down that note for an hour, and you're still going to have only 6 bytes; a Note On and a Note Off message. By contrast, if you want to digitize that whole note, you have to be recording all of the time that the note is sounding. So, for the entire time that you hold down the note, the computer is storing literally thousands of bytes of "waveform" data representing the sound coming out of the instrument's AUDIO OUT. You see, with MIDI a musician records his actions (ie, movements). He presses the note down. Then, he does nothing until he releases the note. With digital audio, you record the instrument's sound. So while the instrument is making sound, it must be recorded.

So why not always "record" and "play" MIDI data instead of WAVE data if the former offers so many advantages? OK, for electronic instruments that's a great idea. But what if you want to record someone singing? You can strip search the person, but you're not going to find a MIDI OUT jack on his body. (Of course,

I anxiously await the day when scientists will be able to offer "human MIDI retrofits". I'd love to have a built-in MIDI OUT jack on my body, interpreting every one of my motions and thoughts into MIDI messages. I'd have it installed at the back of my neck, beneath my hairline. Nobody would ever see it, but when I needed to use it, I'd just push back my hair and plug in the cable). So, to record that singing, you're going to have to record the sound, and digitizing it into a WAVE file is the best digital option right now. That's why sequencer programs exist that record and play both MIDI and WAVE data, in sync.

Hold 2 Pedal

Number: 69

Affects:

When on, this lengthens the release time of the playing notes' VCA (ie, makes the note take longer to fade out after it's released, versus when this pedal is off). Unlike the other Hold Pedal controller, this pedal doesn't permanently sustain the note's sound until the musician releases the pedal. Even though the note takes longer to fade out when this pedal is on, the note may eventually fade out despite the musician still holding down the key and this pedal. If a MultiTimbral device, then each Part usually has its own Hold 2 Pedal setting.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Sound Variation

Number: 70

Affects:

Any parameter associated with the circuitry that produces sound. For example, if a device uses looped digital waveforms to create sound, this controller may adjust the sample rate (ie, playback speed), for a "tuning" control. If a MultiTimbral device, then each Part usually has its own patch with its respective VCA, VCF, tuning, sound sources, etc, parameters that can be adjusted with this controller.

Note: There are other controllers for adjusting VCA attack and release times, VCF cutoff frequency, and other generic sound parameters.

Value Range:

0 to 127, with 0 being minimum setting.

Sound Timbre

Number: 71

Affects:

Controls the (VCF) filter's envelope levels. This controls how the filter shapes the "brightness" of the sound over time). If a MultiTimbral device, then each Part usually has its own patch with its respective VCF cutoff frequency that can be adjusted with this controller.

Note: There are other controllers for adjusting VCA attack and release times, and other generic sound parameters.

Value Range:

0 to 127, with 0 being minimum setting.

Sound Release Time

Number: 72

Affects:

Controls the (VCA) amp's envelope release time, for a control over how long it takes a sound to fade out. If a MultiTimbral device, then each Part usually has its own patch with its respective VCA envelope that can be adjusted with this controller.

Note: There are other controllers for adjusting VCA attack time, VCF cutoff frequency, and other generic sound parameters.

Value Range:

0 to 127, with 0 being minimum setting.

Sound Attack Time

Number: 73

Affects:

Controls the (VCA) amp's envelope attack time, for a control over how long it takes a sound to fade in. If a MultiTimbral device, then each Part usually has its own patch with its respective VCA envelope that can be adjusted with this controller.

Note: There are other controllers for adjusting VCA release time, VCF cutoff frequency, and other generic sound parameters.

Value Range:

0 to 127, with 0 being minimum setting.

Sound Brightness

Number: 74

Affects:

Controls the (VCF) filter's cutoff frequency, for an overall "brightness" control. If a MultiTimbral device, then each Part usually has its own patch with its respective VCF cutoff frequency that can be adjusted with this controller.

Note: There are other controllers for adjusting VCA attack and release times, and other generic sound parameters.

Value Range:

0 to 127, with 0 being minimum setting.

Sound Control 6, 7, 8, 9, 10

Number: 75, 76, 77, 78, 79

Affects:

These 5 controllers can be used to adjust any parameters associated with the circuitry that produces sound. For example, if a device uses looped digital waveforms to create sound, one controller may adjust the sample rate (ie, playback speed), for a "tuning" control. If a MultiTimbral device, then each Part usually has its own patch with its respective VCA, VCF, tuning, sound sources, etc, parameters that can be adjusted with these controllers.

Note: There are other controllers for adjusting VCA attack and release times, and VCF cutoff frequency. These controllers would be used to adjust sound parameters for which there are not already other, defined controllers.

Value Range:

0 to 127, with 0 being minimum setting.

General Purpose Button

Number: 80, 81, 82, 83

Affects:

Whatever the musician sets this controller to affect. There are 4 General Purpose Buttons, with the above controller numbers. These are either on or off, so they are often used to implement on/off functions, such as a Metronome on/off switch on a sequencer. If a MultiTimbral device, then each Part usually has its own responses to the 4 General Purpose Buttons.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

Effects Level

Number: 91

Affects:

The effects amount (ie, level) for the device. Often, this is the reverb or delay level. If a MultiTimbral device, then each Part usually has its own effects level.

Value Range:

0 to 127, with 0 being no effect applied at all.

Tremulo Level

Number: 92

Affects:

The tremulo amount (ie, level) for the device. If a MultiTimbral device, then each Part Parts usually has its own tremulo level.

Value Range:

0 to 127, with 0 being no tremulo applied at all.

Chorus Level

Number: 93

Affects:

The chorus effect amount (ie, level) for the device. If a MultiTimbral device, then each Part usually has its own chorus level.

Value Range:

0 to 127, with 0 being no chorus effect applied at all.

Celeste Level

Number: 94

Affects:

The celeste (detune) amount (ie, level) for the device. If a MultiTimbral device, then each Part usually has its own celeste level.

Value Range:

0 to 127, with 0 being no celeste effect applied at all.

Phaser Level

Number: 95

Affects:

The Phaser effect amount (ie, level) for the device. If a MultiTimbral device, then each Part usually has its own Phaser level.

Value Range:

0 to 127, with 0 being no phaser effect applied at all.

Data Button increment

Number: 96

Affects:

Causes a Data Button to increment (ie, increase by 1) its current value. Usually, this data button's value is being used to set some Registered or Non-Registered Parameter. Which RPN or NRPN parameter is being affected depends upon a preceding RPN or NRPN message (which itself identifies the parameter's number).

Value Range:

The value byte isn't used and defaults to 0.

Data Button decrement

Number: 97

Affects:

Causes a Data Buttonto decrement (ie, decrease by 1) its current value. Usually, this data button's value is being used to set some Registered or Non-Registered Parameter. Which RPN or NRPN parameter is being affected depends upon a preceding RPN or NRPN message (which itself identifies the parameter's number).

Value Range:

The value byte isn't used and defaults to 0.

Non-Registered Parameter Number (NRPN)

Number: 99 (coarse) 98 (fine)

Affects:

Which parameter the Data Button Increment, Data Button Decrement, or Data Entry controllers affect. Since NRPN has a coarse/fine pair (14-bit), the number of parameters that can be registered is 16,384. That's a lot of parameters that a MIDI device could allow to be controlled over MIDI. It's entirely up to each manufacturer which parameter numbers are used for whatever purposes. These don't have to be registered with the MMA.

Value Range:

The same scheme is used as per the Registered Parameter controller. Refer to that. By contrast, the coarse/fine messages for NRPN for the preceding RPN example would be:

```
B0 63 00
B0 62 01
```

Note: Since each device can define a particular NRPN controller number to control anything, it's possible that 2 devices may interpret the same NRPN number in different manners. Therefore, a device should allow a musician to disable receipt of NRPN, in the event that there is a conflict between the NRPN implementations of 2 daisy-chained devices.

Registered Parameter Number (RPN)

Number: 101 (coarse) 100 (fine)

Affects:

Which parameter the Data Button Increment, Data Button Decrement, or Data Entry controllers affect. Since RPN has a coarse/fine pair (14-bit), the number of parameters that can be registered is 16,384. That's a lot of parameters that a MIDI device could allow to be controlled over MIDI. It's up to the MMA to assign Registered Parameter Numbers to specific functions.

Value Range:

0 to 16,384 where each value stands for a different RPN. Here are the currently registered parameter numbers:

Pitch Bend Range (ie, Sensitivity) 0x0000

> **Note:** The coarse adjustment (usually set via Data Entry 6) sets the range in semitones. The fine adjustment (usually set via Data Entry 38) set the range in cents. For example, to adjust the pitch wheel range to go up/down 2 semitones and 4 cents:

```
B0 65 00  Controller/chan 0, RPN coarse (101), Pitch Bend Range
B0 64 00  Controller/chan 0, RPN fine (100), Pitch Bend Range
B0 06 02  Controller/chan 0, Data Entry coarse, +/- 2 semitones
B0 26 04  Controller/chan 0, Data Entry fine, +/- 4 cents
```

Master Fine Tuning (ie, in cents) 0x0001

> **Note:** Both the coarse and fine adjustments together form a 14-bit value that sets the tuning in semitones, where 0x2000 is A440 tuning.

Master Coarse Tuning (in half-steps) 0x0002

> **Note:** Setting the coarse adjustment adjusts the tuning in semitones,

where 0x40 is A440 tuning. There is no need to set a fine adjustment.

RPN Reset 0x3FFF

Note: No coarse or fine adjustments are applicable. This is a "dummy" parameter.

Here's the way that you use RPN. First, you decide which RPN you wish to control. Let's say that we wish to set Master Fine Tuning on a device. That's RPN 0x0001. We need to send the device the RPN Coarse and RPN Fine controller messages in order to tell it to affect RPN 0x0001. So, we divide the 0x0001 into 2 bytes, the fine byte and the coarse byte. The fine byte contains bits 0 to 6 of the 14-bit value. The coarse byte contains bits 7 to 13 of the 14-bit value, right-justified. So, here are the RPN Coarse and Fine messages (assuming that the device is responding to MIDI channel 0):

```
B0 65 00  Controller/chan 0, RPN coarse (101), bits
            7 to 13 of 0x0001, right-justified (with high bit clear)
B0 64 01  Controller/chan 0, RPN fine (100), bits
            0 to 6 of 0x0001, (with high bit clear)
```

Now, we've just told the device that any Data Button Increment, Data Button decrement, or Data Entry controllers it receives should affect the Master Fine Tuning. Let's say that we wish to set this tuning to the 14-bit value 0x2000 (which happens to be centered tuning). We could use the Data Entry (coarse and fine) controller messages as so to send that 0x2000:

```
B0 06 40  Controller/chan 0, Data Entry coarse (6), bits
            7 to 13 of 0x2000, right-justified (with high bit clear)
B0 26 00  Controller/chan 0, Data Entry fine (38), bits
            0 to 6 of 0x2000, (with high bit clear)
```

As a final example, let's say that we wish to increment the Master Fine Tuning by one (ie, to 0x2001). We could use the Data Entry messages again. Or, we could use the Data Button Increment, which doesn't have a coarse/fine pair of controller numbers like Data Entry.

```
B0 60 00  Controller/chan 0, Data Button Increment (96),
```

```
last byte is unused
```

Of course, if the device receives RPN messages for another parameter, then the Data Button Increment, Data Button Decrement, and Data Entry controllers will switch to adjusting that parameter.

RPN 0x3FFF (reset) forces the Data Button increment, Data Button decrement, and Data Entry controllers to not adjust any RPN (ie, disables those buttons' adjustment of any RPN).

All Sound Off

Number: 120

Affects:

Mutes all sounding notes that were turned on by received Note On messages, and which haven't yet been turned off by respective Note Off messages. This message is not supposed to mute any notes that the musician is playing on the local keyboard. So, if a device can't distinguish between notes played via its **MIDI IN** and notes played on the local keyboard, it should not implement All Sound Off.

Note: The difference between this message and All Notes Off is that this message immediately mutes all sound on the device regardless of whether the Hold Pedal is on, and mutes the sound quickly regardless of any lengthy VCA release times. It's often used by sequencers to quickly mute all sound when the musician presses "Stop" in the middle of a song.

Value Range:

The value byte isn't used and defaults to 0.

All Controllers Off

Number: 121

Affects:

Resets specific controllers to default states. In general, switches (such as Hold Pedal) are turned off, and continuous controllers (such as Mod Wheel) are set to minimum positions. If the device is MultiTimbral, this only affects the Part assigned to the MIDI channel upon which this message is received.

For a minimum implementation of "All Controllers off", it is recommended that the following controllers should be reset to the following values:

Modulation Wheel should be set to 0.

Expression should be set to full value (ie, 127 for both coarse, as well as fine if used).

The pedals Hold, Portamento, Sustenuto, Soft, Legato, and Hold 2 should be set to 0 (ie, Off).

Volume should be set to 100.

Pan should be set to center (ie, 64 for both coarse, as well as fine if used).

Any Non-registered Parameters or Registered Parameters may be reset to default values, such as pitch bend range being reset to +/- 2 semitones.

Additionally, the Pitch Wheel should be reset to center position, and Channel Pressure should be set to 0, although these technically aren't adjusted via MIDI controller messages. If a module supports Aftertouch, then the Aftertouch amount for each note (on the MIDI channel) should also be reset to 0.

A device should not reset its mode (as done via Omni Mode Off, Omni Mode On, Mono Operation, and Poly Operation controllers).

A device should include documentation that lists what is reset by an All Controllers Off message.

Value Range:

The value byte isn't used and defaults to 0.

Local Keyboard on/off

Number: 122

Affects:

Turns the device's keyboard on or off locally. If off, the keyboard is disconnected from the device's internal sound generation circuitry. So when the musician presses keys, the device doesn't trigger any of its internal sounds. But, the keyboard still generates Note On, Note Off, Aftertouch, and Channel Pressure messages. In this way, a musician can eliminate a situation where MIDI messages get looped back (over MIDI cables) to the device that created those messages. Furthermore, if a device is only going to be played remotely via MIDI, then the keyboard may be turned off in order to allow the device to concentrate more on dealing with MIDI messages rather than scanning the keyboard for depressed notes and varying pressure.

Value Range:

0 (to 63) is off. 127 (to 64) is on.

All Notes Off

Number: 123

Affects:

Turns off all notes that were turned on by received Note On messages, and which haven't yet been turned off by respective Note Off messages. This message is not supposed to turn off any notes that the musician is playing on the local keyboard. So, if a device can't distinguish between notes played via its **MIDI IN** and notes played on the local keyboard, it should not implement All Notes Off. Furthermore, if a device is in Omni On state, it should ignore this message on any channel.

Note: If the device's Hold Pedal controller is on, the notes aren't actually released until the Hold Pedal is turned off. See All Sound Off controller message for turning off the sound of these notes immediately.

Value Range:

The value byte isn't used and defaults to 0.

Omni Off

Number: 124

Affects:

Turns **Omni** off. See the discussion on MIDI Modes.

Value Range:

The value byte isn't used and defaults to 0.

Note: When a device receives an Omni Off message, it should automatically turn off all playing notes.

Omni On

Number: 125

Affects:

Turns **Omni** on. See the discussion on MIDI Modes.

Value Range:

The value byte isn't used and defaults to 0.

Note: When a device receives an Omni On message, it should automatically turn off all playing notes.

Monophonic Operation

Number: 126

Affects:

Enables **Monophonic operation** (thus disabling Polyphonic operation). See the discussion on MIDI Modes.

Value Range:

If Omni is off, this Value tells how many MIDI channels the device is expected to respond to in Mono mode. In other words, if Omni is off, this value is used to select a limited set of the 16 MIDI channels (ie, 1 to 16) to respond to.Conversely, if Omni is on, this Value is ignored completely, and the device only ever plays one note at a time (unless a MultiTimbral device). So, the following discussion is only relevant if Omni Off.

If Value is 0, then the number of MIDI channels that the device will respond to simultaneously will be equal to how many voices the device can sound simultaneously. In other words, if the device can sound at least 16 voices simultaneously, then it can respond to Voice Category messages on all 16 channels. Of course, being Monophonic operation, the device can only sound one note at a time per each MIDI channel. So, it can sound a note on channel 1 and channel 2 simultaneously, for example, but can't sound 2 notes both on channel 1 simultaneously.

Of course, MultiTimbral devices completely violate the preceding theory. MultiTimbral devices always can play polyphonically on each MIDI channel. If Value is 0, what this means is that the device can play as many MIDI channels as it has Parts. So, if the device can play 16 of its patches simultaneously, then it can respond to Voice Category messages on all 16 channels.

If Value is not 0 (ie, 1 to 16), then that's how many MIDI channels the device is allowed to respond to. For example, a value of 1 would mean that the device

would only be able to respond to 1 MIDI channel. Since the device is also limited to sounding only 1 note at a time on that MIDI channel, then the device would truly be a Monophonic instrument incapable of sounding more than one note at a time. If a device is asked to respond to more MIDI channels than it has voices to accomodate, then it will handle only as many MIDI channels as it has voices. For example, if an 8-voice synth, on Base Channel 0, receives the value 16 in the Mono message, then the synth will play messages on MIDI channels 0 to 7 and ignore messages on 8 to 15.

Again, MultiTimbral devices violate the above theory. A value of 1 would mean that the device would only be able to respond to 1 MIDI channel (and therefore only play 1 Part), but would do so Polyphonically. If a MultiTimbral device is asked to respond to more MIDI channels than it has Parts to accomodate, then it will handle only as many MIDI channels as it has Parts. For example, if a device can play only 5 Patches simultaneously, and receives the value 8 in the Mono message, then the device will play 5 patches on MIDI channels 0 to 4 and ignore messages on channels 5 to 7.

Most devices capable of Monophonic operation, allow the user to specify a **Base Channel**. This will be the lowest MIDI channel that the device responds to. For example, if a Mono message specifies that the device is to respond to only 2 channels, and its Base Channel is 2, then the device responds to channels 2 and 3.

Note: When a device receives a Mono Operation message, it should automatically turn off all playing notes.

Polyphonic Operation

Number: 127

Affects:

Enables **Polyphonic operation** (thus disabling Monophonic operation). See the discussion on MIDI Modes.

Value Range:

The value byte isn't used and defaults to 0.

Note: When a device receives a Poly Operation message, it should automatically turn off all playing notes.

Program Change

Category: Voice

Purpose

To cause the MIDI device to change to a particular **Program** (which some devices refer to as Patch, or Instrument, or Preset, or whatever). Most sound modules have a variety of instrumental sounds, such as Piano, and Guitar, and Trumpet, and Flute, etc. Each one of these instruments is contained in a Program. So, changing the Program changes the instrumental sound that the MIDI device uses when it plays Note On messages. Of course, other MIDI messages also may modify the current Program's (ie, instrument's) sound. But, the Program Change message actually selects which instrument currently plays. There are 128 possible program numbers, from 0 to 127. If the device is a MultiTimbral unit, then it usually can play 16 "Parts" at once, each receiving data upon its own MIDI channel. This message will then change the instrument sound for only that Part which is set to the message's MIDI channel.

For MIDI devices that don't have instrument sounds, such as a Reverb unit which may have several Preset "room algorithms" stored, the Program Change message is often used to select which Preset to use. As another example, a drum box may use Program Change to select a particular rhythm pattern (ie, drum beat).

Status

0xC0 to 0xCF where the low nibble is the MIDI channel.

Data

One data byte follows the status. It is the program number to change to, a number from 0 to 127.

Errata

On MIDI sound modules (ie, whose Programs are instrumental sounds), it

became desirable to define a standard set of Programs in order to make sound modules more compatible. This specification is called General MIDI Standard.

Just like with MIDI channels 0 to 15 being displayed to a musician as channels 1 to 16, many MIDI devices display their Program numbers starting from 1 (even though a Program number of 0 in a Program Change message selects the first program in the device). On the other hand, this approach was never standardized, and some devices use vastly different schemes for the musician to select a Program. For example, some devices require the musician to specify a bank of Programs, and then select one within the bank (with each bank typically containing 8 to 10 Programs). So, the musician might specify the first Program as being bank 1, number 1. Nevertheless, a Program Change of number 0 would select that first Program.

Receipt of a Program Change should not cut off any notes that were previously triggered on the channel, and which are still sustaining.

Channel Pressure

Category: Voice

Purpose

While notes are playing, pressure can be applied to all of them. Many electronic keyboards have pressure sensing circuitry that can detect with how much force a musician is holding down keys. The musician can then vary this pressure, even while he continues to hold down the keys (and the notes continue sounding). The Channel Pressure message conveys the amount of overall (average) pressure on the keys at a given point. Since the musician can be continually varying his pressure, devices that generate Channel Pressure typically send out many such messages while the musician is varying his pressure. Upon receiving Channel Pressure, many devices typically use the message to vary all of the sounding notes' VCA and/or VCF envelope sustain levels, or control LFO amount and/or rate being applied to the notes' sound generation circuitry. But, it's up to the device how it chooses to respond to received Channel Pressure (if at all). If the device is a MultiTimbral unit, then each one of its Parts may respond differently (or not at all) to Channel Pressure. The Part affected by a particular Channel Pressure message is the one assigned to the message's MIDI channel.

It is recommended that Channel Pressure default to controlling the VCA level (ie, a volume swell/fade effect).

Status

0xD0 to 0xDF where the low nibble is the MIDI channel.

Data

One data byte follows the Status. It is the pressure amount, a value from 0 to 127 (where 127 is the most pressure).

Errata

What's the difference between **AfterTouch** and **Channel Pressure**? Well, AfterTouch messages are for individual keys (ie, an Aftertouch message only affects that one note whose number is in the message). Every key that you press down generates its own AfterTouch messages. If you press on one key harder than another, then the one key will generate AfterTouch messages with higher values than the other key. The net result is that some effect will be applied to the one key more than the other key. You have individual control over each key that you play. With Channel Pressure, one message is sent out for the entire keyboard. So, if you press one key harder than another, the module will average out the difference, and then just pretend that you're pressing both keys with the exact same pressure. The net result is that some effect gets applied to all sounding keys evenly. You don't have individual control per each key. A controller normally uses either Channel Pressure or AfterTouch, but usually not both. Most MIDI controllers don't generate AfterTouch because that requires a pressure sensor for each individual key on a MIDI keyboard, and this is an expensive feature to implement. For this reason, many cheaper units implement Channel Pressure instead of Aftertouch, as the former only requires one sensor for the entire keyboard's pressure. Of course, a device could implement both Aftertouch and Channel Pressure, in which case the Aftertouch messages for each individual key being held are generated, and then the average pressure is calculated and sent as Channel Pressure.

Pitch Wheel

Category: Voice

Purpose

To set the Pitch Wheel value. The pitch wheel is used to slide a note's pitch up or down in cents (ie, fractions of a half-step). If the device is a MultiTimbral unit, then each one of its Parts may respond differently (or not at all) to Pitch Wheel. The Part affected by a particular Pitch Wheel message is the one assigned to the message's MIDI channel.

Status

0xE0 to 0xEF where the low nibble is the MIDI channel.

Data

Two data bytes follow the status. The two bytes should be combined together to form a 14-bit value. The first data byte's bits 0 to 6 are bits 0 to 6 of the 14-bit value. The second data byte's bits 0 to 6 are really bits 7 to 13 of the 14-bit value. In other words, assuming that a C program has the first byte in the variable **First** and the second data byte in the variable **Second**, here's how to combine them into a 14-bit value (actually 16-bit since most computer CPUs deal with 16-bit, not 14-bit, integers):

```
unsigned short CombineBytes(unsigned char First, unsigned char Second)
{
   unsigned short _14bit;
   _14bit = (unsigned short)Second;
   _14bit <<= 7;
   _14bit |= (unsigned short)First;
   return(_14bit);
}
```

A combined value of 0x2000 is meant to indicate that the Pitch Wheel is centered (ie, the sounding notes aren't being transposed up or down). Higher values transpose pitch up, and lower values transpose pitch down.

Errata

The Pitch Wheel range is usually adjustable by the musician on each MIDI device. For example, although 0x2000 is always center position, on one MIDI device, a 0x3000 could transpose the pitch up a whole step, whereas on another device that may result in only a half step up. The GM spec recommends that MIDI devices default to using the entire range of possible Pitch Wheel message values (ie, 0x0000 to 0x3FFF) as +/- 2 half steps transposition (ie, 4 half-steps total range). The Pitch Wheel Range (or Sensitivity) is adjusted via an RPN controller message.

System Exclusive (ie, SysEx)

Category: System Common

Purpose

Used to send a large amount of data to a MIDI device, such as a dump of its patch memory or sequencer data or waveform data. Also, SysEx may be used to transmit information that is particular to one model device. For example, a SysEx message might be used to set the feedback level for an operator in a Roland Physical Modeling Synth. This information would likely be useless to an AKAI sample playing device. (By contrast, virtually all devices respond to Modulation Wheel control, for example, so it makes sense to have a defined Modulation Controller message that all manufacturers can support for that purpose).

Status

Begins with 0xF0. Ends with a 0xF7 status (ie, after the data bytes).

Data

There can be any number of data bytes inbetween the initial 0xF0 and the final 0xF7. The most important is the first data byte (after the 0xF0), which should be a Manufacturer's ID.

Errata

Virtually every MIDI device defines the format of its own set of SysEx messages (ie, that only it understands). The only common ground between the SysEx messages of various models of MIDI devices is that all SysEx messages must begin with a 0xF0 status and end with a 0xF7 status. In other words, this is the only MIDI message that has 2 Status bytes, one at the start and the other at the end. Inbetween these two status bytes, any number of data bytes (all having bit #7 clear, ie, 0 to 127 value) may be sent. That's why SysEx needs a 0xF7 status byte at the end; so that a MIDI device will know when the end of

the message occurs, even if the data within the message isn't understood by that device (ie, the device doesn't know exactly how many data bytes to expect before the 0xF7).

Usually, the first data byte (after the 0xF0) will be a defined ***Manufacturer's ID***. The MMA has assigned particular values of the ID byte to various manufacturers, so that a device can determine whether a SysEx message is intended for it. For example, a Roland device expects an ID byte of 0x41. If a Roland device receives a SysEx message whose ID byte isn't 0x41, then the device ignores all of the rest of the bytes up to and including the final 0xF7 which indicates that the SysEx message is finished.

The purpose of the remaining data bytes, however many there may be, are determined by the manufacturer of a product. Typically, manufacturers follow the Manufacturer ID with a Model Number ID byte so that a device can not only determine that it's got a SysEx message for the correct manufacturer, but also has a SysEx message specifically for this model. Then, after the Model ID may be another byte that the device uses to determine what the SysEx message is supposed to be for, and therefore, how many more data bytes follow. Some manufacturers have a checksum byte, (usually right before the 0xF7) which is used to check the integrity of the message's transmission.

The 0xF7 Status byte is dedicated to marking the end of a SysEx message. It should never occur without a preceding 0xF0 Status. In the event that a device experiences such a condition (ie, maybe the MIDI cable was connected during the transmission of a SysEx message), the device should ignore the 0xF7.

Furthermore, although the 0xF7 is supposed to mark the end of a SysEx message, in fact, any status (except for Realtime Category messages) will cause a SysEx message to be considered "done" (ie, actually "aborted" is a better description since such a scenario indicates an abnormal MIDI condition). For example, if a 0x90 happened to be sent sometime after a 0xF0 (but before the 0xF7), then the SysEx message would be considered aborted at that point. It should be noted that, like all System Common messages, SysEx cancels any current running status. In other words, the next Voice Category message (after the SysEx message) must begin with a Status.

Here are the assigned Manufacturer ID numbers (ie, the second byte in a System Exclusive message):

Sequential Circuits	1
Big Briar	2
Octave / Plateau	3
Moog	4
Passport Designs	5
Lexicon	6
Kurzweil	7
Fender	8
Gulbransen	9
Delta Labs	0x0A
Sound Comp.	0x0B
General Electro	0x0C
Techmar	0x0D
Matthews Research	0x0E
Oberheim	0x10
PAIA	0x11
Simmons	0x12
DigiDesign	0x13
Fairlight	0x14
Peavey	0x1B
JL Cooper	0x15
Lowery	0x16
Lin	0x17
Emu	0x18
Bon Tempi	0x20
S.I.E.L.	0x21
SyntheAxe	0x23
Hohner	0x24
Crumar	0x25
Solton	0x26
Jellinghaus Ms	0x27
CTS	0x28
PPG	0x29
Elka	0x2F
Cheetah	0x36
Waldorf	0x3E
Kawai	0x40
Roland	0x41
Korg	0x42
Yamaha	0x43
Casio	0x44
Akai	0x45

You'll note that we use only one byte for the Manufacturer ID. And since a midi data byte can't be greater than 0x7F, that means we have only 127 IDs to dole out to manufacturers. Well, there are more than 127 manufacturers of MIDI products.

To accomodate a greater range of manufacturer IDs, the MMA decided to reserve a manufacturer ID of 0 for a special purpose. When you see a manufacturer ID of 0, then there will be two more data bytes after this. These two data bytes combine to make the real manufacturer ID. So, some manufacturers have IDs that are 3 bytes, where the first byte is always 0. Using this "trick", the range of unique manufacturer IDs is extended to accomodate over 16,000 MIDI manufacturers.

For example, Microsoft's manufacturer ID consists of the 3 bytes 0x00 0x00 0x41. Note that the first byte is 0 to indicate that the real ID is 0x0041, but is still different than Roland's ID which is only the single byte of 0x41.

A manufacturer must get a registered ID from the MMA if he wants to define his own SysEx messages, or use the following:

Educational Use 0x7D

This ID is for educational or development use only, and should never appear in a commercial design.

On the other hand, it is permissible to use another manufacturer's defined SysEx message(s) in your own products. For example, if the Roland S-770 has a particular SysEx message that you could use verbatim in your own design, you're free to use that message (and therefore the Roland ID in it). But, you're not allowed to transmit a mutated version of any Roland message with a Roland ID. Only Roland can develop new messages that contain a Roland ID.

Universal SysEx messages are SysEx messages that are not for any one particular manufacturer, but rather, meant to be utilized by all manufacturer's products. For example, many manufacturers make digital samplers. It became desirable for manufacturers to allow exchange of waveform data between each others' products. So, a standard protocol was developed called **MIDI Sample Dump Standard** (SDS). Of course, since waveforms typically entail large amounts of data, SysEx messages (ie, containing over a hundred bytes each) were chosen as the most suitable vehicle to transmit the data over MIDI. But, it was decided not to use a particular manufacturer's ID for these SysEx messages. So, a universal ID was created. There are actually 2 IDs dedicated to Universal SysEx messages. There's a universal ID meant for realtime messages (ie, ones that need to be responded to immediately), and one for non-realtime (ie, ones which can be processed when the device gets around to it). Here are those ID numbers:

```
RealTime ID         0x7F
Non-RealTime ID  0x7E
```

No particular manufacturer is ever assigned an ID consisting of the single byte 0x7F or 0x7E. These are reserved for Universal SysEx messages adopted by the MMA.

A general template for these two IDs was defined. After the ID byte is a ***SysEx Channel*** byte. This could be from 0 to 127 for a total of 128 SysEx channels. So, although "normal" SysEx messages have no MIDI channel like Voice Category messages do, a Universal SysEx message can be sent on one of 128 SysEx channels. This allows the musician to set various devices to ignore certain Universal SysEx messages (ie, if the device allows the musician to set its Base SysEx Channel. Most devices just set their Base Sysex channel to the same number as the Base Channel for Voice Category messages). On the other hand, a SysEx channel of 127 is actually meant to tell the device to "disregard the channel and pay attention to this message regardless".

After the SysEx channel, the next two bytes are **Sub IDs** which tell what the SysEx is for. There are several Sub IDs defined for particular messages. There is a Sub ID for a Universal SysEx message to set a device's master volume. (This is different than Volume controller which sets the volume for only one

particular MIDI channel). There is a Sub ID for a Universal SysEx message to set a device's Pitch Wheel bend range. There are a couple of Sub IDs for some Universal SysEx messages to implement a waveform (sample) dump over MIDI. Etc.

The next sections of this online book document the defined Universal SysEx messages.

GM System Enable/Disable

This Universal SysEx message enables or disables the General MIDI mode of a sound module. Some devices have built-in GM modules or GM Patch Sets in addition to non-GM Patch Sets or non-GM modes of operation. When GM is enabled, it replaces any non-GM Patch Set or non-GM mode with a GM mode/patch set. This allows a device to have modes or Patch Sets that go beyond the limits of GM, and yet, still have the capability to be switched into a GM-compliant mode when desirable.

```
0xF0  SysEx
0x7E  Non-Realtime
0x7F  The SysEx channel. Could be from 0x00 to 0x7F.
      Here we set it to "disregard channel".
0x09  Sub-ID -- GM System Enable/Disable
0xNN  Sub-ID2 -- NN=00 for disable, NN=01 for enable
0xF7  End of SysEx
```

It is best to respond as quickly as possible to this message, and to be ready to accept incoming note (and other) messages soon after, as this message may be included at the start of a General MIDI file to ensure playback by a GM module. Most modules are fully setup in GM mode by 100 milliseconds after receiving the GM System Enable message.

When GM Mode is first enabled, a device should assume that the "Grand Piano" patch (ie, the first GM patch) is the currently selected patch upon all 16 MIDI channels. The device should also internally reset all controllers and assume the power-up state described in the General MIDI Specification.

While GM mode is enabled, a device should also ignore Bank Select messages (since GM does not have more than one bank of patches). Only when the GM Disable message is received (with Sub-ID2 = 0 to disable GM mode) will a device then respond to Bank Select messages (and knock itself out of GM mode).

Master Volume

This Universal SysEx message adjusts a device's master volume. Remember that in a multitimbral device, the Volume controller messages are used to control the volumes of the individual Parts. So, we need some message to control Master Volume. Here it is.

```
0xF0  SysEx
0x7F  Realtime
0x7F  The SysEx channel. Could be from 0x00 to 0x7F.
      Here we set it to "disregard channel".
0x04  Sub-ID -- Device Control
0x01  Sub-ID2 -- Master Volume
0xLL  Bits 0 to 6 of a 14-bit volume
0xMM  Bits 7 to 13 of a 14-bit volume
0xF7  End of SysEx
```

Identity Request

Sometimes, a device may wish to know what other devices are connected to it. For example, a Patch Editor software running on a computer may wish to know what devices are connected to the computer's MIDI port, so that the software can configure itself to accept dumps from those devices.

The **Identity Request** Universal Sysex message can be sent by the Patch Editor software. When this message is received by some device connected to the computer, that device will respond by sending an **Identity Reply** Universal Sysex message back to the computer. The Patch Editor can then examine the information in the **Identity Reply** message to determine what make and model device is connected to the computer. Each device that understands the Identity Request will reply with its own Identity Reply message.

Here is the Identity Request message:

```
0xF0  SysEx
0x7E  Non-Realtime
0x7F  The SysEx channel. Could be from 0x00 to 0x7F.
      Here we set it to "disregard channel".
0x06  Sub-ID -- General Information
0x01  Sub-ID2 -- Identity Request
0xF7  End of SysEx
```

Here is the Identity Reply message:

```
0xF0  SysEx
0x7E  Non-Realtime
0x7F  The SysEx channel. Could be from 0x00 to 0x7F.
      Here we set it to "disregard channel".
0x06  Sub-ID -- General Information
0x02  Sub-ID2 -- Identity Reply
0xID  Manufacturer's ID
0xf1  The f1 and f2 bytes make up the family code. Each
0xf2  manufacturer assigns different family codes to his products.
0xp1  The p1 and p2 bytes make up the model number. Each
0xp2  manufacturer assigns different model numbers to his products.
0xv1  The v1, v2, v3 and v4 bytes make up the version number.
0xv2
0xv3
```

0xv4
0xF7 End of SysEx

MIDI Sample Dump Standard (SDS) is covered in the document MIDI Sample Dump Standard.

MTC Quarter Frame Message

Category: System Common

Purpose

Some master device that controls sequence playback sends this timing message to keep a slave device in sync with the master.

Status

0xF1

Data

One data byte follows the Status. It's the time code value, a number from 0 to 127.

Errata

This is one of the MIDI Time Code (MTC) series of messages. See MIDI Time Code.

Song Position Pointer

Category: System Common

Purpose

Some master device that controls sequence playback sends this message to force a slave device to cue the playback to a certain point in the song/sequence. In other words, this message sets the device's "Song Position". This message doesn't actually start the playback. It just sets up the device to be "ready to play" at a particular point in the song.

Status

0xF2

Data

Two data bytes follow the status. Just like with the Pitch Wheel, these two bytes are combined into a 14-bit value. (See Pitch Wheel). This 14-bit value is the **MIDI Beat** upon which to start the song. Songs are always assumed to start on a MIDI Beat of 0. Each MIDI Beat spans 6 **MIDI Clocks**. In other words, each MIDI Beat is a 16th note (since there are 24 MIDI Clocks in a quarter note).

Errata

Example: If a Song Position value of 8 is received, then a sequencer (or drum box) should cue playback to the third quarter note of the song. (8 MIDI beats * 6 MIDI clocks per MIDI beat = 48 MIDI Clocks. Since there are 24 MIDI Clocks in a quarter note, the first quarter occurs on a time of 0 MIDI Clocks, the second quarter note occurs upon the 24th MIDI Clock, and the third quarter note occurs on the 48th MIDI Clock).

Often, the slave device has its playback tempo synced to the master via MIDI Clock. See Syncing Sequence Playback.

Song Select

Category: System Common

Purpose

Some master device that controls sequence playback sends this message to force a slave device to set a certain song for playback (ie, sequencing).

Status

0xF3

Data

One data byte follows the status. It's the song number, a value from 0 to 127.

Errata

Most devices display "song numbers" starting from 1 instead of 0. Some devices even use different labeling systems for songs, ie, bank 1, number 1 song. But, a Song Select message with song number 0 should always select the first song.

When a device receives a Song Select message, it should cue the new song at MIDI Beat 0 (ie, the very beginning of the song), unless a subsequent Song Position Pointer message is received for a different MIDI Beat. In other words, the device resets its "Song Position" to 0.

Often, the slave device has its playback tempo synced to the master via MIDI Clock. See Syncing Sequence Playback.

Tune Request

Category: System Common

Purpose

The device receiving this should perform a tuning calibration.

Status

0xF6

Data

None

Errata

Mostly used for sound modules with analog oscillator circuits.

MIDI Clock

Category: System Realtime

Purpose

Some master device that controls sequence playback sends this timing message to keep a slave device in sync with the master. A MIDI Clock message is sent at regular intervals (based upon the master's Tempo) in order to accomplish this.

Status

0xF8

Data

None

Errata

There are 24 MIDI Clocks in every quarter note. (12 MIDI Clocks in an eighth note, 6 MIDI Clocks in a 16th, etc). Therefore, when a slave device counts down the receipt of 24 MIDI Clock messages, it knows that one quarter note has passed. When the slave counts off another 24 MIDI Clock messages, it knows that another quarter note has passed. Etc. Of course, the rate that the master sends these messages is based upon the master's tempo. For example, for a tempo of 120 BPM (ie, there are 120 quarter notes in every minute), the master sends a MIDI clock every 20833 microseconds. (ie, There are 1,000,000 microseconds in a second. Therefore, there are 60,000,000 microseconds in a minute. At a tempo of 120 BPM, there are 120 quarter notes per minute. There are 24 MIDI clocks in each quarter note. Therefore, there should be 24 * 120 MIDI Clocks per minute. So, each MIDI Clock is sent at a rate of 60,000,000/(24 * 120) microseconds).

A slave device might receive (from a master device) a Song Select message to cue a specific song to play (out of several songs), a Song Position Pointer

message to cue that song to start on a particular beat, a MIDI Continue in order to start playback from that beat, periodic MIDI Clocks in order to keep the playback in sync with the master, and eventually a MIDI Stop to halt playback. See Syncing Sequence Playback.

Tick

Category: System Realtime

Purpose

Some master device that controls sequence playback sends this timing message to keep a slave device in sync with the master. A MIDI Tick message is sent at regular intervals of one message every 10 milliseconds.

Status

0xF9

Data

None

Errata

While a master device's "clock" is playing back, it will send a continuous stream of MIDI Tick events at a rate of one per every 10 milliseconds.

A slave device might receive (from a master device) a Song Select message to cue a specific song to play (out of several songs), a Song Position Pointer message to cue that song to start on a particular beat, a MIDI Continue in order to start playback from that beat, periodic MIDI Ticks in order to keep the playback in sync with the master, and eventually a MIDI Stop to halt playback. See Syncing Sequence Playback.

MIDI Start

Category: System Realtime

Purpose

Some master device that controls sequence playback sends this message to make a slave device start playback of some song/sequence from the beginning (ie, MIDI Beat 0).

Status

0xFA

Data

None

Errata

A MIDI Start always begins playback at MIDI Beat 0 (ie, the very beginning of the song). So, when a slave device receives a MIDI Start, it automatically resets its "Song Position" to 0. If the device needs to start playback at some other point (either set by a previous Song Position Pointer message, or manually by the musician), then MIDI Continue is used instead of MIDI Start.

Often, the slave device has its playback tempo synced to the master via MIDI Clock. See Syncing Sequence Playback.

MIDI Stop

Category: System Realtime

Purpose

Some master device that controls sequence playback sends this message to make a slave device stop playback of a song/sequence.

Status

0xFC

Data

None

Errata

When a device receives a MIDI Stop, it should keep track of the point at which it stopped playback (ie, its stopped "Song Position"), in the anticipation that a MIDI Continue might be received next.

Often, the slave device has its playback tempo synced to the master via MIDI Clock. See Syncing Sequence Playback.

MIDI Continue

Category: System Realtime

Purpose

Some master device that controls sequence playback sends this message to make a slave device resume playback from its current "Song Position". The current Song Position is the point when the song/sequence was previously stopped, or previously cued with a Song Position Pointer message.

Status

0xFB

Data

None

Errata

Often, the slave device has its playback tempo synced to the master via MIDI Clock. See Syncing Sequence Playback.

Active Sense

Category: System Realtime

Purpose

A device sends out an Active Sense message (at least once) every 300 milliseconds if there has been no other activity on the MIDI buss, to let other devices know that there is still a good MIDI connection between the devices.

Status

0xFE

Data

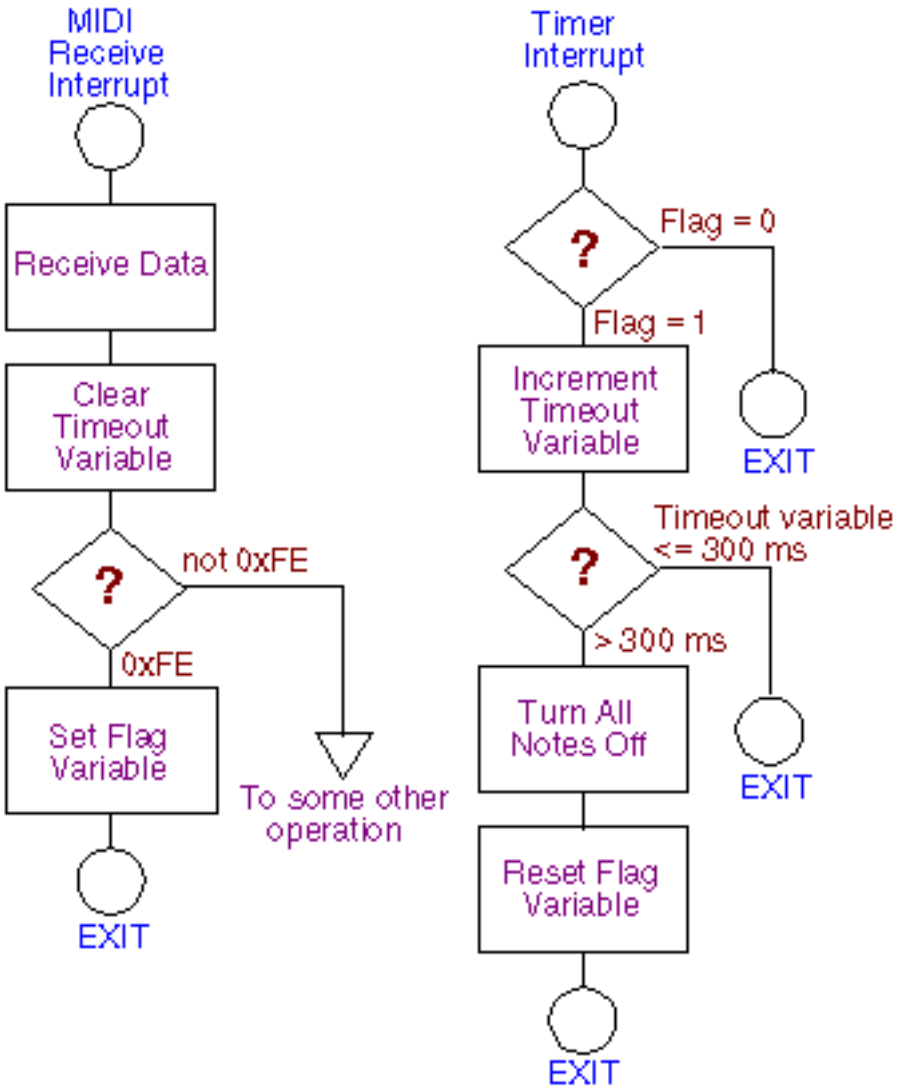
None

Errata

When a device receives an Active Sense message (from some other device), it should expect to receive additional Active Sense messages at a rate of one approximately every 300 milliseconds, whenever there is no activity on the MIDI buss during that time. (Of course, if there are other MIDI messages happening at least once every 300 mSec, then Active Sense won't ever be sent. An Active Sense only gets sent if there is a 300 mSec "moment of silence" on the MIDI buss. You could say that a device that sends out Active Sense "gets nervous" if it has nothing to do for over 300 mSec, and so sends an Active Sense just for the sake of reassuring other devices that this device still exists). If a message is missed (ie, 0xFE nor any other MIDI message is received for over 300 mSec), then a device assumes that the MIDI connection is broken, and turns off all of its playing notes (which were turned on by incoming Note On messages, versus ones played on the local keyboard by a musician). Of course, if a device never receives an Active Sense message to begin with, it should not expect them at all. So, it takes one "nervous" device to start the process by initially sending out an Active Sense message to the other connected devices during a 300 mSec moment of silence on the MIDI bus.

This is an optional feature that only a few devices implement (ie, notably Roland gear). Many devices don't ever initiate this minimal "safety" feature.

Here's a flowchart for implementing Active Sense. It assumes that the device has a hardware timer that ticks once every millisecond. A variable named **Timeout** is used to count the passing milliseconds. Another variable named **Flag** is set when the device receives an Active Sense message from another device, and therefore expects to receive further Active Sense messages.



The logic for active sense detection

Reset

Category: System Realtime

Purpose

The device receiving this should reset itself to a default state, usually the same state as when the device was turned on. Often, this means to turn off all playing notes, turn the local keyboard on, clear running status, set Song Position to 0, stop any timed playback (of a sequence), and perform any other standard setup unique to the device. Also, a device may choose to kick itself into Omni On, Poly mode if it has no facilities for allowing the musician to store a default mode.

Status

0xFF

Data

None

Errata

A Reset message should never be sent automatically by any MIDI device. Rather, this should only be sent when a musician specifically tells a device to do so.

MIDI Modes

Some MIDI devices can be switched in and out of **Omni** state.

When Omni is off, a MIDI device can only respond to Voice Category messages (ie, Status bytes of 0x80 to 0xEF) upon a limited number of channels, usually only 1. Typically, the device allows the musician to pick one of the 16 MIDI channels that the device will respond to. This is then referred to as the device's **Base Channel**. So for example, if a device's Base Channel is set to 1, and a Voice Category message upon channel 2 arrives at the device's MIDI IN, the device ignores that message.

Note: Virtually all modern devices allow the musician to manually choose the Base Channel. A device may even define its own SysEx message that can change its Base Channel. Remember that SysEx messages are of the System Common Category, and therefore aren't (normally) tied to the Base Channel itself.

When Omni is on, a device doesn't respond to just one MIDI channel, but rather, responds to all 16 MIDI channels. The only benefit of Omni On is that, regardless of which channel any message is received upon, a device always responds to the message. This mades it very foolproof for a musician to hook up two devices and always have one device respond to the other regardless of any MIDI channel discrepancies between the device generating the data (ie, referred to as the **transmitter**) and the device receiving the data (ie, referred to as the **receiver**). Of course, if the musician daisy-chains another device, and he wants the 2 devices to play different musical parts, then he has to switch Omni Off on both devices. Otherwise, a device with Omni On will respond to messages intended for the other device (as well as messages intended for itself).

Note: Omni can be switched on or off with the Omni On and Omni Off controller messages. But these messages must be received upon the device's Base Channel in order for the device to respond to them. What this implies is that even when a device is in Omni On state (ie, capable of responding to all 16 channels), it still has a Base Channel for the purpose of turning Omni On or

Off.

One might think that MultiTimbral devices employ Omni On. Because you typically may choose (upto) 16 different Patches, each playing its own musical part, you need the device to be able to respond to more than one MIDI channel so that you can assign each Patch to a different MIDI channel. Actually, MultiTimbral devices do not use Omni On for this purpose. Rather, the device regards itself as having 16 separate sound modules (ie, Parts) inside of it, with each module in Omni Off mode, and capable of being set to its own Base Channel. Usually, you also have a "master" Base Channel which may end up having to be set the same as one of the individual Parts. Most MultiTimbral devices offer the musician the choice of which particular channels to use, and which to ignore (if he doesn't need all 16 patches playing simultaneously on different channels). In this way, he can daisy-chain another multitimbral device and use any ignored channels (on the first device) with this second device. Unfortunately, the MIDI spec has no specific "MultiTimbral" mode message. So, a little "creative reinterpretation" of Monophonic mode is employed, as you'll learn in a moment.

In addition to Omni On or Off, many devices can be switched between Polyphonic or Monophonic operation.

In Polyphonic operation, a device can respond to more than one Note On upon a given channel. In other words, it can play chords on that channel. For example, assume that a device is responding to Voice Category messages on channel 1. If the device receives a Note On for middle C on channel 1, it will sound that note. If the device then receives a Note On for high C also on channel 1 (before receiving a Note Off for middle C), the device will sound the high C as well. Both notes will then be sounding simultaneously.

In Monophonic operation, a device can only respond to one Note On at a time upon a given channel. It can't play chords; only single note "melodies". For example, assume that a device is responding to Voice Category messages on channel 1. If the device receives a Note On for middle C on channel 1, it will play that note. If the device then receives a Note On for high C (before receiving a Note Off for middle C), the device will automatically turn off the

middle C before playing the high C. So what's the use of forcing a device capable of playing chords into such a Monophonic state? Well, there are lots of Monophonic instruments in the world, for example, most brass and woodwinds. They can only play one note at a time. If using a Trumpet Patch, a keyboard player might want to force a device into Monophonic operation in order to better simulate a Trumpet. Some devices have special effects that only work in Monophonic operation such as Portamento, and smooth transition between notes (ie, skipping the VCA attack when moving from one Note On that "overlaps" another Note On -- this is often referred to as **legato** and makes for a more realistic musical performance for brass and woodwind patches). That's in theory how Mono operation is supposed to work, but MultiTimbral devices created long after the MIDI spec was designed, had to subvert Mono operation into Polyphonic operation in order to come up with a "MultiTimbral mode", as you'll learn.

Note: A device can be switched between Polyphonic or Monophonic with the Polyphonic and Monophonic controller messages. But these messages must be received upon the device's Base Channel in order for the device to respond to them.

Of course, a MIDI device could have Omni On and be in Polyphonic state. Or, the device could have Omni On but be in Monophonic state. Or, the device could have Omni Off and be in Polyphonic state. Or, the device could have Omni Off but be in Monophonic state. There are 4 possible combinations here, and MIDI refers to these as 4 **Modes**. For example, Mode 1 is the aforementioned Omni On / Polyphonic state. Here are the 4 Modes:

Mode 1 - Omni On / Poly

The device plays all MIDI data received on all 16 MIDI channels. If a MultiTimbral device, then it often requires the musician to manually select which one Patch to play all 16 channels, and this setting is usually saved in "patch memory".

Mode 2 - Omni On / Mono

The device plays only one note out of all of the MIDI data received on all 16 MIDI channels. This mode is seldom implemented because playing one note out of all the data happening on all 16 channels is not very useful.

Mode 3 - Omni Off / Poly

The device plays all MIDI data received on 1 specific MIDI channel. The musician usually gets to choose which channel he wants that to be. If a MultiTimbral device, then it often requires the musician to manually select which one Patch to play that MIDI channel, and this setting is usually saved in "patch memory".

Mode 4 - Omni Off / Mono

In theory, the device plays one note at a time on 1 (or more) specific MIDI channels. In practice, the manufacturers of MultiTimbral threw the entire concept of Monophonic out the window, and use this for "MultiTimbral mode". On a MultiTimbral device, this mode means that the device plays polyphonically on 1 (or more) specific MIDI channels. The Monophonic controller message has a Value associated with it. This Value is applicable in Mode 4 (whereas it's ignored in Mode 2), and determines how many MIDI channels are responded to. If 1, then on a non-MultiTimbral device, this would give you a truly monophonic instrument. Of course, on a MultiTimbral device, it gives you the same thing as Mode 3. If the Value is 0, then a non-MultiTimbral device uses as many MIDI channels as it has voices. So, for an 8 voice synth, it would use 8 MIDI Channels, and each of those channels would play one note at a time. For a MultiTimbral device, if the Value is 0, then the device uses as many MIDI channels as it has Parts. So, if a MultiTimbral device can play only 8 patches simultaneously, then it would use 8 MIDI Channels, and each of those channels could play polyphonically.

Some devices do not support all of these modes. The device should ignore controller messages which attempt to switch it into an unsupported state, or switch to the next closest mode.

If a device doesn't have some way of saving the musician's choice of Mode when the unit is turned off, the device should default to Mode 1 upon the next

power up.

On final question arises. If a MultiTimbral device doesn't implement a true monophonic mode for Mode 4, then how do you get one of its Parts to play in that useful Monophonic state (ie, where you have Portamento and legato features)? Well, many MultiTimbral devices allow a musician to manually enable a "Solo Mode" per each Part. Some devices even use the Legato Pedal controller (or a General Purpose Button controller) to enable/disable that function, so that you can turn it on/off for each Part over MIDI.

Note: A device that can both generate MIDI messages (ie, perhaps from an electronic piano keyboard) as well as receive MIDI messages (ie, to be played on its internal sound circuitry), is allowed to have its transmitter set to a different Mode and MIDI channel than its receiver, if this is desired. In fact, on MultiTimbral devices with a keyboard, the keyboard often has to switch between MIDI channels so that the musician can access the Parts one at a time, without upsetting the MIDI channel assignments for those Parts.

RealTime Category Messages

Each RealTime Category message (ie, Status of 0xF8 to 0xFF) consists of only 1 byte, the Status. These messages are primarily concerned with timing/syncing functions which means that they must be sent and received at specific times without any delays. Because of this, MIDI allows a RealTime message to be sent at any time, even interspersed within some other MIDI message. For example, a RealTime message could be sent inbetween the two data bytes of a Note On message. A device should always be prepared to handle such a situation; processing the 1 byte RealTime message, and then subsequently resume processing the previously interrupted message as if the RealTime message had never occurred.

For more information about RealTime, read the sections Running Status, Ignoring MIDI Messages, and Syncing Sequence Playback.

Running Status

The MIDI spec allows for a MIDI message to be sent without its Status byte (ie, just its data bytes are sent) as long as the previous, transmitted message had the same Status. This is referred to as **running status**. Running status is simply a clever scheme to maximize the efficiency of MIDI transmission (by removing extraneous Status bytes). The basic philosophy of running status is that a device must always remember the last Status byte that it received (except for RealTime), and if it doesn't receive a Status byte when expected (on subsequent messages), it should assume that it's dealing with a running status situation. A device that generates MIDI messages should always remember the last Status byte that it sent (except for RealTime), and if it needs to send another message with the same Status, the Status byte may be omitted.

Let's take an example of a device creating a stream of MIDI messages. Assume that the device needs to send 3 Note On messages (for middle C, E above middle C, and G above middle C) on channel 0. Here are the 3 MIDI messages to which I'm referring.

```
0x90 0x3C 0x7F
0x90 0x40 0x7F
0x90 0x43 0x7F
```

Notice that the Status bytes of all 3 messages are the same (ie, Note On, Channel 0). Therefore the device could implement running status for the latter 2 messages, sending the following bytes:

```
0x90 0x3C 0x7F
0x40 0x7F
0x43 0x7F
```

This allows the device to save time since there are 2 less bytes to transmit. Indeed, if the message that the device sent before these 3 also happened to be a Note On message on channel 0, then the device could have omitted the first message's Status too.

Now let's take the perspective of a device receiving this above stream. It receives the first message's Status (ie, 0x90) and thinks "Here's a Note On

Status on channel 0. I'll remember this Status byte. I know that there are 2 more data bytes in a Note On message. I'll expect those next". And, it receives those 2 data bytes. Then, it receives the data byte of the second message (ie, 0x40). Here's when the device thinks "I didn't expect another data byte. I expected the Status byte of some message. This must be a running status message. The last Status byte that I received was 0x90, so I'll assume that this is the same Status. Therefore, this 0x40 is the first data byte of another Note On message on channel 0".

Remember that a Note On message with a velocity of 0 is really considered to be a Note Off. With this in mind, you could send a whole stream of note messages (ie, turning notes on and off) without needing a Status byte for all but the first message. All of the messages will be Note On status, but the messages that really turn notes off will have 0 velocity. For example, here's how to play and release middle C utilizing running status:

```
0x90 0x3C 0x7F
0x3C 0x00     <-- This is really a Note Off because of 0 velocity
```

RealTime Category messages (ie, Status of 0xF8 to 0xFF) do not effect running status in any way. Because a RealTime message consists of only 1 byte, and it may be received at any time, including interspersed with another message, it should be handled transparently. For example, if a 0xF8 byte was received inbetween any 2 bytes of the above examples, the 0xF8 should be processed immediately, and then the device should resume processing the example streams exactly as it would have otherwise. Because RealTime messages only consist of a Status, running status obviously can't be implemented on RealTime messages.

System Common Category messages (ie, Status of 0xF0 to 0xF7) cancel any running status. In other words, the message after a System Common message must begin with a Status byte. System Common messages themselves can't be implemented with running status. For example, if a Song Select message was sent immediately after another Song Select, the second message would still need a Status byte.

Running status is only implemented for Voice Category messages (ie, Status is

0x80 to 0xEF).

A recommended approach for a receiving device is to maintain its "running status buffer" as so:

1. Buffer is cleared (ie, set to 0) at power up.
2. Buffer stores the status when a Voice Category Status (ie, 0x80 to 0xEF) is received.
3. Buffer is cleared when a System Common Category Status (ie, 0xF0 to 0xF7) is received.
4. Nothing is done to the buffer when a RealTime Category message is received.
5. Any data bytes are ignored when the buffer is 0.

Syncing Sequence Playback

A sequencer is a software program or hardware unit that "plays" a musical performance complete with appropriate rhythmic and melodic inflections (ie, plays musical notes in the context of a musical beat).

Often, it's necessary to synchronize a sequencer to some other device that is controlling a timed playback, such as a drum box playing its internal rhythm patterns, so that both play at the same instant and the same tempo. Several MIDI messages are used to cue devices to start playback at a certain point in the sequence, make sure that the devices start simultaneously, and then keep the devices in sync until they are simultaneously stopped. One device, the master, sends these messages to the other device, the slave. The slave references its playback to these messages.

The message that controls the playback rate (ie, ultimately tempo) is MIDI Clock. This is sent by the master at a rate dependent upon the master's tempo. Specifically, the master sends 24 MIDI Clocks, spaced at equal intervals, during every quarter note interval.(12 MIDI Clocks are in an eighth note, 6 MIDI Clocks in a 16th, etc). Therefore, when a slave device counts down the receipt of 24 MIDI Clock messages, it knows that one quarter note has passed. When the slave counts off another 24 MIDI Clock messages, it knows that another quarter note has passed.

For example, if a master is set at a tempo of 120 BPM (ie, there are 120 quarter notes in every minute), the master sends a MIDI clock every 20833 microseconds. (ie, There are 1,000,000 microseconds in a second. Therefore, there are 60,000,000 microseconds in a minute. At a tempo of 120 BPM, there are 120 quarter notes per minute. There are 24 MIDI clocks in each quarter note. Therefore, there should be 24 * 120 MIDI Clocks per minute. So, each MIDI Clock is sent at a rate of 60,000,000/(24 * 120) microseconds).

Alternately, if a sequencer wishes to control playback independent of tempo, it can use Tick messages. These are sent at a rate of 1 message every 10 milliseconds. Of course, it is then up to the slave device to maintain and update its clock based upon these messages. The slave will be doing its own counting

off of how many milliseconds are supposed to be in each "beat" at the current tempo.

The master needs to be able to start the slave precisely when the master starts. The master does this by sending a MIDI Start message. The MIDI Start message alerts the slave that, upon receipt of the very next MIDI Clock message, the slave should start the playback of its sequence. In other words, the MIDI Start puts the slave in "play mode", and the receipt of that first MIDI Clock marks the initial downbeat of the song (ie, **MIDI Beat** 0). What this means is that (typically) the master sends out that MIDI Clock "downbeat" immediately after the MIDI Start. (In practice, most masters allow a 1 millisecond interval inbetween the MIDI Start and subsequent MIDI Clock messages in order to give the slave an opportunity to prepare itself for playback). In essense, a MIDI Start is just a warning to let the slave know that the next MIDI Clock represents the downbeat, and playback is to start then. Of course, the slave then begins counting off subsequent MIDI Clock messages, with every 6th being a passing 16th note, every 12th being a passing eighth note, and every 24th being a passing quarter note.

A master stops the slave simultaneously by sending a MIDI Stop message. The master may then continue to send MIDI Clocks at the rate of its tempo, but the slave should ignore these, and not advance its "song position". Of course, the slave may use these continuing MIDI Clocks to ascertain what the master's tempo is at all times.

Sometimes, a musician will want to start the playback point somewhere other than at the beginning of a song (ie, he may be recording an overdub in a certain part of the song). The master needs to tell the slave what beat to cue playback to. The master does this by sending a Song Position Pointer message. The 2 data bytes in a Song Position Pointer are a 14-bit value that determines the **MIDI Beat** upon which to start playback. Sequences are always assumed to start on a MIDI Beat of 0 (ie, the downbeat). Each MIDI Beat spans 6 **MIDI Clocks**. In other words, each MIDI Beat is a 16th note (since there are 24 MIDI Clocks in a quarter note, therefore 4 MIDI Beats also fit in a quarter). So, a master can sync playback to a resolution of any particular 16th note.

For example, if a Song Position value of 8 is received, then a slave should cue playback to the third quarter note of the song. (8 MIDI beats * 6 MIDI clocks per MIDI beat = 48 MIDI Clocks. Since there are 24 MIDI Clocks in a quarter note, the first quarter occurs on a time of 0 MIDI Clocks, the second quarter note occurs upon the 24th MIDI Clock, and the third quarter note occurs on the 48th MIDI Clock).

A Song Position Pointer message should not be sent while the devices are in play. This message should only be sent while devices are stopped. Otherwise, a slave might take too long to cue its new start point and miss a MIDI Clock that it should be processing.

A MIDI Start always begins playback at MIDI Beat 0 (ie, the very beginning of the song). So, when a slave receives a MIDI Start, it automatically resets its "Song Position" to 0. If the master needs to start playback at some other point (as set by a Song Position Pointer message), then a MIDI Continue message is sent instead of MIDI Start. Like a MIDI Start, the MIDI Continue is immediately followed by a MIDI Clock "downbeat" in order to start playback then. The only difference with MIDI Continue is that this downbeat won't necessarily be the very start of the song. The downbeat will be at whichever point the playback was set via a Song Position Pointer message or at the point when a MIDI Stop message was sent (whichever message last occurred). What this implies is that a slave must always remember its "current song position" in terms of MIDI Beats. The slave should keep track of the nearest previous MIDI beat at which it stopped playback (ie, its stopped "Song Position"), in the anticipation that a MIDI Continue might be received next.

Some playback devices have the capability of containing several sequences. These are usually numbered from 0 to however many sequences there are. If 2 such devices are synced, a musician typically may set up the sequences on each to match the other. For example, if the master is a sequencer with a reggae bass line for sequence 0, then a slaved drum box might have a reggae drum beat for sequence 0. The musician can then select the same sequence number on both devices simultaneously by having the master send a Song Select message whenever the musician selects that sequence on the master. When a slave receives a Song Select message, it should cue the new song at MIDI Beat 0 (ie,

reset its "song position" to 0). The master should also assume that the newly selected song will start from beat 0. Of course, the master could send a subsequent Song Position Pointer message (after the Song Select) to cue the slave to a different MIDI Beat.

If a slave receives MIDI Start or MIDI Continue messages while it's in play, it should ignore those messages. Likewise, if it receives MIDI Stop messages while stopped, it ignores those.

Ignoring MIDI Messages

A device should be able to "ignore" all MIDI messages that it doesn't use, including currently undefined MIDI messages (ie Status is 0xF4, 0xF5, or 0xFD). In other words, a device is expected to be able to deal with all MIDI messages that it could possibly be sent, even if it simply ignores those messages that aren't applicable to the device's functions.

If a MIDI message is not a RealTime Category message, then the way to ignore the message is to throw away its Status and all data bytes (ie, bit #7 clear) up to the next received, non-RealTime Status byte. If a RealTime Category message is received interspersed with this message's data bytes (remember that all RealTime Category messages consist of only 1 byte, the Status), then the device will have to process that 1 Status byte, and then return to the process of skipping the initial message. Of course, if the next received, non-RealTime Status byte is for another message that the device doesn't use, then the "skip procedure" is repeated.

If the MIDI message is a RealTime Category message, then the way to ignore the message is to simply ignore that one Status byte. All RealTime messages have only 1 byte, the Status. Even the two undefined RealTime Category Status bytes of 0xF9 and 0xFD should be skipped in this manner. Remember that RealTime Category messages do not cancel running status and also could be sent interspersed with some other message, so any data bytes after a RealTime Category message must belong to some other message.

Many devices use playback of digital audio waveforms as their audio source. It was desirable to implement a sub-protocol within MIDI in which devices could exchange this digital audio waveform data. In other words, a protocol was needed that allowed devices to exchange waveform data over MIDI cables within the parameters of MIDI. The only way to do this was with System Exclusive messages, and so several specific SysEx messages were defined in order to implement *Sample Dump Standard* (SDS). Many samplers support this protocol.

The device that sends the waveform data is the transmitter, and the device that receives it is the receiver.

A waveform exchange (ie, *dump*) can be done with or without handshaking. In the non-handshaking version, the transmitter's MIDI OUT is connected to the receiver's MIDI IN, and only the transmitter sends MIDI messages to the receiver. In the handshaking version, the transmitter's MIDI OUT is connected to the receiver's MIDI IN and the receiver's MIDI OUT is connected to the transmitter's MIDI IN. The transmitter sends a portion of the waveform data, after which it expects some sort of acknowledgement from the receiver that the portion has been received successfully or otherwise, and then the transmitter sends the next portion. All of this is accomplished with the devices passing defined SysEx messages between themselves.

The SysEx messages are the DUMP REQUEST, ACK, NAK, WAIT, CANCEL, Dump Header, and Data Packet messages. The first 5 (capitalized) are generated by the receiver. The last 2 are generated by the transmitter.

The dump procedure works as follows. The transmitter sends a Dump Header to indicate a dump start to the receiver. (This could have happened as a result of the receiver requesting the transmitter to start a dump via the DUMP REQUEST). The transmitter then waits for upto 2 seconds for a response from the receiver. This gives the receiver a chance to decide if it wants to and can accept the waveform. If no response is received, then the transmitter assumes a non-handshaking action, and proceeds to send out the first Data Packet. If an expected response (ie, handshake) is received instead, then the transmitter bases its next action upon the receiver's response. If the response is an ACK, the transmitter proceeds to send out the first Data Packet. If the response is a NAK, the transmitter sends the Dump Header again. If the response is a CANCEL, the transmitter aborts the dump. If the response is a WAIT, the transmitter pauses indefinitely until it subsequently receives one of the preceding responses. After the first data packet is sent, the transmitter waits for upto 20 milliseconds for a response from the receiver. This gives the receiver time to perform certain error-checking on the packet's contents. If no response is received, then the transmitter assumes a non-handshaking action, and proceeds to send out the next Data Packet. If an expected response is received instead, then the transmitter bases its next action upon the receiver's response. If the response is an ACK, the transmitter proceeds to send out the next Data Packet. If the response is a NAK, the transmitter resends that same (ie, first) Data Packet again. If the response is a CANCEL, the transmitter aborts the dump. If the response is a WAIT, the transmitter pauses indefinitely until it subsequently receives one of the preceding responses. Eventually, the transmitter sends out as many Data Packets as are needed to pass all of the waveform data to the receiver, repeating this handshake procedure after each packet (or assuming a non-handshaking action after each packet). After that happens, the dump is done.

Here are the messages (with all bytes in hex). In each message, the byte notated as **cc** represents the SysEx channel that the message is being sent upon. There are 128 possible SysEx channels that a device can be set to (ie, 0 to 127).This allows various devices to be set to different SysEx channels along the daisy-chain, and have the dump occur between 2 particular devices with matching SysEx channels.

DUMP REQUEST

F0 7E **cc** 03 **sl sh** F7

If a receiving device wishes to initiate the dump (ie, tell some other device to send some waveform data), then the receiver sends the DUMP REQUEST. The **sl sh** is the 14-bit number (ie 0 to 16,384) of the waveform which the receiver is requesting from the transmitter. Most samplers number their internal waveforms from 0 to how ever many waveforms there are. Note that the 14-bit sample number is transmitted as 2 bytes where the first byte (sl) contains bits 0 to 6 (with high bit clear), and the second byte (sh) contains bits 7 to 13, right-justified (with high bit clear). When the transmitter gets this request, if such a sample number is available, the transmitter will kick off the dump with a Dump Header. Otherwise, the transmitter will cancel the dump. Typically, the receiver will wait for the Dump Header for a few seconds, and if not received, will abort the operation.

Dump Header

F0 7E **cc** 01 **sl sh** *ee* **pl pm ph** *gl gm gh* **hl hm hh** *il im ih* **jj** F7

The transmitter sends this to the receiver to provide information about the waveform data that is about to be sent (in Data Packet messages). The **sl sh** is the waveform's 14-bit number (ie 0 to 16,384). See Dump Request.

ee is the number of significant bits of the waveform. For example, a 16-bit resolution waveform would have a 16 here.

pl pm ph is the sample period in nanoseconds (ie, 1,000,000,000/sample rate in Hertz). For example, a waveform sampled at 41667 Hertz will have a period of 23,999 nanoseconds. This value is transmitted as 3 bytes where pl is bits 0 to 6, pm is bits 7 to 13 **right-justified**, and ph is bits 14 to 20 **right-justified** (ie, for a total of 20 bits of resolution) with the high bit of all 3 bytes clear. So, our 23,999 (0x5DBF) becomes the 3 bytes 3F 3B 01.

gl gm gh is the waveform length **in words**. (What this implies is that if you have 8-bit or less resolution, the waveform length will be half the number of sample points that you intend to dump. You always end up having to send an even number of points).

hl hm hh is the word offset (from 0, ie, the very first sample point in the waveform) where the *sustain loop* starts. **il im ih** is the word offset where the *sustain loop* ends (ie, where the playback loops back to the *sustain loop* start). **jj** is the looptype where 00 means "forward only" (most common) and 01

means "backward/forward", and 7F means "no loop point" (ie, the waveform is played through once only without looping). Note that older MIDI samplers didn't support the 7F value for looptype. For these older samplers, usually, if you set both th start and end loop points to the same value as the waveform length, a sampler will consider this to be a non-looped waveform. So to be safe, when you want to indicate that a waveform is not to be looped, you should set looptype to 7F, and set the start and end loop positions to the same value as the waveform's length.

Data Packet

F0 7E **cc** 02 **kk** *[120 bytes here]* **ll** F7

The data packet is what is used to transfer the actual waveform data. It transfers 120 bytes of waveform data at a time. So, the total size of a packet is 127 bytes.

kk is the packet number from 0 to 127. The first packet that is sent is number 0. The second packet is number 1. After packet number 127, the count rolls over to 0 again (ie, packet 128 becomes 0 again). This number is used by the receiver to ensure that it hasn't missed any packets. The packet number is also used to distinguish new packets from resent packets. After all, if packet number 3 follows packet number 1, then either packet number 2 has been missed by the receiver, or the transmitter sent packets out of order. For example, assume that a device has gotten packet 1, found an error in it, and sends a NAK to the transmitter. But, the transmitter has already assumed non-handshaking and started sending packet 2. The receiver would note that the next arriving packet is number 2. Then, the transmitter finally sees the receiver's late NAK to packet number 1, and resends that packet. The receiver can then note that it has received packet 1 out of order.

The 120 bytes of waveform data follow. The transmitter has to pack up each sample point of its waveform data. With a 16-bit waveform, the transmitter must break up each 16-bit word into 3 bytes for transmission where the first contains bits 15 to 9, the second contains bits 8 to 2, and the third contains bits 1 and 0. In other words, unlike with the waveform length of the DUMP HEADER, the DATA PACKET's bytes are **left-justified**. The first data byte contains the highest 7 bits (which are placed in bit positions 0 to 6, since you'll remember that all transmitted data bytes must have bit 7 clear). The second data byte contains the next highest 7 bits. And the last data byte contains the remaining, lowest bits, which for a 16-bit point means the last two bits. For example, the 16-bit sample word 0xF0F0 would be 0x78 0x3C 0x00. Because each 16-bit word must be broken up into 3 bytes, and because there must be only 120 bytes in a packet, that means that a packet can contain 40, 16-bit sample points. In fact, waveforms with resolutions of 15 to 21 bits pack up likewise. Waveforms with resolutions of 8 to 14 bits pack each sample point into 2 bytes (for 60 points per packet). Waveforms with resolutions of 22 to 28 bits pack each sample point into 4 bytes (for 30 points per packet). Sample points are represented by 0 being full negative value. So, in a 16-bit waveform, 0x0000 is full negative value and 0xFFFF is full positive value (ie, signed shorts aren't used, unlike in the WAVE file format, so you have to subtract a 16-bit point by 0x8000 after unpacking the 3 bytes into an unsigned short, if you want to adjust to a signed short).

Here's a C example of how to unpack 3 bytes of a DATA PACKET into a 16-bit point (ie, # of significant bits = 16). It is passed a pointer to the first of those 3 bytes, and returns a signed 16-bit

point.

```
short unpack3(unsigned char * ptr)
{
    unsigned short num;

    /* Unpack 3 bytes into an unsigned short */
    num = ((unsigned short)(*ptr) << 9) |
       ((unsigned short)(*(ptr+1)) << 2) |
       (*(ptr+2) >> 5);

    /* Change unsigned range to signed range */
    num -= 0x8000;

    return((short)num);
}
```

NOTE: Even the last packet must have 120 data bytes in it. If a particular waveform packs up such that there aren't 120 bytes for the last packet, then that last packet's data should be padded out with 0 bytes to 120 bytes total. The receiver should ACK this last packet also.

ll is the checksum. This is the XOR of the bytes 0x7E, **cc**, 0x02, **kk**, and all 120 bytes of waveform data (with bit 7 of result masked off to 0). The receiver uses this to check that no errors occurred in the packet transmission. If so, the receiver will NAK this packet, and expect the transmitter to resend it.

ACK

F0 7E **cc** 7F **kk** F7

The receiver sends this after successfully receiving a Dump Header and after each successfully received Data Packet. It means "the last message was received correctly. Proceed with the next message". **kk** is the packet number that was received correctly (0 if responding to a Dump Header). The transmitter uses this to determine which particular packet the receiver has accepted (in case packet dumps get out of order).

NAK

F0 7E **cc** 7E **kk** F7

The receiver sends this after unsuccessfully receiving a Dump Header and after each unsuccessfully received Data Packet. It means "the last message was **not** received correctly. Resend that message". **kk** is the packet number that was received incorrectly (0 if responding to a Dump Header). The transmitter uses this to determine which particular packet the receiver has rejected (in case packet dumps get out of order).

CANCEL

F0 7E **cc** 7D **kk** F7

The receiver sends this when it wishes the transmitter to stop the dump. **kk** is the packet number upon which the dump is aborted (0 if responding to a Dump Header).

WAIT

F0 7E **cc** 7C **kk** F7

The receiver sends this when it wants the transmitter to pause the dump operation. The transmitter will send nothing until it receives another message from the receiver; an ACK to continue, a NAK to resend, or a CANCEL to abort the dump. **kk** is the packet number upon which the wait was initiated (0 if responding to a Dump Header).

This is useful for receivers which need to perform lengthy operations at certain times, such as writing data to floppy disk. If the receiver did not issue a WAIT, then the transmitter might count down its 20 millisecond timeout, and assume a non-handshaking action such as sending the next packet, without waiting for a response from the receiver. A WAIT tells the transmitter to wait indefinitely for a response.

Some people like to use computer-based wave editing software to find and set loops points. This is because the computer's large display, and mouse support, is more conducive to displaying a waveform and quickly locating satisfactory loop points, than the typically small LCD upon MIDI samplers (plus a lack of pointing devices such as a mouse). The task of finding satisfactory loop points typically involves much trial-and-error. The user has to set the start and end loop points, listen to the result, and then choose other points if the result is not yet satisfactory. Because the waveform usually has to be sent back to the sampler in order to properly judge the results, and because a MIDI Sample Dump can be a time-consuming procedure, this means that the user typically wastes a lot of time waiting for samples to be transferred. For this reason, 2 messages were added to the SDS specification. (Note that many early MIDI samplers do not support these newer messages). One message allows a transmitter (such as a computer) to ask the receiver to send only the position of the loop points for a given waveform. That means that the transmitter can quickly get information about loop points without needing to transfer an entire waveform dump. The other message allows a transmitter (such as a computer) to tell the receiver to set the start and end loop points to particular positions for a given waveform. That means that the transmitter can quickly set new loop positions without needing to transfer an entire waveform dump.

It is also possible to send/receive multiple loop points (up to 16384) in one message (as described below).

LOOP POINT TRANSMIT

F0 7E **cc** 05 01 **sl sh** *ll lh* **jj** *hl hm hh* **il im ih** *more* F7

The transmitter sends this to the receiver to set the start loop and end loop positions for a particular waveform. The receiver should set those loop positions for that waveform and ACK this message if successful. Otherwise, a NAK is returned. The **sl sh** is the waveform's 14-bit number (ie 0 to 16,384). See Dump Request.

The **ll lh** is the loop's 14-bit number (ie 0 to 16,384). Many samplers allow more than one loop to set for a given waveform, for example, there can be a sustain loop (ie, the part of the waveform looped while the user holds down a key and the sustain portion of a VCA is sustaining the sound), and a release loop (ie, the part of the waveform looped after the user releases the key and the release portion of a VCA is slowly fading out the sound). The sampler numbers the loops from 0 to how ever many loops are supported per waveform. Note that the 14-bit sample number is transmitted as 2 bytes where the first byte (ll) contains bits 0 to 6 (with high bit clear), and the second byte (lh) contains bits 7 to 13, right-justified (with high bit clear). The number of loops supported will likely vary from manufacturer to manufacturer, but a loop number of 00 00 always refers to the sustain loop. A loop number of 7F 7F is reserved to mean "delete all loops" (ie, the sampler will delete all loops that are currently set for the waveform. This is an easy way to start with a "clean slate", but note that not all samplers support this special request).

hl hm hh, **il im ih**, and **jj** are the loop start position, loop end position, and looptype. They are specified in the same way as per the Dump Header message.

It is also possible to specify more loop points (up to 16384) in one message. Where you see *more* in the above template, you could put another loop number, followed by its looptype, loop start position, and loop end position. After this, you could repeat these fields for the next loop, etc. So how does the receiver know how many loop points he is getting? Well, if he doesn't find an F7 where he expects one, then he must be dealing with the *ll* byte of the next loop's number. Therefore he should expect to find a following *lh* **jj** *hl hm hh* **il im ih** bytes. After that should be an F7, but of course, it could be yet another loop's ll byte.

LOOP POINT REQUEST

F0 7E **cc** 05 02 **sl sh** *ll lh* F7

The transmitter sends this to the receiver to ask it to send the start loop and end loop positions for a particular waveform. The receiver will then return a Loop Point Transmit message containing the requested information, or a NAK if it can't handle the request successfully. The **sl sh** is the waveform's 14-bit number (ie 0 to 16,384). See Dump Request.

The **ll lh** is the loop's 14-bit number (ie 0 to 16,384). See Loop Point Transmit.

I don't have enough information to determine what happens when you use a loop number of 7F 7F. This may cause the receiver to return a Loop Point Transmit containing all of the loops for that waveform. Or, I don't know as if you can specify several loop numbers in the above message, in order to have the receiver return all of those loops in one Loop Point Transmit message. You'll have to experiment to deduce this information. If someone does some experiments with a sampler that supports these Loop Point messages, please inform me of the results.

MIDI Time Code (MTC) is a sub-protocol within MIDI, and is used to keep 2 devices that control some sort of timed performance (ie, maybe a sequencer and a video deck) in sync. MTC messages are an alternative to using MIDI Clocks and Song Position Pointer messages. MTC is essentially SMPTE mutated for transmission over MIDI. SMPTE timing is referenced from an absolute "time of day". On the other hand, MIDI Clocks and Song Position Pointer are based upon musical beats from the start of a song, played at a specific Tempo. For many (non-musical) cues, it's easier for humans to reference time in some absolute way rather than based upon musical beats at a certain tempo.

There are several MIDI messages which make up the MTC protocol. All but one are specially defined SysEx messages.

Quarter Frame

The most important message is the *Quarter Frame* message (which is not a SysEx message). It has a status of 0xF1, and one subsequent data byte. This message is sent periodically to keep track of the running SMPTE time. It's analogous to the MIDI Clock message. The Quarter Frame messages are sent at a rate of 4 per each SMPTE Frame. In other words, by the time that a slave has received 4 Quarter Frame messages, a SMPTE Frame has passed. So, the Quarter Frame messages provide a "sub-frame" clock reference. (With 30 fps SMPTE, this "clock tick" happens every 8.3 milliseconds).

But the Quarter Frame is more than just a quarter frame "clock tick". The Quarter Frame message's data byte contains the SMPTE time (ie, hours, minutes, seconds, and frames). SMPTE time is normally expressed in 80 bits. Obviously, this is too many bits to be contained in 1 8-bit data byte. So, each Quarter Frame message contains just one piece of the time (for example, one Quarter Frame may contain only the hours). In order to get the entire SMPTE time at any given point, a slave needs to receive several Quarter Frame messages, and piece the current SMPTE time together from those messages. It takes 8 Quarter Frame messages to convey the current SMPTE time. In other words, by the time that a slave can piece together the current SMPTE time, two SMPTE frames have passed (ie, since there are 4 Quarter Frame messages in each frame). So, MTC's version of SMPTE time actually counts in increments of 2 SMPTE Frames per each update of the current SMPTE time.

The first (of 8) Quarter Frame message contains the low nibble (ie, bits 0 to 3) of the Frame Time. The second Quarter Frame message contains the high nibble (ie, bits 4 to 7) of the Frame Time. The third and fourth messages contain the low and high nibbles of the Seconds Time. The fifth and sixth messages contain the low and high nibbles of the Minutes Time. The seventh and eighth messages contain the low and high nibbles of the Hours Time. The eighth message also contains the SMPTE frames-per-second Type (ie, 24, 25, 30 drop, or 30 fps). If you were to break up the Quarter Frame's data byte into its 7 bits, the format is:

0nnn dddd

where **nnn** is one of 7 possible values which tell you what **dddd** represents. Here are the 7 values, and

what each causes **dddd** to represent.

```
Value          dddd
  0        Current Frames Low Nibble
  1        Current Frames High Nibble
  2        Current Seconds Low Nibble
  3        Current Seconds High Nibble
  4        Current Minutes Low Nibble
  5        Current Minutes High Nibble
  6        Current Hours Low Nibble
  7        Current Hours High Nibble and SMPTE Type
```

0xF1 0x25

means that the 5 is the low nibble of the Seconds Time (because **nnn** is 2). If the following Quarter Frame is subsequently received,

0xF1 0x32

then this means that 2 is the high nibble of the Seconds Time. Therefore, the current SMPTE Seconds is 0x25 (ie, 37 seconds).

In the data byte for the Hours High Nibble and SMPTE Type, the bits are interpreted as follows:

0nnn **x** yy **d**

where **nnn** is 7. **x** is unused and set to 0. **d** is bit 4 of the Hours Time. **yy** tells the SMPTE Type as follows:

```
0 = 24 fps
1 = 25 fps
2 = 30 fps (Drop-Frame)
3 = 30 fps
```

When MTC is running in the forward direction (ie, time is advancing), the Quarter Frame messages are sent in the order of Frames Low Nibble to Hours High Nibble. In other words, the order looks something like this:

```
0xF1 0x0n
   where n is the current Frames Low Nibble

0xF1 0x1n
   where n is the current Frames High Nibble

0xF1 0x2n
```

where **n** etc.

0xF1 0x3**n**

0xF1 0x4**n**

0xF1 0x5**n**

0xF1 0x6**n**

0xF1 0x7**n**

When MTC is running in reverse (ie, time is going backwards), these are sent in the opposite order, ie, the Hours High Nibble is sent first and the Frames Low Nibble is last.

The arrival of the 0xF1 0x0**n** and 0xF1 0x4**n** messages always denote where SMPTE Frames actually occur in realtime.

Since 8 Quarter Frame messages are required to piece together the current SMPTE time, timing lock can't be achieved until the slave has received all 8 messages. This will take from 2 to 4 SMPTE Frames, depending upon when the slave comes online.

The Frame number (contained in the first 2 Quarter Frame messages) is the SMPTE Frames Time for when the first Quarter Frame message is sent. But, because it takes 7 more Quarter Frames to piece together the current SMPTE Time, when the slave does finally piece the time together, it is actually 2 SMPTE Frames behind the real current time. So, for display purposes, the slave should always add 2 frames to the current time.

Full Frame

For cueing the slave to a particular start point, Quarter Frame messages are not used. Instead, an MTC *Full Frame* message should be sent. The Full Frame is a SysEx message that encodes the entire SMPTE time in one message as so (in hex):

F0 7F **cc** 01 01 **hr** *mn* **sc** *fr* F7

cc is the SysEx channel (0 to 127). It is suggested that a device default to using its Manufacturer's SysEx ID number for this channel, giving the musician the option of changing it. Channel number 0x7F is used to indicate that all devices on the daisy-chain should recognize this Full Frame message.

The **hr**, **mn**, **sc**, and **fr** are the hours, minutes, seconds, and frames of the current SMPTE time. The hours byte also contains the SMPTE Type as per the Quarter Frame's Hours High Nibble message.

The Full Frame simply cues a slave to a particular SMPTE time. The slave doesn't actually start running until it starts receiving Quarter Frame messages. (Which implies that a slave is stopped whenever it is not receiving Quarter Frame messages). The master should pause after sending a Full Frame, and before sending a Quarter Frame, in order to give the slave time to cue to the desired SMPTE time.

During fast forward or rewind (ie, shuttle) modes, the master should not continuously send Quarter Frame messages, but rather, send Full Frame messages at regular intervals.

User Bits

SMPTE also provides for 32 "user bits", information for special functions which vary with each product. (Usually, these bits can only be programmed from equipment that supports such). Upto 4 characters or 8 digits can be written. Examples of use are adding a date code or reel number to a tape. The user bits tend not to change throughout a run of time code, so rather than stuffing this information into a Quarter Frame, MTC provides a separate SysEx message to transmit this info.

F0 7F **cc** 01 02 **u1 u2 u3 u4 u5 u6 u7 u8 u9** F7

cc is the SysEx channel (0 to 127). Only the low nibble of each of the first 8 data bytes is used. Only the 2 low bits of u9 is used.

u1 = 0000**aaaa**
u2 = 0000**bbbb**
u3 = 0000**cccc**
u4 = 0000**dddd**
u5 = 0000**eeee**
u6 = 0000**ffff**
u7 = 0000**gggg**
u8 = 0000**hhhh**
u9 = 000000**ii**

These nibbles decode into an 8-bit format of aaaabbbb ccccdddd eeeeffff ggggghhh ii. It forms 4 8-bit characters, and a 2 bit Format Code. u1 through u8 correspond to the SMPTE Binary Groups 1 through 8. u9 are the 2 Binary Group Flag Bits, defined by SMPTE.

The Users Bits messages can be sent at any time, whenever these values must be passed to some device on the daisy-chain.

Notation Information

There are two Notation Information messages which can be used to setup a device that needs to interact with the musician using musical bars and beats.

Time Signature

The Time Signature message can setup Time Signature or indicate a change of meter.

F0 7F **cc** 03 *ts* **ln** *nn dd* **qq** [nn dd...] F7

cc is the SysEx channel (0 to 127).

ts is 02 if the Time Signature is to be changed now, or 42 if the Time Signature is to be changed at the end of the currently playing measure.

ln is the number of data bytes following this field. Normally, this will be a 3 if there is not a compound time signature in the measure.

nn dd are the Numerator and Denominator of the Time Signature, respectively. Like with MIDI File Format's Time Signature MetaEvent, the Denominator is expressed as a power of 2.

qq is the number of notated 32nd notes in a MIDI quarter note. Again, this is similiar to the same field in MIDI File Format's Time Signature MetaEvent.

[nn dd ...] are optional, additional pairs of num/denom, to define a compound time signature within the same measure.

Bar Marker

The Bar Marker message indicates the start of a musical measure. It could also be used to setup and mark off bars of an introductory "count down".

F0 7F **cc** 03 01 **lb mb** F7

cc is the SysEx channel (0 to 127).

lb mb is the desired bar number, with the LSB first (ie, Intel order). This is a signed 14-bit value (low 7 bits are in lb, right-justified, and bits 8 to 14 are in mb, right-justified). Zero and negative numbers up to -8,190 indicate count off measures. For example, a value of -1 (ie, lb mb = 7F 7F) means that there is a one measure introduction. A value of zero would indicate no count off. Positive values indicate measures of the piece. The first measure is bar 1 (ie, lb mb = 01 00). A maximum neg number (lb mb = 00 40) indicates "stopped play" condition. A maximum positive value (lb mb = 7E 3F) indicates running condition, but no idea about measure number. This would be used by a device

wishing to mark the passage of measures without keeping track of the actual measure number.

Setup Message

The Setup message can be used to implement one of 19 defined "events". A master device uses this message to tell slave units what "events" to perform, and when to perform those events. Here's the general template for the message.

F0 7F **cc** 04 *id* **hr mn sc fr ff** *sl sm* **[more info]** F7

cc is the SysEx channel (0 to 127).

hr mn sc fr ff is the SMPTE time when the event is to occur. This is just like the Full Frame message, except that there is also a fractional frame parameter, ff, which is 1/100 of a frame (ie, a value from 0 to 99).

sl sm is this event's 14-bit Event Number (0 to 16,383). sl is bits 0 to 6, and sm is bits 7 to 13.

id tells what this Event Type is. Depending upon the Type, the message may have additional bytes placed where is. The following values for Event Types are defined, and here's what each does.

Special (00)

Contains the setup information that affects a device globally, as opposed to individual tracks, sounds, programs, sequences, etc.). In this case, the Event Number is actually a word which further describes what the event is, as so:

Time Code Offset (00 00) refers to a relative Time Code offset for each unit. For example, a piece of video and a piece of music that are supposed to go together may be created at different times, and likely have different absolute time code positions. Therefore, one must be offset from the other so that they will match up. Each slave on the daisy-chain needs its own offset so that all can be matched up to the master's SMPTE start time.

Enable Event List (01 00) means for a slave to enable execution of events in its internal "list of events" when each one's respective SMPTE time occurs.

Disable Event List (02 00) means for a slave to disable execution of events in its internal "list of events", but not to erase the list.

Clear Event List (03 00) means for a slave to erase all events in its internal list.

System Stop (04 00) refers to a time when the slave may shut down. This serves as a protection against Event Starts without Event Stops, tape machines running past the end of a reel, etc.

Event List Request (05 00) is sent by the master, and requests the slave (whose channel matches the message) to send all events in its list as a series of Setup messages, starting from the SMPTE time in this message.

NOTE: For the first 5 Special messages, the SMPTE time isn't used and is ignored.

Punch In (01) and Punch Out (02)

These refer to the enabling and disabling of record mode on a slave. The Event Number refers to the track to be recorded. Multiple Punch In and Punch Out points (and any of the other Event Types below) may be specified by sending multiple Setup messages with different SMPTE times.

Delete Punch In (03) and Delete Punch Out (04)

Deletes the Punch In or Punch Out (with the matching Event Number and SMPTE Time) from the slave's event list. In other words, it deletes a previously sent Punch In or Punch Out Setup message.

Event Start (05) and Event Stop (06)

These refer to the start/stop (ie, playback) of some continuous action (ie, an action that begins when an Event Start is received, and continues until an Event Stop is received). The Event Number refers to which action on the slave is to be started/stopped. Such actions may include playback of a specific looped waveform, a fader moving on an automated mixer, etc.

Event Start (07) and Event Stop (08) with additional info

Almost the same as the above 2 Event Types, but these have additional bytes before the final 0xF7. Such additional bytes could be for an effect unit's changing parameters, the volume level of a sound effect being adjusted, etc. The additional info should be nibblized with the lowest bits first. For example, if the Note On message 0x91 0x46 0x7F was to be encoded in some additional info bytes, they would be 0x01 0x09 0x06 0x04 0x0F 0x07.

Delete Event Start (09) and Delete Event Stop (0A)

Deletes the Event Start or Event Stop (with the matching Event Number and SMPTE Time) from the slave's event list. In other words, it deletes a previously sent Event Start or Event Stop Setup message (either the Types without additional info, or with additional info).

Cue Point (0B)

Sets an action to be triggered (ie, an action that does something once and automatically stops afterward) or a marker at the specified SMPTE time. These include a "hit" point for a sound effect, a

marker for an edit point, etc. The Event Number should represent the action or marker. For example, Event Number 3 could be to trigger a car crash sound effect. Then, several car crashes could be specified by sending several Cue Point Setup messages, each with Event Number 3, but different SMPTE times.

Cue Point (0C) with additional info

Like the above, but this message may have additional bytes before the final 0xF7. Such additional bytes could be for an effect unit's parameters, the volume level of a sound effect, etc. The additional info should be nibblized with the lowest bits first.

Delete Cue Point (0D)

Deletes one of the preceding 2 Setup messages (with the same Event Number and SMPTE time) from the slave's event list.

Event Name (0E) with additional info)

This assigns an ascii name to the event with the matching Event Number and SMPTE time. It for the musician's point of reference. The additional info bytes are the ascii name. For a newline character, include both a carriage return (0x0A) and line feed (0x0D). The ascii bytes are nibblized. For example, ascii 'A' (0x41) becomes the two bytes, 0x01 0x04.

Summary of Play Mode

To summarize the interaction between master and slave depending upon "play mode":

Play Mode

The master is in normal play at normal or vari-speed rates. The master is sending Quarter Frame messages to the slave. The messages are in ascending order, starting with 0xF1 0x0**n** and ending with 0xF1 0x7**n**. If the master is capable of reverse play, then the messages are sent in reverse, starting with 0xF1 0x7**n** and ending with 0xF1 0x0**n**.

Cue Mode

The master is being "rocked" or "cued" by hand. For example, a tape machine may have the tape still in contact with the playback head so that the musician can cue the contents of the tape to a specific point. The master is sending Quarter Frame messages to the slave. The messages are in ascending order, starting with 0xF1 0x0**n** and ending with 0xF1 0x7**n**. If the master is playing in a reverse

direction, then the messages are sent in reverse, starting with 0xF1 0x7**n** and ending with 0xF1 0x0**n**. Because the musician may be changing the tape direction rapidly, the order of the Quarter Frames must change along with the tape direction.

Fast Forward or Rewind Mode

The master is rewinding or fast forwarding tape. No contact is made with the playback head. So, no cueing is happening. Therefore, the master only need send the slave periodic Full Frame messages at regular intervals as a rough indication of the master's position. The SMPTE time indicated by the last Full Frame message actually takes affect upon the reception of the next Quarter Frame message (ie, when Play Mode resumes).

Standard MIDI File (SMF) Format

The **Standard MIDI File** (SMF) is a file format used to store MIDI data (plus some other kinds of data typically needed by a sequencer).

This format stores the standard MIDI messages (ie, status bytes with appropriate data bytes) plus a time-stamp for each message (ie, a series of bytes that represent how many clock pulses to wait before "playing" the event). The format also allows saving information about tempo, time and key signatures, the names of tracks and patterns, and other information typically needed by a sequencer. One SMF can store information for numerous patterns and tracks so that any sequencer can preserve these structures when loading the file.

NOTE: A **track** usually is analogous to one musical part, such as a Trumpet part. A **pattern** would be analogous to all of the musical parts (ie, Trumpet, Drums, Piano, etc) for one song.

The format was designed to be generic so that the most important data can be read by all sequencers. Think of a MIDI file as a musical version of an ASCII text file (except that the MIDI file contains binary data), and the various sequencer programs as text editors all capable of reading that file. But, unlike ASCII, MIDI file format saves data in **chunks** (ie, groups of bytes preceded by an ID and size) which can be parsed, loaded, skipped, etc. Therefore, SMF format is flexible enough for a particular sequencer to store its own proprietary, "extra" data in such a way that another sequencer won't be confused when loading the file and can safely ignore this extra stuff that it doesn't need. For example, maybe a sequencer wants to save a "flag byte" that indicates whether the user has turned on an audible metronome click. The sequencer can save this flag byte in such a way that another sequencer can skip this byte without having to understand what that byte is for. In the future, the SMF format can also be extended to include new "official"

chunks that all sequencer programs may elect to load and use. This can be done without making old data files obsolete, nor making old sequencers no longer able to load the new files. So, the format is designed to be extensible in a backwardly compatible way.

Of course, SMF files may be used by other MIDI software than just sequencers. Since SMF files can store any and all types of MIDI messages, including System Exclusive messages, they may be used to store/load data by all kinds of MIDI software, such as a Patch Editor that wants to save some System Exclusive messages it received from a MIDI module. (The "timestamp" for each message may be irrelevant to such a Patch Editor. But it's easily ignored for programs that don't really need it).

In conclusion, any software that saves or loads MIDI data should use SMF format for its data files.

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In conclusion, any software that saves or loads MIDI data should use SMF format for its data files.

Data is always saved within a **chunk**. There can be many chunks inside of a MIDI file.

Each chunk can be a different size (and likely will be). A chunk's size is how many (8-bit) bytes are contained in the chunk.

The data bytes in a chunk are typically related in some way. For example, all of the bytes in one chunk may be for one particular sequencer track. The bytes for another sequencer track may be put in a different chunk. So, a chunk is simply a group of related bytes.

Each chunk must begin with a 4 character (ie, 4 ascii bytes) **ID** which tells what "type" of chunk this is.

The next 4 bytes must form a 32-bit length (ie, size) of the chunk.

All chunks must begin with these two fields (ie, 8 bytes), which are referred to as the **chunk header**.

Here's what a chunk's header looks like if you defined it in C:

```
struct CHUNK_HEADER
{
   char           ID[4];
   unsigned long  Length;
};
```

NOTE: The **Length** does not include the 8 byte chunk header. It simply tells you how many bytes of data are in the chunk following this header.

And here's an example chunk header (with bytes expressed in hex) if you examined it with a hex editor:

4D 54 68 64 00 00 00 06

Note that the first 4 bytes make up the ascii ID of **MThd** (ie, the first four bytes are the ascii values for 'M', 'T', 'h', and 'd'). The next 4 bytes tell us that there should be 6 more data bytes in the chunk (and after that we should find the next

chunk header or the end of the file).

NOTE: The 4 bytes that make up the **Length** are stored in (Motorola) "Big Endian" byte order, not (Intel) "Little Endian" reverse byte order. (ie, The 06 is the fourth byte instead of the first of the four).

In fact, all MIDI files begin with the above **MThd header** (and that's how you know that it's a MIDI file).

The MThd header has an ID of **MThd**, and a Length of **6**.

Let's examine the 6 data bytes (which follow the MThd header) in an MThd chunk.

The first two data bytes tell the **Format** (which I prefer to call "type"). There are actually 3 different types (ie, formats) of MIDI files. A type of 0 means that the file contains one single track containing midi data on possibly all 16 midi channels. If your sequencer sorts/stores all of its midi data in one single block of memory with the data in the order that it's "played", then it should read/write this type. A type of 1 means that the file contains one or more simultaneous (ie, all start from an assumed time of 0) tracks, perhaps each on a single midi channel. Together, all of these tracks are considered one sequence or pattern. If your sequencer separates its midi data (i.e. tracks) into different blocks of memory but plays them back simultaneously (ie, as one "pattern"), it will read/write this type. A type of 2 means that the file contains one or more sequentially independant single-track patterns. If your sequencer separates its midi data into different blocks of memory, but plays only one block at a time (ie, each block is considered a different "excerpt" or "song"), then it will read/write this type.

The next 2 bytes tell how many tracks are stored in the file, **NumTracks**. Of course, for format type 0, this is always 1. For the other 2 types, there can be numerous tracks.

The last two bytes indicate how many Pulses (i.e. clocks) Per Quarter Note (abbreviated as PPQN) resolution the time-stamps are based upon, **Division**. For example, if your sequencer has 96 ppqn, this field would be (in hex):

00 60

NOTE: The 2 bytes that make up the **Division** are stored in (Motorola) "Big Endian" byte order, not (Intel) "Little Endian" reverse byte order. The same is true for the **NumTracks** and **Format**.

Alternately, if the first byte of Division is negative, then this represents the division of a second that the time-stamps are based upon. The first byte will be -24, -25, -29, or -30, corresponding to the 4 SMPTE standards representing frames per second. The second byte (a positive number) is the resolution within a frame (ie, subframe). Typical values may be 4 (MIDI Time Code), 8, 10, 80 (SMPTE bit resolution), or 100.

You can specify millisecond-based timing by the data bytes of -25 and 40 subframes.

Here's what an MThd chunk looks like if you defined it in C:

```
struct MTHD_CHUNK
{
   /* Here's the 8 byte header that all chunks must have */
   char           ID[4];  /* This will be 'M','T','h','d' */
   unsigned long  Length; /* This will be 6 */

   /* Here are the 6 bytes */
   unsigned short Format;
   unsigned short NumTracks;
   unsigned short Division;
};
```

And here's an example of a complete MThd chunk (with header) if you examined it in a hex editor:

```
4D 54 68 64      MThd ID
00 00 00 06      Length of the MThd chunk is always 6.
00 01            The Format type is 1.
00 02            There are 2 MTrk chunks in this file.
E7 28            Each increment of delta-time represents a
millisecond.
```

After the MThd chunk, you should find an **MTrk chunk**, as this is the only other currently defined chunk. (If you find some other chunk ID, it must be proprietary to some other program, so skip it by ignoring the following data bytes indicated by the chunk's Length).

An MTrk chunk contains all of the midi data (with timing bytes), plus optional non-midi data for one track. Obviously, you should encounter as many MTrk chunks in the file as the MThd chunk's NumTracks field indicated.

The MTrk header begins with the ID of **MTrk**, followed by the Length (ie, number of data bytes for this track). The Length will likely be different for each track. (After all, a track containing the violin part for a Bach concerto will likely contain more data than a track containing a simple 2 bar drum beat).

Here's what an MTrk chunk looks like if you defined it in C:

```
struct MTRK_CHUNK
{
   /* Here's the 8 byte header that all chunks must have */
   char           ID[4];   /* This will be 'M','T','r','k' */
   unsigned long  Length;  /* This will be the actual size of Data[] */

   /* Here are the data bytes */
   unsigned char  Data[];  /* Its actual size is Data[Length] */
};
```

Think of a track in the MIDI file in the same way that you normally think of a track in a sequencer. A sequencer track contains a series of **events**. For example, the first event in the track may be to sound a middle C note. The second event may be to sound the E above middle C. These two events may both happen at the same time. The third event may be to release the middle C note. This event may happen a few musical beats after the first two events (ie, the middle C note is held down for a few musical beats). Each event has a "time" when it must occur, and the events are arranged within a "chunk" of memory in the order that they occur.

In a MIDI file, an event's "time" precedes the data bytes that make up that event itself. In other words, the bytes that make up the event's time-stamp come first. A given event's time-stamp is referenced from the previous event. For example, if the first event occurs 4 clocks after the start of play, then its "delta-time" is 04. If the next event occurs simultaneously with that first event, its time is 00. So, a delta-time is the duration (in clocks) between an event and the preceding event.

NOTE: Since all tracks start with an assumed time of 0, the first event's delta-time is referenced from 0.

A delta-time is stored as a series of bytes which is called a **variable length quantity**. Only the first 7 bits of each byte is significant (right-justified; sort of like an ASCII byte). So, if you have a 32-bit delta-time, you have to unpack it into a series of 7-bit bytes (ie, as if you were going to transmit it over midi in a SYSEX message). Of course, you will have a variable number of bytes depending upon your delta-time. To indicate which is the last byte of the series, you leave bit #7 clear. In all of the preceding bytes, you set bit #7. So, if a delta-time is between 0-127, it can be represented as one byte. The largest delta-time allowed is 0FFFFFFF, which translates to 4 bytes variable length. Here are examples of delta-times as 32-bit values, and the variable length quantities that they translate to:

NUMBER	VARIABLE QUANTITY
00000000	00
00000040	40

```
0000007F              7F
00000080            81 00
00002000            C0 00
00003FFF            FF 7F
00004000          81 80 00
00100000          C0 80 00
001FFFFF          FF FF 7F
00200000        81 80 80 00
08000000        C0 80 80 00
0FFFFFFF        FF FF FF 7F
```

Here are some C routines to read and write variable length quantities such as delta-times. With **WriteVarLen()**, you pass a 32-bit value (ie, unsigned long) and it spits out the correct series of bytes to a file. **ReadVarLen()** reads a series of bytes from a file until it reaches the last byte of a variable length quantity, and returns a 32-bit value.

```
void WriteVarLen(register unsigned long value)
{
   register unsigned long buffer;
   buffer = value & 0x7F;

   while ( (value >>= 7) )
   {
     buffer <<= 8;
     buffer |= ((value & 0x7F) | 0x80);
   }

   while (TRUE)
   {
      putc(buffer,outfile);
      if (buffer & 0x80)
          buffer >>= 8;
      else
          break;
   }
}

unsigned long ReadVarLen()
{
    register unsigned long value;
    register unsigned char c;
```

```
    if ( (value = getc(infile)) & 0x80 )
    {
       value &= 0x7F;
       do
       {
         value = (value << 7) + ((c = getc(infile)) & 0x7F);
       } while (c & 0x80);
    }

    return(value);
}
```

NOTE: The concept of variable length quantities (ie, breaking up a large value into a series of bytes) is used with other fields in a MIDI file besides delta-times, as you'll see later.

For those not writing in C, you may benefit from a psuedo-code explanation of the above routines. In pseudo-code, ReadVarLen() is:

1. Initialize the variable which will hold the value. Set it to 0. We'll call this variable 'result'.
2. Read the next byte of the Variable Length quantity from the MIDI file.
3. Shift all of the bits in 'result' 7 places to the left. (ie, Multiply 'result' by 128).
4. Logically OR 'result' with the byte that was read in, but first mask off bit #7 of the byte. (ie, AND the byte with hexadecimal 7F before you OR with 'result'. But make sure you save the original value of the byte for the test in the next step).
5. Test if bit #7 of the byte is set. (ie, Is the byte AND hexadecimal 80 equal to hexadecimal 80)? If so, loop back to step #2. Otherwise, you're done, and 'result' now has the appropriate value.

In pseudo code, WriteVarLen() could be:

1. Assume that you have a variable named 'result' which contains the value to write out as a Variable Length Quantity.
2. Declare an array which can contain 4 numbers. We'll call this variable 'array'. Initialize a variable named 'count' to 0.

3. Is 'result' less than 128? If so, skip to step #8.
4. Take the value 'result' AND with hexadecimal 7F, and OR with hexadecimal 80, and store it in 'count' element of 'array'. (ie, The first time through the loop, this gets stored in the first element of 'array'). NOTE: Don't alter the value of 'result' itself.
5. Increment 'count' by 1.
6. Shift all bits in 'result' 7 places to the right. (This can be done by dividing by 128).
7. Loop back to step #3.
8. Take the value 'result' AND with hexadecimal 7F, and store it in 'count' element of 'array'.
9. Increment 'count' by 1.
10. Write out the values stored in 'array'. Start with the last element stored above, and finish with the first element stored. (ie, Write them out in reverse order so that the first element of 'array' gets written to the MIDI file last). NOTE: The variable 'count' tells you how many total bytes to write. It also can be used as an index into the array (if you subtract one from it, and keep writing out bytes until it is -1).

An MTrk can contain MIDI events and non-MIDI events (ie, events that contain data such as tempo settings, track names, etc).

The first (1 to 4) byte(s) in an MTrk will be the first event's delta-time as a variable length quantity. The next data byte is actually the first byte of that event itself. I'll refer to this as the event's **Status**. For MIDI events, this will be the actual MIDI Status byte (or the first midi data byte if running status). For example, if the byte is hex 90, then this event is a **Note-On** upon midi channel 0. If for example, the byte was hex 23, you'd have to recall the previous event's status (ie, midi running status). Obviously, the first MIDI event in the MTrk must have a status byte. After a midi status byte comes its 1 or 2 data bytes (depending upon the status - some MIDI messages only have 1 subsequent data byte). After that you'll find the next event's delta time (as a variable quantity), etc.

SYSEX events

SYSEX (system exclusive) events (status = F0) are a special case because a SYSEX event can be any length. After the F0 status (which is always stored -- no running status here), you'll find yet another series of variable length bytes. Combine them with ReadVarLen() and you'll come up with a 32-bit value that tells you how many more bytes follow which make up this SYSEX event. This length doesn't include the F0 status.

For example, consider the following SYSEX MIDI message:

F0 7F 7F 04 01 7F 7F F7

This would be stored in a MIDI file as the following series of bytes (minus the delta-time bytes which would precede it):

F0 07 7F 7F 04 01 7F 7F F7

The 07 above is the variable length quantity (which happens to fit in just one byte for this example). It indicates that there are seven, following bytes that

comprise this SYSEX message.

Really oddball midi units send a system exclusive message as a series of small "packets" (with a time delay inbetween transmission of each packet). The first packet begins with an F0, but it doesn't end with an F7. The subsequent packets don't start with an F0 nor end with F7. The last packet doesn't start with an F0, but does end with the F7. So, between the first packet's opening F0 and the last packet's closing F7, there's 1 SYSEX message there. (Note: only extremely poor designs, such as the crap marketed by Casio exhibit such horrid behavior). Of course, since a delay is needed inbetween each packet, you need to store each packet as a separate event with its own time in the MTrk. Also, you need some way of knowing which events shouldn't begin with an F0 (ie, all of them except the first packet). So, the MIDI file redefines a midi status of F7 (normally used as an end mark for SYSEX packets) as a way to indicate an event that doesn't begin with F0. If such an event follows an F0 event, then it's assumed that the F7 event is the second "packet" of a series. In this context, it's referred to as a SYSEX CONTINUATION event. Just like the F0 type of event, it has a variable length followed by data bytes. On the other hand, the F7 event could be used to store MIDI REALTIME or MIDI COMMON messages. In this case, after the variable length bytes, you should expect to find a MIDI Status byte of F1, F2, F3, F6, F8, FA, FB, FC, or FE. (Note that you wouldn't find any such bytes inside of a SYSEX CONTINUATION event). When used in this manner, the F7 event is referred to as an ESCAPED event.

Non-MIDI events

A status of FF is reserved to indicate a special non-MIDI event. (Note that FF is used in MIDI to mean "reset", so it wouldn't be all that useful to store in a data file. Therefore, the MIDI file arbitrarily redefines the use of this status). After the FF status byte is another byte that tells you what **Type** of non-MIDI event it is. It's sort of like a second status byte. Then after this byte is another byte(s -- a variable length quantity again) that tells how many more data bytes follow in this event (ie, its Length). This Length doesn't include the FF, Type byte, nor the Length byte. These special, non-MIDI events are called **Meta-**

Events, and most are optional unless otherwise noted. The section of this online book entitled "Meta-Events" lists the currently defined Meta-Events. Note that unless otherwise mentioned, more than one of these events can be placed in an MTrk (even the same Meta-Event) at any delta-time. (Just like all midi events, Meta-Events have a delta-time from the previous event regardless of what type of event that may be. So, you can freely intermix MIDI and Meta events).

Sequence Number

FF 00 02 ***ss ss***

or...

FF 00 00

This optional event must occur at the beginning of a MTrk (ie, before any non-zero delta-times and before any midi events). It specifies the sequence number. The two data bytes ***ss ss***, are that number which corresponds to the **MIDI Cue** message. In a format 2 MIDI file, this number identifies each "pattern" (ie, Mtrk) so that a "song" sequence can use the MIDI Cue message to refer to patterns.

If the ***ss ss*** numbers are omitted (ie, the second form shown above), then the MTrk's location in the file is used. (ie, The first MTrk chunk is sequence number 0. The second MTrk is sequence number 1. Etc).

In format 0 or 1, which contain only one "pattern" (even though format 1 contains several MTrks), this event is placed in only the first MTrk. So, a group of format 0 or 1 files with different sequence numbers can comprise a "song collection".

There can be only one of these events per MTrk chunk in a Format 2. There can be only one of these events in a Format 0 or 1, and it must be in the first MTrk.

Text

FF 01 ***len text***

Any amount of text (amount of bytes = ***len***) for any purpose. It's best to put this event at the beginning of an MTrk. Although this text could be used for any purpose, there are other text-based Meta-Events for such things as orchestration, lyrics, track name, etc. This event is primarily used to add "comments" to a MIDI file which a program would be expected to ignore when loading that file.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Copyright

FF 02 ***len text***

A copyright message. It's best to put this event at the beginning of an MTrk.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Sequence/Track Name

FF 03 ***len text***

The name of the sequence or track. It's best to put this event at the beginning of an MTrk.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Instrument

FF 04 ***len text***

The name of the instrument (ie, MIDI module) being used to play the track. This may be different than the Sequence/Track Name. For example, maybe the name of your sequence (ie, Mtrk) is "Butterfly", but since the track is played upon a Roland S-770, you may also include an Instrument Name of "Roland S-770".

It's best to put one (or more) of this event at the beginning of an MTrk to provide the user with identification of what instrument(s) is playing the track. Usually, the instruments (ie, patches, tones, banks, etc) are setup on the audio devices via **MIDI Program Change** and **MIDI Bank Select Controller** events within the MTrk. So, this event exists merely to provide the user with visual feedback of what instruments are used for a track.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Lyric

FF 05 ***len text***

A song lyric which occurs on a given beat. A single Lyric MetaEvent should contain only one syllable.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Marker

FF 06 ***len text***

The text for a marker which occurs on a given beat. Marker events might be used to denote a loop start and loop end (ie, where the sequence loops back to a previous event).

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Cue Point

FF 07 ***len text***

The text for a cue point which occurs on a given beat. A Cue Point might be used to denote where a WAVE (ie, sampled sound) file starts playing, for example, where the ***text*** would be the WAVE's filename.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Program Name

FF 08 ***len text***

The name of the program (ie, patch) used to play the MTrk. This may be different than the Sequence/Track Name. For example, maybe the name of your sequence (ie, Mtrk) is "Butterfly", but since the track is played upon an electric piano patch, you may also include a Program Name of "ELECTRIC PIANO".

Usually, the instruments (ie, patches, tones, banks, etc) are setup on the audio devices via **MIDI Program Change** and **MIDI Bank Select Controller** events within the MTrk. So, this event exists merely to provide the user with visual feedback of what particular patch is used for a track. But it can also give a hint to intelligent software if patch remapping needs to be done. For example, if the MIDI file was created on a non-General MIDI instrument, then the **MIDI Program Change** event will likely contain the wrong value when played on a General MIDI instrument. Intelligent software can use the Program Name event to look up the correct value for the **MIDI Program Change** event.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

Device (Port) Name

FF 09 ***len text***

The name of the MIDI device (port) where the track is routed. This replaces the "MIDI Port" Meta-Event which some sequencers formally used to route MIDI tracks to various MIDI ports (in order to support more than 16 MIDI channels).

For example, assume that you have a MIDI interface that has 4 MIDI output ports. They are listed as "MIDI Out 1", "MIDI Out 2", "MIDI Out 3", and "MIDI Out 4". If you wished a particular MTrk to use "MIDI Out 1" then you would put a Port Name Meta-event at the beginning of the MTrk, with "MIDI Out 1" as the ***text***.

All MIDI events that occur in the MTrk, after a given Port Name event, will be routed to that port.

In a format 0 MIDI file, it would be permissible to have numerous Port Name events intermixed with MIDI events, so that the one MTrk could address numerous ports. But that would likely make the MIDI file much larger than it need be. The Port Name event is useful primarily in format 1 MIDI files, where each MTrk gets routed to one particular port.

Note that **len** could be a series of bytes since it is expressed as a variable length quantity.

End of Track

FF 2F 00

This event is not optional. It must be the last event in every MTrk. It's used as a definitive marking of the end of an MTrk. Only 1 per MTrk.

Tempo

FF 51 03 ***tt tt tt***

Indicates a tempo change. The 3 data bytes of **tt tt tt** are the tempo in microseconds per quarter note. In other words, the microsecond tempo value tells you how long each one of your sequencer's "quarter notes" should be. For example, if you have the 3 bytes of 07 A1 20, then each quarter note should be 0x07A120 (or 500,000) microseconds long.

So, the MIDI file format expresses tempo as "the amount of time (ie, microseconds) per quarter note".

NOTE: If there are no tempo events in a MIDI file, then the tempo is assumed to be 120 BPM

In a format 0 file, the tempo changes are scattered throughout the one MTrk. In format 1, the very first MTrk should consist of only the tempo (and time signature) events so that it could be read by some device capable of generating a "tempo map". It is best not to place MIDI events in this MTrk. In format 2, each MTrk should begin with at least one initial tempo (and time signature) event.

See also: Tempo and Timebase.

SMPTE Offset

FF 54 05 ***hr mn se fr ff***

Designates the SMPTE start time (hours, minutes, seconds, frames, subframes) of the MTrk. It should be at the start of the MTrk. The hour should not be encoded with the SMPTE format as it is in **MIDI Time Code**. In a format 1 file, the SMPTE OFFSET must be stored with the tempo map (ie, the first MTrk), and has no meaning in any other MTrk. The ***ff*** field contains fractional frames in 100ths of a frame, even in SMPTE based MTrks which specify a different frame subdivision for delta-times (ie, different from the subframe setting in the MThd).

Time Signature

FF 58 04 ***nn dd cc bb***

Time signature is expressed as 4 numbers. ***nn*** and ***dd*** represent the "numerator" and "denominator" of the signature as notated on sheet music. The denominator is a negative power of 2: 2 = quarter note, 3 = eighth, etc.

The ***cc*** expresses the number of MIDI clocks in a metronome click.

The ***bb*** parameter expresses the number of notated 32nd notes in a MIDI quarter note (24 MIDI clocks). This event allows a program to relate what MIDI thinks of as a quarter, to something entirely different.

For example, 6/8 time with a metronome click every 3 eighth notes and 24 clocks per quarter note would be the following event:

FF 58 04 06 03 18 08

NOTE: If there are no time signature events in a MIDI file, then the time signature is assumed to be 4/4.

In a format 0 file, the time signatures changes are scattered throughout the one MTrk. In format 1, the very first MTrk should consist of only the time signature (and tempo) events so that it could be read by some device capable of generating a "tempo map". It is best not to place MIDI events in this MTrk. In format 2, each MTrk should begin with at least one initial time signature (and tempo) event.

Key Signature

FF 59 02 ***sf mi***

sf = -7 for 7 flats, -1 for 1 flat, etc, 0 for key of c, 1 for 1 sharp, etc.

mi = 0 for major, 1 for minor

Proprietary Event

FF 7F ***len data***

This can be used by a program to store proprietary data. The first byte(s) should be a unique ID of some sort so that a program can identity whether the event belongs to it, or to some other program. A 4 character (ie, ascii) ID is recommended for such.

Note that ***len*** could be a series of bytes since it is expressed as a variable length quantity.

The MIDI file format's Tempo Meta-Event expresses tempo as "the amount of time (ie, microseconds) per quarter note". For example, if a Tempo Meta-Event contains the 3 bytes of 07 A1 20, then each quarter note should be 0x07A120 (or 500,000) microseconds long.

BPM

Normally, musicians express tempo as "the amount of quarter notes in every minute (ie, time period)". This is the opposite of the way that the MIDI file format expresses it.

When musicians refer to a "beat" in terms of tempo, they are referring to a quarter note (ie, a quarter note is always 1 beat when talking about tempo, regardless of the time signature. Yes, it's a bit confusing to non-musicians that the time signature's "beat" may not be the same thing as the tempo's "beat" -- it won't be unless the time signature's beat also happens to be a quarter note. But that's the traditional definition of BPM tempo). To a musician, tempo is therefore always "how many quarter notes happen during every minute". Musicians refer to this measurement as BPM (ie, Beats Per Minute). So a tempo of 100 BPM means that a musician must be able to play 100 steady quarter notes, one right after the other, in one minute. That's how "fast" the "musical tempo" is at 100 BPM. It's very important that you understand the concept of how a musician expresses "musical tempo" (ie, BPM) in order to properly present tempo settings to a musician, and yet be able to relate it to how the MIDI file format expresses tempo.

To convert the Tempo Meta-Event's tempo (ie, the 3 bytes that specify the amount of microseconds per quarter note) to BPM:

BPM = 60,000,000/(tt tt tt)

For example, a tempo of 120 BPM = 07 A1 20 microseconds per quarter note.

So why does the MIDI file format use "time per quarter note" instead of "quarter notes per time" to specify its tempo? Well, its easier to specify more precise tempos with the former. With BPM, sometimes you have to deal with

fractional tempos (for example, 100.3 BPM) if you want to allow a finer resolution to the tempo. Using microseconds to express tempo offers plenty of resolution.

Also, SMPTE is a time-based protocol (ie, it's based upon seconds, minutes, and hours, rather than a musical tempo). Therefore it's easier to relate the MIDI file's tempo to SMPTE timing if you express it as microseconds. Many musical devices now use SMPTE to sync their playback.

PPQN Clock

A sequencer typically uses some internal hardware timer counting off steady time (ie, microseconds perhaps) to generate a software "PPQN clock" that counts off the timebase (Division) "ticks". In this way, the time upon which an event occurs can be expressed to the musician in terms of a musical bar:beat:PPQN-tick rather than how many microseconds from the start of the playback. Remember that musicians always think in terms of a beat, not the passage of seconds, minutes, etc.

As mentioned, the microsecond tempo value tells you how long each one of your sequencer's "quarter notes" should be. From here, you can figure out how long each one of your sequencer's PPQN clocks should be by dividing that microsecond value by your MIDI file's Division. For example, if your MIDI file's Division is 96 PPQN, then that means that each of your sequencer's PPQN clock ticks at the above tempo should be 500,000 / 96 (or 5,208.3) microseconds long (ie, there should be 5,208.3 microseconds inbetween each PPQN clock tick in order to yield a tempo of 120 BPM at 96 PPQN. And there should always be 96 of these clock ticks in each quarter note, 48 ticks in each eighth note, 24 ticks in each sixteenth, etc).

Note that you can have any timebase at any tempo. For example, you can have a 96 PPQN file playing at 100 BPM just as you can have a 192 PPQN file playing at 100 BPM. You can also have a 96 PPQN file playing at either 100 BPM or 120 BPM. Timebase and tempo are two entirely separate quantities. Of course, they both are needed when you setup your hardware timer (ie, when

you set how many microseconds are in each PPQN tick). And of course, at slower tempos, your PPQN clock tick is going to be longer than at faster tempos.

MIDI Clock

MIDI clock bytes are sent over MIDI, in order to sync the playback of 2 devices (ie, one device is generating MIDI clocks at its current tempo which it internally counts off, and the other device is syncing its playback to the receipt of these bytes). Unlike with SMPTE frames, MIDI clock bytes are sent at a rate related to the musical tempo.

Since there are 24 MIDI Clocks in every quarter note, the length of a MIDI Clock (ie, time inbetween each MIDI Clock message) is the microsecond tempo divided by 24. In the above example, that would be 500,000/24, or 20,833.3 microseconds in every MIDI Clock. Alternately, you can relate this to your timebase (ie, PPQN clock). If you have 96 PPQN, then that means that a MIDI Clock byte must occur every 96 / 24 (ie, 4) PPQN clocks.

SMPTE

SMPTE counts off the passage of time in terms of seconds, minutes, and hours (ie, the way that non-musicians count time). It also breaks down the seconds into smaller units called "frames". The movie industry created SMPTE, and they adopted 4 different frame rates. You can divide a second into 24, 25, 29, or 30 frames. Later on, even finer resolution was needed by musical devices, and so each frame was broken down into "subframes".

So, SMPTE is not directly related to musical tempo. SMPTE time doesn't vary with "musical tempo".

Many devices use SMPTE to sync their playback. If you need to sychronize with such a device, then you may need to deal with SMPTE timing. Of course,

you're probably still going to have to maintain some sort of PPQN clock, based upon the passing SMPTE subframes, so that the user can adjust the tempo of the playback in terms of BPM, and can consider the time of each event in terms of bar:beat:tick. But since SMPTE doesn't directly relate to musical tempo, you have to interpolate (ie, calculate) your PPQN clocks from the passing of subframes/frames/seconds/minutes/hours (just as we previously calculated the PPQN clock from a hardware timer counting off microseconds).

Let's take the easy example of 25 Frames and 40 SubFrames. As previously mentioned in the discussion of Division, this is analogous to millisecond based timing because you have 1,000 SMPTE subframes per second. (You have 25 frames per second. Each second is divided up into 40 subframes, and you therefore have 25 * 40 subframes per second. And remember that 1,000 milliseconds are also in every second). Every millisecond therefore means that another subframe has passed (and vice versa). Every time you count off 40 subframes, a SMPTE frame has passed (and vice versa). Etc.

Let's assume you desire 96 PPQN and a tempo of 500,000 microseconds. Considering that with 25-40 Frame-SubFrame SMPTE timing 1 millisecond = 1 subframe (and remember that 1 millisecond = 1,000 microseconds), there should be 500,000 / 1,000 (ie, 500) subframes per quarter note. Since you have 96 PPQN in every quarter note, then every PPQN ends up being 500 / 96 subframes long, or 5.2083 milliseconds (ie, there's how we end up with that 5,208.3 microseconds PPQN clock tick just as we did above in discussing PPQN clock). And since 1 millisecond = 1 subframe, every PPQN clock tick also equals 5.2083 subframes at the above tempo and timebase.

Formulas

BPM = 60,000,000/MicroTempo
MicrosPerPPQN = MicroTempo/TimeBase
MicrosPerMIDIClock = MicroTempo/24
PPQNPerMIDIClock = TimeBase/24
MicrosPerSubFrame = 1000000 * Frames * SubFrames
SubFramesPerQuarterNote = MicroTempo/(Frames * SubFrames)

SubFramesPerPPQN = SubFramesPerQuarterNote/TimeBase
MicrosPerPPQN = SubFramesPerPPQN * Frames * SubFrames

To illustrate how best to use various Meta-Events when you save a MIDI file, let's consider an example sequencer track.

Assume that the user has created a sequencer track that he named "My track". He is using a Roland JV-1080 to play this track. His JV-1080 is connected to the third MIDI Out port on his computer's MIDI interface (which supports multiple MIDI Outs -- for more than 16 MIDI channels). The operating system lists this port with a name of "MIDI Out 3". He has picked out a patch on his JV-1080 called "Gonzo Harp". This is not a General MIDI patch. It happens to be in the third bank of patches, and is patch number 0. This track will play on the JV-1080's MIDI channel 1.

How would you write out an MTrk for this sequencer track?

First, you would start off with a Sequence/Track Name Meta-Event. The text for this event would be "My track". (Note that you do not need to nul-terminate text in a Meta-Event. The Meta-Event's variable quantity length tells how many characters are in the name). You would put this at a delta-time of zero. So let's examine our MTrk chunk so far, as if we were using a hex editor:

```
4D 54 72 6B     MTrk ID
00 00 00 0C     Length of the MTrk chunk so far.

                Our Track Name Meta-Event
00              Delta-time is 0.
FF              A Meta-Event.
03              Track Name type.
08              Length of "My track"
4D              ASCII 'M'
79              ASCII 'y'
20              ASCII ' '
74              ASCII 't'
72              ASCII 'r'
61              ASCII 'a'
63              ASCII 'c'
6B              ASCII 'k'
```

Next, you may choose to put an Instrument Name Meta-Event (although this isn't as important as other events, and you may skip it if your software neither knows, nor cares, what specific MIDI module is playing the track). The text for this event would be "Roland JV-1080".

```
                Our Instrument Name Meta-Event
00              Delta-time is 0.
FF              A Meta-Event.
04              Instrument Name type.
0E              Length of "Roland JV-1080"
52              ASCII 'R'
6F              ASCII 'o'
6C              ASCII 'l'
61              ASCII 'a'
6E              ASCII 'n'
64              ASCII 'd'
20              ASCII ' '
4A              ASCII 'J'
56              ASCII 'V'
2D              ASCII '-'
31              ASCII '1'
30              ASCII '0'
38              ASCII '8'
30              ASCII '0'
```

Next, you would put a Device (Port) Name Meta-Event. The text for this event would be "MIDI Out 3".

```
                Our Device (Port) Name Meta-Event
00              Delta-time is 0.
FF              A Meta-Event.
09              Device (Port) Name type.
0A              Length of "MIDI Out 3"
4D              ASCII 'M'
49              ASCII 'I'
44              ASCII 'D'
```

```
49                  ASCII 'I'
20                  ASCII ' '
4F                  ASCII 'O'
75                  ASCII 'u'
74                  ASCII 't'
20                  ASCII ' '
33                  ASCII '3'
```

Next, you would put an Program (Patch) Name Meta-Event to indicate the name of the patch being used. The text for this event would be "Gonzo Harp".

```
                    Our Program (Patch) Name Meta-Event
00                  Delta-time is 0.
FF                  A Meta-Event.
08                  Program (Patch) Name type.
0A                  Length of "Gonzo Harp"
47                  ASCII 'G'
6F                  ASCII 'o'
6E                  ASCII 'n'
7A                  ASCII 'z'
6F                  ASCII 'o'
20                  ASCII ' '
48                  ASCII 'H'
61                  ASCII 'a'
72                  ASCII 'r'
70                  ASCII 'p'
```

Next, you need to put the MIDI Bank Select Controller events and the MIDI Program Change event that select the desired patch ("Gonzo Harp") on the JV-1080. Let's assume that the Bank Select values will be 0 for MSB controller and 3 for LSB. We already said that the patch number is 0.

```
                    Bank Select MSB
00                  Delta-time is 0.
B0                  Controller on channel 1.
00                  Bank Select MSB.
```

```
00              MSB = 0
                Bank Select LSB
00              Delta-time is 0.
B0              Controller on channel 1.
20              Bank Select LSB.
03              LSB = 3
                Program Change
00              Delta-time is 0.
C0              Program Change on channel 1.
00              Patch = 0
```

After this, you would put other data for the track, such as MIDI Note-on, Note-off, etc, and other Meta-Events.

Now let's examine loading that same file created above.

First, you'll encounter the Track Name event. Use this to set the name for the sequencer track which will store this MTrk's data.

Next, you'll encounter the Instrument Name. If your program keeps a database of instruments that are "installed" on the computer, this could come in useful for several reasons. First, if your database lists which instruments are attached to which MIDI ports on the system, then you could ignore any Device (Port) Name event (just in case it happens to refer to someone else's MIDI port that isn't applicable on your system). You'll know which MIDI port to use for this MTrk based solely upon the Instrument Name and your own database. Secondly, if you have that instrument listed in your database, then you choose to forego any patch remapping. (ie, You may choose not to check that the values in any Bank Select and Program Change messages are correct, under the assumption that the desired patch is already located where it should be). At least, it may make any patch remapping easier.

Next, you'll encounter the Device (Port) Name event. If you weren't able to use the Instrument Name to deduce which MIDI port to use on your system, then you should check this port name. If it exists on your system, use that port. If

not, use a default port.

Next, you'll encounter the Program Name event. You can use this to implement patch remapping. If your instrument database also lists what patches are available upon each instrument, then you can check for the existence of this patch in your database. (If you were able to use the Instrument Name, then you already know upon what instrument that patch should be found). Once you find that patch in your database, hopefully your database will contain the correct Bank Select and Program Change values needed to select that patch. In that case, you could ignore the next Bank Select and Program Change events you encounter in the MTrk. What you've effectively done is remap a patch change. So for example, if the MTrk was created upon an instrument with a unique arrangement of patches (and therefore its Bank Select and Program Change events have the wrong values when played upon a different instrument), you've remapped it to play the correct patch on another instrument.

Finally, you'll encounter the Bank Select and Program Change events. If you weren't able to use the Instrument and Program Name Meta-Events to deduce the correct Bank Select and Program Change values, then you may use these events verbatim.

A free Dynamic Link Library called "GenMidi" is available to programmers writing MIDI software. This DLL can be used to maintain a database of Instrument names, and the names of all patches on those instruments (including which Bank Select and Program Change values select each patch). Included with the DLL is a tool that allows a user to create a database of such.

By using this DLL in your program, you can more easily implement patch remapping, and better coordinate with other programs that also use this DLL. Plus, you don't have to develop your own tools to allow users to create such databases.

The DLL is downloadable from the Software Programs page.

The method of saving data in chunks (ie, where the data is preceded by an 8 byte header consisting of a 4 char ID and a 32-bit size field) is the basis for Interchange File Format. You should now read the article About Interchange File Format for background information.

As mentioned, MIDI File format is a "broken" IFF. It lacks a file header at the start of the file. One bad thing about this is that a standard IFF parsing routine will choke on a MIDI file (because it will expect the first 12 bytes to be the group ID, filesize, and type ID fields). In order to fix the MIDI File format so that it strictly adheres to IFF, Microsoft simply made up a 12-byte header that is prepended to MIDI files, and thereby came up with the RMID format. An RMID file begins with the group ID (4 ascii chars) of 'R', 'I', 'F', 'F', followed by the 32-bit filesize field, and then the type ID of 'R', 'M', 'I', 'D'. Then, the chunks of a MIDI file follow (ie, the MThd and MTrk chunks). So, somewhere after the first 12 bytes of an RMID file, then you should find an embedded MIDI file (although there may be other "chunks" of data before it. Simply skip over those as you would any unknown chunk).

Note that chunks within a MIDI file are not padded out (with an extra 0 byte) to an even number of bytes. I don't know as if the RMID format corrects this aberration of the MIDI file format too.

The following Meta-Events are considered obsolete and should not be used. (The MMA would like you to know that they never endorsed their use, although since certain versions of CakeWalk utilized them, you may find existing MIDI files with these events). Use the Device (Port) Name Meta-Event instead of the MIDI Port Meta-Event.

MIDI Channel

FF 20 01 ***cc***

This optional event which normally occurs at the beginning of an MTrk (ie, before any non-zero delta-times and before any MetaEvents except Sequence Number) specifies to which MIDI Channel any subsequent MetaEvent or System Exclusive events are associated. The data byte **cc**, is the MIDI channel, where 0 would be the first channel.

The MIDI spec does not give a MIDI channel to System Exclusive events. Nor do MetaEvents have an imbedded channel. When creating a Format 0 MIDI file, all of the System Exclusive and MetaEvents go into one track, so its hard to associate these events with respective MIDI Voice messages. (ie, For example, if you wanted to name the musical part on MIDI channel 1 "Flute Solo", and the part on MIDI Channel 2 "Trumpet Solo", you'd need to use 2 Track Name MetaEvents. Since both events would be in the one track of a Format 0 file, in order to distinguish which track name was associated with which MIDI channel, you would place a MIDI Channel MetaEvent with a channel number of 0 before the "Flute Solo" Track Name MetaEvent, and then place another MIDI Channel MetaEvent with a channel number of 1 before the "Trumpet Solo" Track Name MetaEvent.

It is acceptable to have more than one MIDI channel event in a given track, if that track needs to associate various events with various channels.

MIDI Port

FF 21 01 ***pp***

This optional event which normally occurs at the beginning of an MTrk (ie,

before any non-zero delta-times and before any midi events) specifies out of which MIDI Port (ie, buss) the MIDI events in the MTrk go. The data byte ***pp***, is the port number, where 0 would be the first MIDI buss in the system.

The MIDI spec has a limit of 16 MIDI channels per MIDI input/output (ie, port, buss, jack, or whatever terminology you use to describe the hardware for a single MIDI input/output). The MIDI channel number for a given event is encoded into the lowest 4 bits of the event's Status byte. Therefore, the channel number is always 0 to 15. Many MIDI interfaces have multiple MIDI input/output busses in order to work around limitations in the MIDI bandwidth (ie, allow the MIDI data to be sent/received more efficiently to/from several external modules), and to give the musician more than 16 MIDI Channels. Also, some sequencers support more than one MIDI interface used for simultaneous input/output. Unfortunately, there is no way to encode more than 16 MIDI channels into a MIDI status byte, so a method was needed to identify events that would be output on, for example, channel 1 of the second MIDI port versus channel 1 of the first MIDI port. This MetaEvent allows a sequencer to identify which MTrk events get sent out of which MIDI port. The MIDI events following a MIDI Port MetaEvent get sent out that specified port.

It is acceptable to have more than one Port event in a given track, if that track needs to output to another port at some point in the track.

Introduction

A sequencer is a machine that "plays" musical performances. It tells equipment that can make musical sounds (ie, play pitches, chords, or make any kind of noise) what musical notes to play, and when to play them. **It does this using MIDI messages.**

Recording a musical performance using MIDI

Most sequencers allow a musician to record his performance into the sequencer just as if he were recording his performance upon a cassette tape. But, there is no tape involved, and the sequencer records MIDI messages instead of the actual sounds. (The messages are usually stored in RAM during recording, and later saved to a floppy disk or hard drive for permanent storage. Most sequencers today save the MIDI messages in MIDI File Format data files). In other words, the sequencer electronically records all of your finger, foot, what-have-you movements. It does this by storing all of the MIDI messages which the electronic musical instrument generates when you physically play that instrument.

The sequencer also keeps track of a "musical beat" during the recording so that it can record the "rhythm" of your movements (ie, the rhythms of your playing). Usually, the sequencer generates a metronome sound which the musician follows during the recording phase in order to have the sequencer accurately capture his rhythms while it is storing the generated MIDI messages.

Playing a musical performance using MIDI

Then, upon playback, the sequencer plays the instrument (that you just did) by electronically recreating all of your gestures. The sequencer does this by sending back (to that instrument's MIDI IN jack) those MIDI messages that the musical instrument generated when you played it. The sequencer doesn't have fingers, so it may send a MIDI message that tells an electronic keyboard to pretend that a human has just pressed middle C for example. The keyboard then sounds middle C.

Sequencers can send MIDI messages that tell the keyboard to release a key that it played (ie, remove the finger from middle C), move the pitch wheel, press down a sustain pedal, turn the volume knob, etc. In other words, a sequencer can fully duplicate a musician's performance. (Note that the keyboard only pretends to do these things. That C key doesn't physically depress as if some ghost is sitting at the keyboard. It does make the appropriate musical pitch though, so it sounds as if there is a ghost playing the keyboard).

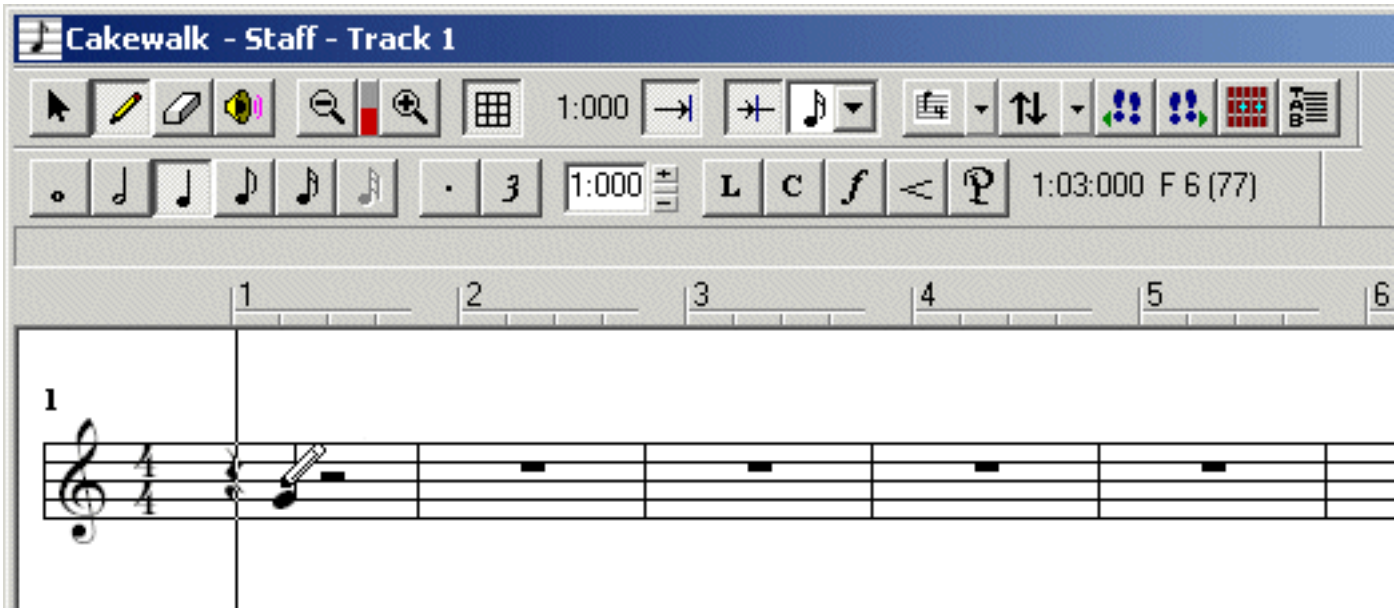
Advantages of sequencers

A sequencer usually has a tempo control so that it can play the performance faster or slower. Unlike with digital audio playback, when you change the tempo of the MIDI sequencer's playback, it doesn't alter the pitch nor the timbre of the sound. (One exception to this general rule is the use of time-stretching to change the tempo of a digital audio recording of a musical performance). It alters only the tempo. So, a sequencer offers complete control of tempo independent of pitch or timbre changes. So too, this means that you can record a part at a slower tempo, and then speed up the playback without any change in timbre or pitch.

MIDI sequencers also offer the facilities to easily transpose musical parts (to other key signatures), mute or solo individual parts, reroute musical parts to play upon various sound modules, and make many other changes in timbre/volume/panning/effects levels/etc to individual musical parts. These changes can all be (theoretically) done in real-time (ie, during playback) without time-consuming calculations made upon the data, unlike with digital audio playback.

The sequencer may have many other features to easily edit the performance (even upon a per note basis) so that the sequencer can correct the mistakes that you recorded into it. It's infinitely easier than trying to correct mistakes recorded onto magnetic tape, or even digital audio recorders, which is why musicians use sequencers so much nowadays.

Most sequencers offer the facility to enter notes (and other control data such as volume/pan/vibrato/etc settings) by "step-entering" the data. You don't have to play the musical part in real-time. Instead, you can slowly enter each musical note (and other setting), and specify what musical bar and beat upon which it gets "played". This is a boon to people whose musical technique is not sufficiently developed to physically play the musical parts in real-time. Some sequencers even offer very intuitive interfaces, such as allowing you to step-enter notes by clicking the mouse pointer upon a graphical sheet of music manuscript drawn upon the screen. Digital audio recorders do not typically have such features. With digital audio recorders, you pretty much have to be able to play some musical parts.



An example of step entry upon Cakewalk's staff view. I moved the pencil over the staff where a G note would occur upon the second beat, and clicked once. This inserted a G note. Because I have the quarter note length button depressed, a quarter note is inserted upon that beat.

A specialized breed of sequencers called "Algorithmic Composition programs" even have the capacity to create a complete musical arrangement on your behalf. See Algorithmic Composition for more details. Digital audio recorders do not offer such a feature.

Computers as sequencers

Most computers have software programs which turn the computer into a sequencer. With a sound card installed inside of the computer, the sequencer can playback musical performances without even needing external MIDI sound modules (since most sound cards now have an internal, multitimbral General MIDI module, usually a wavetable synth, that can recognize and properly "play" the MIDI messages that the sequencer outputs to the sound card's driver).

Examples of sequencer software are CakeWalk, Cubase, Logic, Mark of the Unicorn, etc. There are many such programs, as sequencers are perhaps the most often used tools in music production.

An example of a very simple sequencer is Windows MultiMedia Player. This software does not have any facility to record MIDI messages, nor edit them. It can only playback a MIDI performance (stored in MIDI File Format). Such **simple "playback only" sequencers are often called "MIDI Players".**

Support for digital audio playback

Because MIDI controls only electronic instruments, and can't record the sound of acoustic instruments or human voice, digital audio recording/playback is a feature that is often added to sequencers. The sequencer is designed to record/playback tracks of digital audio in sync with the MIDI tracks. See Digital Audio on a computer for more information about digital audio recording/playback.

Preface

After the MIDI 1.0 standard was finalized in the early 1980's, numerous musical instruments with MIDI jacks appeared upon the market. Musicians started to attach these instruments via their MIDI ports, and quickly discovered that the MIDI 1.0 specification had overlooked some important concerns.

One typical scenario may have been as follows:

A musician attaches his Roland D-10 to his Yamaha DX-7, because he prefers the front panel of the D-10, but prefers the sound of the DX-7, and he wants to use the D-10 to "play" the DX-7. He selects the patch labeled "Piano" on the D-10, and he plays the D-10 keyboard, and on the DX-7 he hears... a trumpet? How did this happen? Well, it happened because MIDI sends a program change message that contains only a patch number -- not the actual name of the patch. So if patch #1 on the DX-7 is a trumpet sound, then that's what he gets on the DX-7, despite the fact that selecting patch #1 on the D-10 yields a piano sound on the D-10. The MIDI 1.0 specification did not require that particular sounds be assigned to particular patch numbers, so every manufacturer used his own discretion as to "patch mapping".

But the real problem was with MIDI files that the musician made. MIDI files contain only MIDI messages. So, any program change event in a MIDI file refers only to a patch number as well -- not the actual patch name. So, this musician creates a MIDI file using his D-10. He has a piano track, so he puts a program change event at the track's beginning to select patch #1, which happens to be a Piano sound on his D-10. He takes that MIDI file to a friend's house. The friend has a DX-7. They play the MIDI file on that DX-7, and suddenly, the piano part is playing with a trumpet sound. Well, that's because patch #1 on the DX-7 is not a piano -- it's a trumpet sound. To "fix" the MIDI file, now the musician with the DX-7 has to edit the MIDI tracks and change every MIDI Program Change event so that it refers to the correct patch number on his DX-7. This deviance among MIDI sound modules made it very difficult for musicians to create MIDI arrangements that played properly upon various MIDI sound modules.

There were also some other deviances among early MIDI modules that made it more difficult to use them together via MIDI. To address these concerns, Roland proposed an addendum to the MIDI 1.0 specification in the late 1980's. This new addendum was called "General MIDI" (GM). It added some new requirements to the base MIDI 1.0 specification (but does not supplant any parts of the 1.0 specification -- the 1.0 specification is still the base level to which all MIDI devices should adhere). GM has now been adopted as part of the MIDI 2.0 specification.

General MIDI Patches

So to make MIDI Program Change messages of more practical use, Roland found it necessary to adopt a standard "patch bank". In other words, what was needed was to assign specific instrument sounds to specific patch numbers. For example, it was decided that patch number 1 upon all sound modules should be the sound of an *Acoustic Grand Piano*. In this way, no matter what MIDI sound module you

use, when you select patch number 1, you always hear some sort of Acoustic Grand Piano sound. A standard was set for 128 patches which must appear in a specific order, and this standard is called *General MIDI* (GM). For example, patch number 25 upon a GM module must be a *Nylon String Guitar*. The chart, GM Patches, shows you the names of all GM Patches, and their respective *Program Change* numbers.

Nowadays, most modules (including the built-in sound modules of computer sound cards) ship with a GM bank (of 128 patches) so that it is easy to play MIDI files upon any MIDI module, without needing to edit all of the Program Change events in the file.

General MIDI Multi-Timbral requirement

Another burgeoning technology in the late 1980's was the multi-timbral module. Typically, there were deviances in the way that various manufacturers implemented this, since the 1.0 specification did not specifically address such devices. For example, some early multi-timbral modules supported only a limited set of the 16 MIDI channels simultaneously, so if you had a MIDI file with tracks upon unsupported MIDI channels, you wouldn't hear those tracks play back. You may not have even realized that those parts weren't being played.

So, one requirement of a GM-compliant module is that it must be fully multi-timbral, meaning that it can play MIDI messages upon all 16 channels simultaneously, with a different GM Patch sounding for each channel.

General MIDI Note Number assignments

There were also deviances in regards to Note Number mapping. For example, some manufacturers mapped middle 'C' to MIDI Note Number 60. Others mapped it to Note Numbers 72 or 48. Some modules even had middle C mapped to various places in different patches, depending upon the instrument. For example, a bass guitar patch may have middle C mapped to the highest C on the keyboard (since the most useful range on a bass guitar is below middle C). A flute patch may have middle C mapped to the lowest C on the keyboard.

The result was that, it became confusing to keep track of which key (ie, MIDI Note Number) played middle C for each patch. Also, when a MIDI track was played back upon certain modules, the part may play back an octave too high or low.

It therefore was decided that all patches must sound an A440 pitch when receiving a MIDI note number of 69. (ie, Note Number 69 plays the A above middle C, and therefore Note Number 60 is middle C).

There were deviances in regards to "drum machines" as well. Most drum machines (and drum units built into multi-timbral modules) play a different drum sound for each MIDI Note Number. But the 1.0 specification never spelled out which drum sounds were assigned to which MIDI note numbers. So, whereas note number 60 may play a snare upon one drum unit, upon another drum unit, it may play a crash cymbal. Again, this caused trouble with MIDI files, since sometimes a drum part would play back with the wrong drum sounds.

To address this discrepancy, the GM addendum contains a "drum map". This assigns about 48 common drum sounds to 48 specific MIDI Note Numbers. The assignments of drum sounds to MIDI notes is shown in the chart, GM Drum Sounds. Also, it was decided that a GM drum unit should default to using MIDI channel 10 to receive MIDI messages. Therefore, a composer of a GM MIDI file can safely assume that his drum part will play correctly if he uses the GM Drum note assignments and records the drum part upon MIDI channel 10.

General MIDI polyphony

Polyphony is how many notes a module can sound simultaneously. For example, perhaps a module can sound 32 notes simultaneously. Early MIDI modules typically had very limited polyphony. For example, the Prophet 5 could sound only 5 notes simultaneously.

This discrepancy in polyphony among MIDI modules made it difficult for arrangers to create MIDI files that played properly upon various modules. For example, if the arranger created too "busy" an arrangement, it could exceed the polyphony of a particular module, and therefore some of the notes may not be heard.

To address this discrepancy, the GM addendum stipulated that a GM module should be capable of sounding at least 24 notes simultaneously. (Ie, It must have 24 note polyphony). It could exceed 24 note polyphony, but it had to have at least that level of polyphony. In this way, if an arranger ensured that he never had more than 24 notes sounding simultaneously in his MIDI file, all notes of his arrangement would be heard upon any GM module.

Other General MIDI requirements

Finally, the GM addendum attempted to address some other discrepancies by spelling out a few more requirements.

A GM module should respond to velocity (ie, for note messages). This typically controls the VCA level (ie, volume) of each note, but the GM addendum unfortunately did not set a specific function for velocity. Some modules may allow velocity to affect other parameters on some patches.

The pitch wheel bend range should default to +/- 2 semitones. This allows an arranger to use pitch

bend messages in his arrangement without worrying whether a bend that is supposed to be up 2 whole steps will instead jump up 2 octaves upon a certain sound module.

The module also should respond to Channel Pressure (often used to control VCA level or VCO level for vibrato depth). Again, the GM addendum unfortunately did not set a specific function for channel pressure, although typically it defaults to controlling the volume of a note while it is being held.

Finally, a GM module should also respond to the following MIDI controller messages: Modulation (1) (usually hard-wired to control LFO amount, ie, vibrato), Channel Volume (7), Pan (10), Expression (11), Sustain (64), Reset All Controllers (121), and All Notes Off (123). Additionally, the module should respond to these Registered Parameter Numbers: Pitch Wheel Bend Range (0), Fine Tuning (1), and Coarse Tuning (2).

There were also some default settings that a GM module should apply upon power up. Channel Volume should default to 90, with all other controllers and effects off (including pitch wheel offset of 0). Initial tuning should be standard, A440 reference.

General MIDI messages

The GM addendum did specify a couple System Exclusive messages to alter settings that are common to all GM units, but which were not addressed by the 1.0 specification.

One such message is for Master Volume -- not just the volume of a patch upon any one MIDI channel, but the master volume of the module itself.

There is also a System Exclusive message that can be used to turn a module's General MIDI mode on or off. This is useful for modules that also offer more expansive, non-GM playback modes or extra, programmable banks of patches beyond the GM set, but need to allow the musician to switch to GM mode when desired.

Conclusion

GM Standard makes it easy for musicians to put *Program Change* messages in their MIDI (sequencer) song files, confident that those messages will select the correct instruments on all GM sound modules, and the song file would therefore play all of the correct instrumentation automatically. Furthermore, musicians need not worry about parts being played back in the wrong octave. Finally, musicians didn't have to worry that a snare drum part, for example, would be played back on a Cymbal. The GM Standard also spells out other minimum requirements that a GM module should meet, such as being able to respond to Pitch and Modulation Wheels, and also being able to play 24 notes simultaneously (with dynamic voice allocation between the 16 Parts). All of these standards help to ensure that MIDI

Files play back properly upon setups of various equipment.

The GM standard is actually not encompassed in the MIDI specification proper (ie, it's an addendum), and there's no reason why someone can't set up the Patches in his sound module to be entirely different sounds than the GM set. After all, most MIDI sound modules offer such programmability. But, most have a GM option so that musicians can easily play the many MIDI files that expect a GM module.

NOTE: The GM Standard doesn't dictate how a module produces sound. For example, one module could use cheap FM synthesis to simulate the Acoustic Grand Piano patch. Another module could use 24 digital audio waveforms of various notes on a piano, mapped out across the MIDI note range, to create that one Piano patch. Obviously, the 2 patches won't sound exactly alike, but at least they will both be piano patches on the 2 modules. So too, GM doesn't dictate VCA envelopes for the various patches, so for example, the Sax patch upon one module may have a longer release time than the same patch upon another module.

GM Patches

This chart shows the names of all 128 GM Instruments, and the MIDI Program Change numbers which select those Instruments.

The patches are arranged into 16 "families" of instruments, with each family containing 8 instruments. For example, there is a *Reed* family. Among the 8 instruments within the Reed family, you will find Saxophone, Oboe, and Clarinet.

Prog#	Instrument	Prog#	Instrument
	PIANO		CHROMATIC PERCUSSION
1	Acoustic Grand	9	Celesta
2	Bright Acoustic	10	Glockenspiel
3	Electric Grand	11	Music Box
4	Honky-Tonk	12	Vibraphone
5	Electric Piano 1	13	Marimba
6	Electric Piano 2	14	Xylophone
7	Harpsichord	15	Tubular Bells
8	Clavinet	16	Dulcimer
	ORGAN		GUITAR
17	Drawbar Organ	25	Nylon String Guitar
18	Percussive Organ	26	Steel String Guitar
19	Rock Organ	27	Electric Jazz Guitar
20	Church Organ	28	Electric Clean Guitar
21	Reed Organ	29	Electric Muted Guitar
22	Accoridan	30	Overdriven Guitar

23 Harmonica
24 Tango Accordian

BASS
33 Acoustic Bass
34 Electric Bass(finger)
35 Electric Bass(pick)
36 Fretless Bass
37 Slap Bass 1
38 Slap Bass 2
39 Synth Bass 1
40 Synth Bass 2

ENSEMBLE
49 String Ensemble 1
50 String Ensemble 2
51 SynthStrings 1
52 SynthStrings 2
53 Choir Aahs
54 Voice Oohs
55 Synth Voice
56 Orchestra Hit

REED
65 Soprano Sax
66 Alto Sax
67 Tenor Sax
68 Baritone Sax
69 Oboe
70 English Horn
71 Bassoon
72 Clarinet

SYNTH LEAD
81 Lead 1 (square)
82 Lead 2 (sawtooth)
83 Lead 3 (calliope)
84 Lead 4 (chiff)
85 Lead 5 (charang)
86 Lead 6 (voice)
87 Lead 7 (fifths)
88 Lead 8 (bass+lead)

SYNTH EFFECTS
97 FX 1 (rain)
98 FX 2 (soundtrack)
99 FX 3 (crystal)

31 Distortion Guitar
32 Guitar Harmonics

SOLO STRINGS
41 Violin
42 Viola
43 Cello
44 Contrabass
45 Tremolo Strings
46 Pizzicato Strings
47 Orchestral Strings
48 Timpani

BRASS
57 Trumpet
58 Trombone
59 Tuba
60 Muted Trumpet
61 French Horn
62 Brass Section
63 SynthBrass 1
64 SynthBrass 2

PIPE
73 Piccolo
74 Flute
75 Recorder
76 Pan Flute
77 Blown Bottle
78 Skakuhachi
79 Whistle
80 Ocarina

SYNTH PAD
89 Pad 1 (new age)
90 Pad 2 (warm)
91 Pad 3 (polysynth)
92 Pad 4 (choir)
93 Pad 5 (bowed)
94 Pad 6 (metallic)
95 Pad 7 (halo)
96 Pad 8 (sweep)

ETHNIC
105 Sitar
106 Banjo
107 Shamisen

100	FX 4 (atmosphere)	**108**	Koto
101	FX 5 (brightness)	**109**	Kalimba
102	FX 6 (goblins)	**110**	Bagpipe
103	FX 7 (echoes)	**111**	Fiddle
104	FX 8 (sci-fi)	**112**	Shanai
	PERCUSSIVE		SOUND EFFECTS
113	Tinkle Bell	**121**	Guitar Fret Noise
114	Agogo	**122**	Breath Noise
115	Steel Drums	**123**	Seashore
116	Woodblock	**124**	Bird Tweet
117	Taiko Drum	**125**	Telephone Ring
118	Melodic Tom	**126**	Helicopter
119	Synth Drum	**127**	Applause
120	Reverse Cymbal	**128**	Gunshot

Prog# refers to the MIDI Program Change number that causes this *Patch* to be selected. These decimal numbers are what the user normally sees on his module's display (or in a sequencer's "Event List"), but note that MIDI modules count the first Patch as 0, not 1. So, the value that is sent in the Program Change message would actually be one less. For example, the Patch number for Reverse Cymbal is actually sent as 119 rather than 120. But, when entering that Patch number using sequencer software or your module's control panel, the software or module understands that humans normally start counting from 1, and so would expect that you'd count the Reverse Cymbal as Patch 120. Therefore, the software or module automatically does this subtraction when it generates the MIDI Program Change message.

So, sending a MIDI Program Change with a value of 120 (ie, actually 119) to a Part causes the Reverse Cymbal Patch to be selected for playing that Part's MIDI data.

GM Drum Sounds

This chart shows what drum sounds are assigned to each MIDI note for a GM module (ie, that has a drum part).

MIDI Note #	Drum Sound	MIDI Note #	Drum Sound
35	Acoustic Bass Drum	**59**	Ride Cymbal 2
36	Bass Drum 1	**60**	Hi Bongo
37	Side Stick	**61**	Low Bongo
38	Acoustic Snare	**62**	Mute Hi Conga
39	Hand Clap	**63**	Open Hi Conga
40	Electric Snare	**64**	Low Conga

41	Low Floor Tom	**65**	High Timbale
42	Closed Hi-Hat	**66**	Low Timbale
43	High Floor Tom	**67**	High Agogo
44	Pedal Hi-Hat	**68**	Low Agogo
45	Low Tom	**69**	Cabasa
46	Open Hi-Hat	**70**	Maracas
47	Low-Mid Tom	**71**	Short Whistle
48	Hi-Mid Tom	**72**	Long Whistle
49	Crash Cymbal 1	**73**	Short Guiro
50	High Tom	**74**	Long Guiro
51	Ride Cymbal 1	**75**	Claves
52	Chinese Cymbal	**76**	Hi Wood Block
53	Ride Bell	**77**	Low Wood Block
54	Tambourine	**78**	Mute Cuica
55	Splash Cymbal	**79**	Open Cuica
56	Cowbell	**80**	Mute Triangle
57	Crash Cymbal 2	**81**	Open Triangle
58	Vibraslap		

A note-on with note number 42 will trigger a Closed Hi-Hat. This should cut off any Open Hi-Hat or Pedal Hi-Hat sound that may be sustaining. So too, a Pedal Hi-Hat should cut off a sustaining Open Hi-Hat or Closed Hi-Hat. In other words, only one of these three drum sounds can be sounding at any given time.

Similiarly, a Short Whistle should cut off a Long Whistle. A Short Guiro should cut off a Long Guiro. An Mute Triangle should cut off an Open Triangle. A Mute Cuica should cut off an Open Cuica.

Normally, all the above drum sounds have a fixed duration. Regardless of the time between when a Note-On is received and when a matching Note-Off is received, the drum sound always plays for a given duration. For example, assume that a device has a "Crash Cymbal 1" sound that plays for 4 seconds. If a Note-On for note number 49 is received, that cymbal sound starts playing. If a Note-Off for note number 49 is received only 1 second later, that should not cut off the remaining 3 seconds of the sound. The exceptions may be Long Whistle and Long Guiro, which may use the duration between the Note-On and Note-off to determine how "long" the sound plays.

If a drum is still sounding when another one of its Note-Ons is received, typically, another voice "stacks" another instance of that sound playing.

Introduction

Most MIDI sound modules today are "multi-timbral". This means that the module can listen to all 16 MIDI channels at once, and play any 16 of its "patches" simultaneously, with each of the 16 patches set to a different MIDI channel.

It's as if the module had 16 smaller "sub-modules" inside of it. Each sub-module plays its own patch (ie, instrument) upon its own MIDI channel.

MIDI Channels and the Multi-Timbral Module

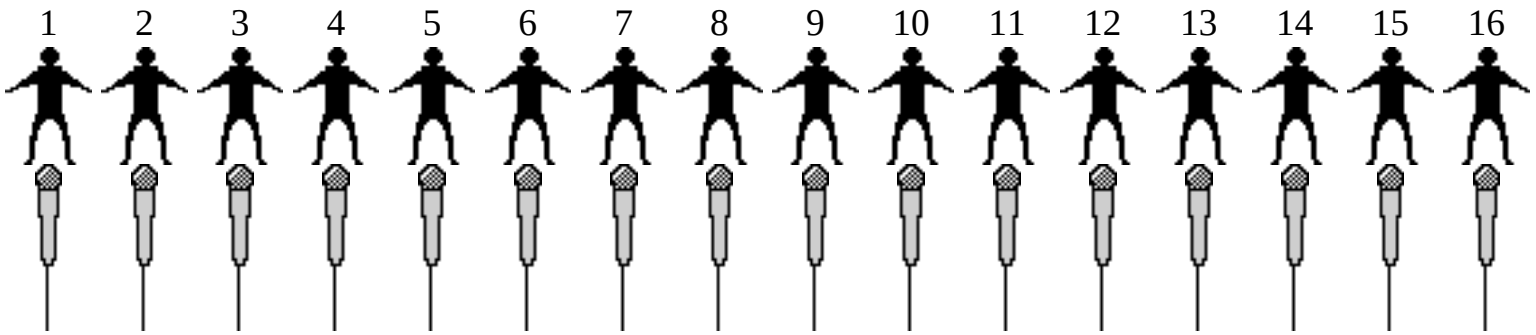
It may help if I draw some analogies here to explain the above.

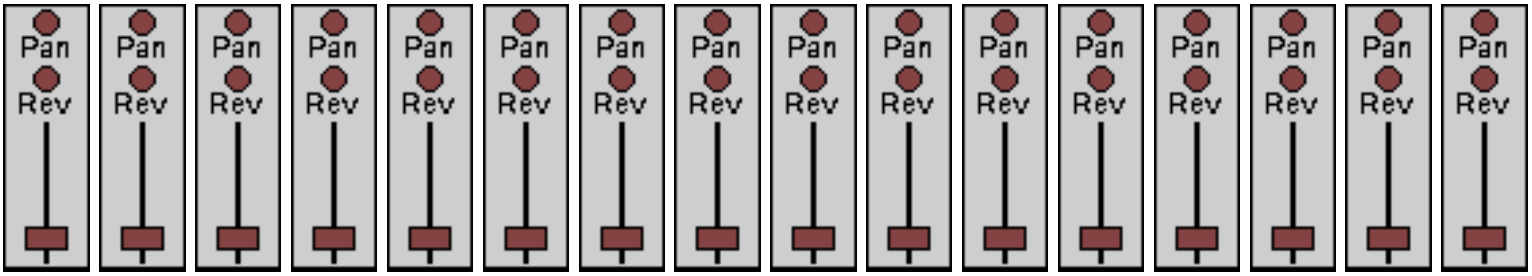
Think of these sub-modules as robotic musicians. I'll call them "robomusicians". You have 16 of them inside one multi-timbral module.

Now think of MIDI channels as channels (ie, inputs) upon a mixing console. You have 16 of them in any one MIDI setup. (I assume one discrete MIDI bus in this "MIDI setup". Some setups have multiple MIDI Ins/Outs with more than 16 MIDI channels. But here, let's talk about a typical MIDI setup which is limited to 16 channels).

Each robomusician (ie, sub-module) has his own microphone plugged into one channel of that 16 channel mixer, so you have individual control over his volume, panning, reverb and chorus levels, and perhaps other settings.

The diagram below illustrates the concept of a multi-timbral module in my MIDI setup. There are 16 "robomusicians" representing the 16 sub-modules. And there are 16 channels on the mixer, representing the 16 MIDI channels. Each robomusician has his own channel on the mixer, which means he has his own volume, pan, reverb level, and other such settings.





But MIDI Channels aren't exactly like inputs on a mixer, because not only are they inputs from those robomusicians, they are also outputs to those robomusicians. In other words, let's say that there is a button on each of those mixer channels. When you push that button and speak into a microphone, your voice is heard through the headphones worn by only the one robomusician plugged into that channel. For example, if you push the button on channel 10, only robomusician 10 hears what you say to him. In this way, you can give individual instructions to each robomusician.

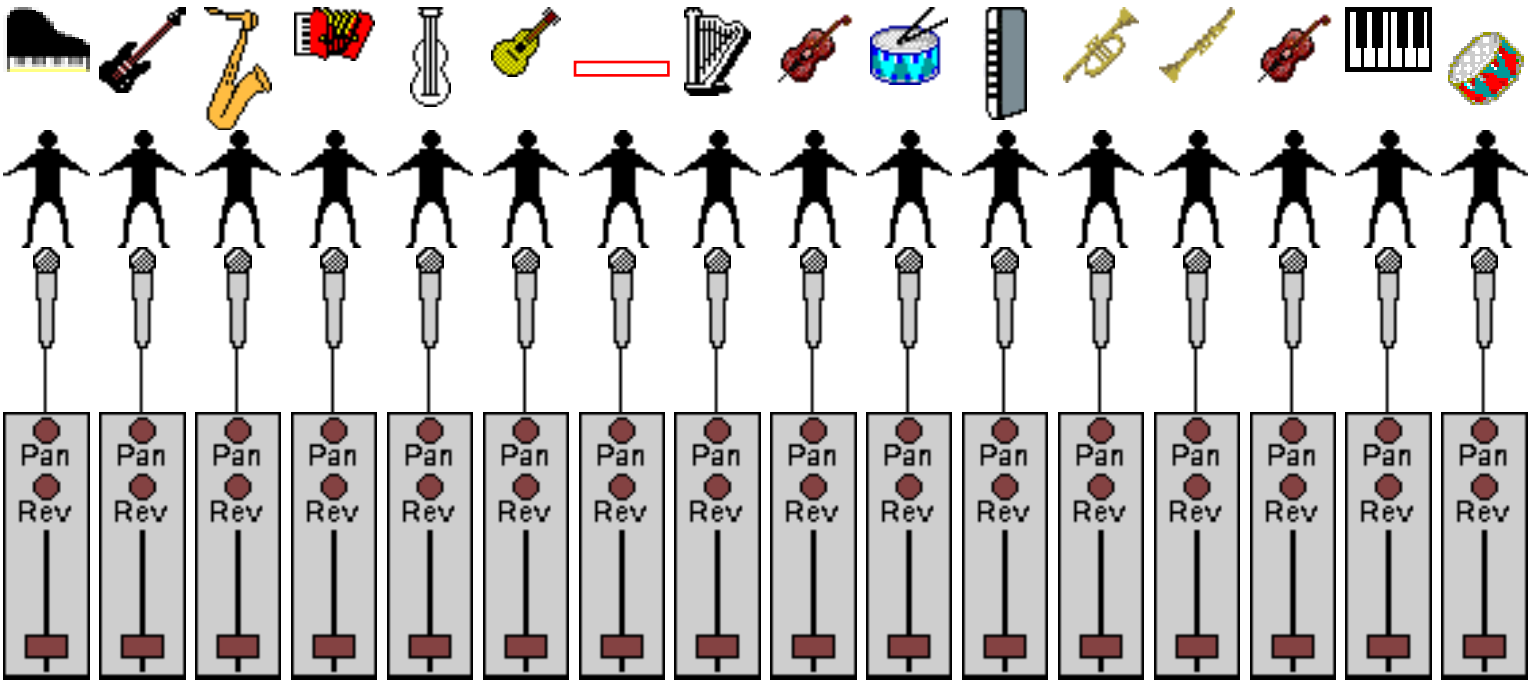
Assigning a patch to a "robomusician"

Think of a patch as a "musical instrument". For example, you typically have Piano, Flute, Saxophone, Bass Guitar, etc, patches in a sound module (even the ones built into a computer sound card -- often referred to as a "wavetable synth"). Typically, most modules have hundreds of patches (ie, musical instruments) to chose from. The patches are numbered. For example, a trumpet patch may be the fifty-seventh patch available among all of the choices.

Since you have 16 robomusicians, you can pick out any 16 instruments (ie, patches) among those hundreds, to be played simultaneously by your 16 robomusicians. Each robomusician can of course play only one instrument at a time. (On the other hand, each robomusician can play chords upon any instrument he plays, even if it's traditionally an instrument that can't play chords. For example, if the robomusician plays a trumpet patch, he can play chords on it, even though a real trumpet is incapable of sounding more than one pitch at a time).

As an example, maybe your arrangement needs a drum kit, a bass guitar, a piano, and a saxophone. Let's say that the drums are played by robomusician #10. (He's on MIDI channel 10 of the mixer). (In fact, with some MIDI modules, channel #10 is reserved for only drums. In other words, robomusician #10 can play only drums, and maybe he's the only robomusician who can play the drums). The other robomusicians are super musicians. Each robomusician can play any of the hundreds of instruments (ie, patches) in your module, but of course, he still is restricted to playing only one instrument at a time. So let's say that you tell robomusician 1 to sit at a piano, and robomusician 2 to pick up a bass guitar, and robomusician 3 to pick up a saxophone. Let's say that you tell the remaining 12 robomusicians to pick up an accordian, violin, acoustic guitar, flute, harp, cello, harmonica, trumpet, clarinet, etc, so that each robomusician has a different instrument to play.

Here's what we want the instrument assignment to look like:



How do you tell the robomusician to pick up a certain instrument? Hit that button upon his channel and give him a message telling him the number of the patch/instrument you want him to play. How do you do that over MIDI? Well, that's what MIDI messages are for. The MIDI Program Change message is the one that instructs a robomusician to pick up a certain instrument. Contained in the MIDI Program Change message is the number of the desired patch/instrument. (For example, above that would be #57 for the Trumpet patch). So, you send (to the multi-timbral module's MIDI In) a MIDI Program Change message upon the MIDI channel for that robomusician. For example, to tell robomusician 3 to pick up a sax, you send a MIDI Program Change (with a value that selects the Saxophone patch) on MIDI channel 3.

Note: To discover what value (ie, number) you need for the Program Change message, in order to select a particular patch, consult the manual for your sound module. If your module follows the General MIDI Patch set, then consult that standard for what numbers select which patches.

Individual control via each MIDI Channel

After you've told the 16 robomusicians what instruments to pick up, you can now have them play a MIDI arrangement with these 16 instruments -- each robomusician playing simultaneously with individual control over his volume, panning, etc.

How do you tell a robomusician what notes to play? You send him MIDI Note messages on his channel. Remember that only that one robomusician "hears" these messages. The other robomusicians see only those messages on their respective channels. (ie, Each robomusician ignores messages that aren't on his channel, and takes notice of only those messages that are on his channel). For example, the sax player is robomusician 3, so you send him note messages on MIDI channel 3.

How do you tell a robomusician to change his volume? You send him Volume Controller messages on

his MIDI channel. How do you tell a robomusician to bend his pitch? You send him Pitch Wheel messages on his MIDI channel. In fact, there are many different things that a robomusician can do independently of the other 15 robomusicians, because there are many different MIDI controller messages that can be sent on any given MIDI channel.

And that's why I say that it's as if there are are 16 "sub-modules" inside of one multi-timbral module -- because these 16 robomusicians really do have independent control over their musical performances, thanks to there being 16 MIDI channels in that one MIDI cable that runs to the multi-timbral module's MIDI In.

Changing instrumentation

OK, let's say that at one point in your arrangement, a 17th instrument needs to be played -- maybe a Banjo. Well, at that point you've got to have one of your 16 robomusicians put down his current instrument and pick up a Banjo instead. Let's say that the sax player isn't supposed to be playing anything at this point in the arrangement. So, you send a MIDI Program Change to robomusician 3 (ie, on MIDI channel 3 -- remember that he's the guy who was playing the sax), telling him to pick up a Banjo. Now when you send him note messages, he'll be playing that banjo. Later on, you can send him another MIDI Program Change to tell him to put down the Banjo and pick up the saxophone again (or some other instrument). So, although you're limited to 16 robomusicians playing 16 instruments simultaneously, any of your robomusicians can change their instruments during the arrangement. (Well, maybe robomusician 10 is limited to playing only drums. Even then, he may be able to choose from among several different drum kits).

Parts

So is there a name for these 16 "robomusicians" or "sub-modules" inside of your MIDI module? Well, different manufacturers refer to them in different ways, and I'm going to use the Roland preference, a **Part**. A Roland multi-timbral module has 16 Parts inside of it, and each usually has its own settings for such things as Volume, Panning, Reverb and Chorus levels, etc, and its MIDI channel (ie, which MIDI data the Part "plays"). Furthermore, each Part has its own way of reacting to MIDI data such as *Channel Pressure* (often used to adjust volume or brightness), *MOD Wheel controller* (often used for a vibrato effect), and *Pitch Wheel* (used to slide the pitch up and down). For example, one Part can cause its patch to sound brighter when it receives Channel Pressure messages that increase in value. On the other hand, another Part could make its volume increase when it receives increasing Channel Pressure messages. These Parts are completely independent of each other. Just because one Part is receiving a Pitch Wheel message and bending its pitch doesn't mean that another Part has to do the same.

Stereo output

You'll note that sound of all 16 robomusicians typically comes out of a stereo output of your sound module (or computer card). That's because most multi-timbral modules have an internal mixer (which can be adjusted by MIDI controller messages to set volume, panning, brightness, reverb level, etc) that mixes the output of all 16 Parts to a pair of stereo output jacks. (ie, The 16 microphones and 16 channel mixing console I alluded to earlier are built into the MIDI module itself. The stereo outputs of the module are like the stereo outputs of that mixing console).

Recommended reading:

What's MIDI?

Electronic Arts is a company that deserves credit for helping make life easier for both programmers and end users. By establishing **Interchange Format Files** (ie, IFF) and releasing the documentation for such, as well as C source code for reading and writing IFF type of files, Electronic Arts has helped make it easier for programmers to develop "backward compatible" and "extensible" file formats. IFF also helps developers write programs that easily read data files created with each others' IFF compliant software, even if there is no business relationship between the developers. In a nutshell, IFF helps minimize problems such as new versions of a particular program having trouble reading data files produced by older versions, or needing a new file format everytime a new version needs to store additional information. It also encourages standardized file formats that aren't tied to a particular product. All of this is good for endusers because it means that their valuable data isn't locked into some proprietary standard that can't be used with a wide variety of hardware and software. Above all else, endusers don't want their work to be held hostage by a single, corporate entity over whom the enduser has no direct control, but that's exactly what happens whenever an enduser saves his data using a program that produces a proprietary, unpublished file format. IFF helps to break this needlessly proprietary stranglehold that developers have exerted upon endusers' works.

An IFF file is a set of data that is in a form that many, unrelated programs can read. An IFF file should not have anything in it that was intended specifically for just one, particular program. If a program must save some "personal" (ie, proprietary) data in an IFF file, it must be saved in a manner which allows another program to "skip over" this data. There are several different types of IFF files. ILBM and GIFF files store picture data. SMUS files store musical scores. WAVE and AIFF files store sampled sounds. Each of these files must start with an ID which indicates that it is indeed an IFF file, followed by an ID that indicates which type of file. So what is an ID? An ID is four, printable ascii characters (ie, 8-bit bytes). If you use a file viewer (capable of displaying each byte as an ascii character) to look at an IFF file, you will notice that every so often you will see 4 "readable" characters in a row. These 4 characters are an ID. Every IFF file must start with one of the following 3 IDs. (I've enclosed each ID in single quotes).

'FORM'

'LIST'

'CAT '

If the first 4 chars (bytes) in a file are not one of these, then it is not an IFF file. These IDs are referred to as **group ID**s in EA literature because each is like a "master ID" after which there may follow more IDs (ie, chunks) that are grouped under that master ID.

Note that the last character in the 'CAT ' ID is a blank space (ie, ascii 32).

After this group ID, there is an UNSIGNED LONG (ie, 32-bit binary value) that indicates how many bytes are in the entire file. This count does not include the 4 byte group ID, nor this ULONG. This ULONG is useful if you wish to load the rest of the file into memory to examine it. After this ULONG, there is an ID that indicates which type of IFF file this is. As mentioned earlier, "ILBM", "WAVE", and "AIFF" are 3 types of IFF files. There are many more, and programmers are always inventing new types for lack of better things to do. Here is the beginning of a typical ILBM file.

'FORM'	OK. This really is an IFF file because it has one of the 3 defined group IDs.
13000	There are 13000 more bytes after this ULONG.
'ILBM'	It is an ILBM (picture) file.

All IFF files start with something similiar to the above, 12 byte "header", except that instead of 'FORM', the group ID can be 'LIST' or 'CAT '. Of course, the ULONG size and file type ID may be different in various files, but nevertheless, a 12 byte header always appears at the beginning of an IFF file. For example, here's an example AIFF header:

'FORM'	OK. This really is an IFF file because it has one of the 3 defined group IDs.
4000	There are 4000 more bytes after this ULONG.
'AIFF'	It is an AIFF (digital audio) file.

What you find after the header depends on which type it is (ie, From here on, an ILBM will be different than an AIFF).

One thing that all IFF files do have in common after the group ID, byte count, and type ID, is that data is organized into chunks. OK, more jargon. What's a chunk? A chunk consists of an ID, a ULONG that tells how many bytes of data are in the chunk, and then all those data bytes. For example, here is a CMAP chunk (which would be found in an ILBM file).

'CMAP'	This is the 4 byte chunk ID.
6	This tells how many data bytes are in the chunk (ie, This is the chunkSize).
0,0,0,1,1,4	Here are the 6 data bytes.

Notice that the chunk size doesn't include the 4 byte ID or the ULONG for the chunk Size.

So, all IFF files are made up of several chunks (ie, groups of data). Each group of data starts with a convenient ID (so that a program can ascertain what kind of data is in the chunk) and a ULONG size (so that a program can ascertain how many bytes of data are in the chunk). There are a few other details to note. A chunk cannot have an odd number of data bytes (such as 3). If necessary, an extra zero byte must be written to make an even number of data bytes. The chunk Size doesn't include this extra byte. So for example, if you want to write 3 bytes in a CMAP chunk, it would look like this:

'CMAP'	
3	Note that chunk Size is 3.
0,1,33,0	Note that there is an extra zero byte.

The reason for this extra "pad byte" for odd-sized chunks has to do with Motorola's 68000 CPU requiring that LONGs be aligned to even memory addresses. IFF files were first used on 68000 based computers, and padding out odd-sized chunks made it easier to load and parse an IFF file on such a

computer (ie, if you load the entire file into a single block of RAM starting upon an even address, all of the chunk IDs and Sizes will conveniently fall upon even memory addresses).

In the preceding example, the group ID was 'FORM'. There are 2 other group IDs as well. A 'CAT ' is a collection of many different FORMs all stuck together consecutively in 1 IFF file. For example, if you had an animation with 6 sound effects, you might save the animation frames in an ANIM FORM, and you might save the sound effects in several AIFF FORMs (one per sound effect). You could save the animation and sound in 7 separate files. The ANIM file would start this way:

FORM	
120000	Whatever the size happens to be (this is expressed in 32 bits).
ANIM	

Each AIFF file would start this way:

FORM	
8000	whatever size.
AIFF	

If the user wanted to copy the data to another disk, he would have to copy 7 files. On the other hand, you could save all the data in one CAT file.

CAT	
4+120008+8008+2028+...	The total size of the ANIM and the 6 AIFF files.
' '	Type of CAT. 4 spaces for the type ID means "a grab bag" of IFF FORMs are going to be inside of this CAT. If it just so happened that all of the enclosed FORMs were 1 type, such as ILBM, then this type ID would be 'ILBM'.

FORM	
120000	
ANIM	
	...all the chunks in the ANIM file placed here. (Note: ANIMs have imbedded ILBM FORMs. The guy who devised the ANIM type of IFF file broke the rules by mistake, and nobody caught his error until it was too late).

FORM	
8000	
AIFF	

	...all the chunks in the first sound effect here.

FORM	
2020	
AIFF	
	...all the chunks in the second sound effect here.

	...etc. for the other 4 sound effects.

To further complicate matters, there are LISTs. LISTs are a lot like CATs except that there is an optional, additional group ID associated with LISTs. That ID is a PROP. LISTs can have imbedded PROPS just like an ILBM can have an imbedded CMAP chunk. A PROP header looks very much like a FORM header in that you must follow it with a type ID. For example, here is an ILBM PROP with a CMAP in it.

PROP	Here's a PROP.
4+14	Here's how many bytes follow in the PROP.
ILBM	It's an ILBM PROP.
'CMAP'	Here's a CMAP chunk inside of this ILBM PROP.
6	There are 6 bytes following in this CMAP chunk.
0,0,0,1,1,4	

LISTs are meant to encompass similiar FORMs (i.e. several AIFF files stuck together). Often, when you have similiar FORMs stuck together, some of the chunks in the individual FORMs are the same. For example, assume that we have 2 AIFF sound effects. AIFF FORMs can have a NAME chunk which contains the ascii string that is the name of the sound effect. Also assume that both sounds are called "car crash". With a CAT, we end up having 2 identical NAME chunks in each AIFF FORM like so:

CAT	We put the 2 files into 1 CAT.
4+1008+508	
AIFF	It's a CAT of several AIFF FORMs.

FORM	Here's the start of the first sound effect file.
1000	
AIFF	
	...other chunks may be inserted here.
NAME	Here's the name chunk for the 1st sound effect.
9	

'car crash',0

...other chunks may be inserted here.

FORM Here's the start of the 2nd sound effect file.
500
AIFF

...other chunks may be inserted here.

NAME Here's the name chunk for the 2nd sound effect. Look familiar?
9
'car crash',0

...other chunks may be inserted here.

With a LIST, we can have PROPs. A PROP is group ID that allows us to place chunks that pertain to all the FORMs in the LIST. So, we can rip out the NAME chunks inside both AIFF FORMs and replace it with one NAME chunk inside of a PROP.

LIST Notice that we use a LIST instead of a CAT.
4+30+990+490+...
AIFF

PROP Here's where we put chunks intended for ALL the subsequent FORMS; inside a PROP.
22
AIFF Type of PROP.
NAME Here's the name chunk inside of the PROP.
9
'car crash',0

FORM Here's the start of the first sound effect file.
982 Size is 18 bytes less because no NAME chunk here.
AIFF

...other chunks may be inserted here, but no NAME chunk needed.

FORM Here's the start of the 2nd sound effect file.
482
AIFF

...other chunks may be inserted here, but no NAME needed for this guy either.

Notice that the PROP group ID is followed by a type ID (in this case AIFF). This means that the

PROP only affects any AIFF FORMs. If you were to sneak in an SMUS FORM at the end, the NAME chunk would not apply to it. Also, if you included a NAME chunk in one of the AIFF FORMs, it would override the PROP. For example, assume that you have a LIST containing 10 AIFF FORMs. All but 1 of them is named "Snare Hit". You can store a NAME chunk in a PROP AIFF for "Snare Hit". Then, in the one AIFF FORM whose name is not "Snare Hit", you can include a NAME chunk to override the NAME chunk in the PROP.

It should be noted that you can take several LISTs and squash them together inside of a CAT or another LIST. Personally, I have never seen a data file with this level of nesting, and doubt that it would be of much use.

In the above examples, psuedo code was used to represent the headers. Let's look at how a hex file viewer might display the actual contents of an IFF file (in hex bytes). First, an IFF header for a FORM AIFF, psuedo code.

FORM
4096
AIFF

Now here's a view of the actual data file.

46 4F 52 4D	FORM
00 00 10 00	hex 00001000, or 4096 decimal
41 49 46 46	AIFF

Note that the ULONG byte count is stored in Big Endian order (ie, the Most Significant Byte is first, and the Least Significant Byte is last). This is how the Motorola 680x0 stores long values in memory (ie, the opposite order of Intel 80x86). IFF files use Big Endian order for all 16-bit (ie, SHORT) and 32-bit (ie, LONG) values.

Microsoft decided that IFF was a good idea, but since Windows is traditionally tethered to Intel CPUs, a version of IFF was needed which stored LONG or SHORT values in Little Endian order. So, MS decided to create some new group IDs. MS took the FORM ID and created a Little Endian version of it known as RIFF. For example, the WAVE file format has a RIFF group ID. All of the SHORT and LONG values in the file are stored in Little Endian order. Let's take a look at an example header for a WAVE file. Assume that there are 258 bytes of data after the byte count.

52 49 46 46	RIFF
02 01 00 00	hex 00000102, or 258 decimal
57 41 56 45	WAVE

Note that the ULONG byte count is stored in Little Endian order (ie, the Least Significant Byte is

first, and the Most Significant Byte is last). Good old backwards-thinking Intel.

Now, there's some real justification for creating a RIFF group ID, if you're working with an Intel CPU. But Microsoft couldn't stop there. True to their "not made here, so if we're going to accept it, we have to inflict our brutish, unneeded brand upon it" mentality, Microsoft created another group ID called RIFX. What's an RIFX file? It's simply a FORM with RIFX replacing the FORM ID. So, if you want to turn a FORM AIFF into a RIFX AIFF, you just change the first 4 bytes to RIFX. Needless to say, nobody has ever used the RIFX group ID, and it will undoubtably suffer a justifiably ignoble disappearance.

Just like everyone else, programmers make mistakes. As mentioned before, the Amiga's ANIM file format was a mistake. It puts FORM headers inside of a FORM group ID. That's not supposed to happen. You can put FORM headers inside of a CAT or LIST, but not another FORM. A mistake was also made with the MIDI file format. The programmer who devised it didn't put a proper IFF header on the file. It should be:

FORM	group ID. Indicates an IFF file that contains one type of data.
3000	whatever size the file happens to be.
MIDI	type of data. What follows will be chunks as defined by the MIDI type of IFF file.

But the programmer omitted the FORM group ID, and simply put the MThd chunk first. So, a MIDI file starts as so:

MThd	Chunk ID.
6	size of MThd chunk.

Another deviation from the standard occurs with padding out odd-sized chunks with an extra byte. Some programmers didn't bother doing this when devising new IFF type files, and occasionally, one will come across some specification for a new IFF type that allows odd-sized chunks.

Unfortunately, these programmers released their work based upon these aberrations before getting that work reviewed by other programmers who might have offered good reasons why the aberrations should be corrected. It makes it that much harder for software to read and write files if it has to deal with aberrations of the IFF standard. There's no reason for that, particularly when a strict adherence to the standard sacrifices almost nothing in the way of quality and efficiency over an aberration. But try to tell that to a paranoid programmer who thinks that if he shows anyone what he's doing before his product is shrink-wrapped, someone will steal his soul... well, IFF does give the computer industry a means for resolving needless hassles with data file formats, and it has worked very successfully in a number of instances, although occasionally people don't always use the standard wisely, or don't quite grasp EA's altruistic notion that there is no good reason why a file format should ever be proprietary or unpublished. (I urge consumers to avoid products where that is the case).

WAVE File Format is a file format for storing digital audio (waveform) data. It supports a variety of bit resolutions, sample rates, and channels of audio. This format is very popular upon IBM PC (clone) platforms, and is widely used in professional programs that process digital audio waveforms. It takes into account some peculiarities of the Intel CPU such as little endian byte order.

This format uses Microsoft's version of the Electronic Arts Interchange File Format method for storing data in "chunks". You should read the article About Interchange File Format before proceeding.

Data Types

A C-like language will be used to describe the data structures in the file. A few extra data types that are not part of standard C, but which will be used in this document, are:

pstring Pascal-style string, a one-byte count followed by that many text bytes. The total number of bytes in this data type should be even. A pad byte can be added to the end of the text to accomplish this. This pad byte is not reflected in the count.

ID A chunk ID (ie, 4 ASCII bytes) as described in About Interchange File Format.

Also note that when you see an array with no size specification (e.g., char ckData[];), this indicates a variable-sized array in our C-like language. This differs from standard C arrays.

Constants

Decimal values are referred to as a string of digits, for example 123, 0, 100 are all decimal numbers. Hexadecimal values are preceded by a 0x - e.g., 0x0A, 0x1, 0x64.

Data Organization

All data is stored in 8-bit bytes, arranged in Intel 80x86 (ie, little endian) format. The bytes of multiple-byte values are stored with the low-order (ie, least significant) bytes first. Data bits are as follows (ie, shown with bit numbers on top, "lsb" stands for "least significant byte" and "msb" stands for "most significant byte"):

```
         7  6  5  4  3  2  1  0
       +-----------------------+
 char: | lsb                   |
       +-----------------------+

         7  6  5  4  3  2  1  0 15 14 13 12 11 10  9  8
       +-----------------------+-----------------------+
short: | lsb     byte 0        |       byte 1      msb |
       +-----------------------+-----------------------+

         7  6  5  4  3  2  1  0 15 14 13 12 11 10  9  8 23 22 21 20 19 18 17 16 31 30
29 28 27 26 25 24
       +-----------------------+-----------------------+-----------------------+------
-----------------+
 long: | lsb     byte 0        |       byte 1          |         byte 2        |
byte 3      msb |
       +-----------------------+-----------------------+-----------------------+------
-----------------+
```

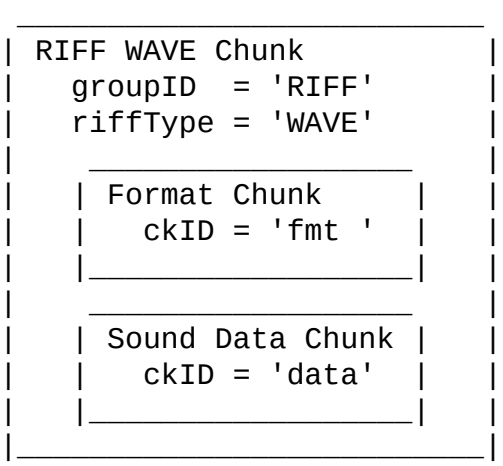
File Structure

A WAVE file is a collection of a number of different types of chunks. There is a required Format ("fmt ") chunk which contains important parameters describing the waveform, such as its sample rate. The Data chunk, which contains the actual waveform data, is also required. All other chunks are optional. Among the other optional chunks are ones which define cue points, list instrument parameters, store application-specific information, etc. All of these chunks are described in detail in the following sections of this document.

All applications that use WAVE must be able to read the 2 required chunks and can choose to selectively ignore the optional chunks. A program that copies a WAVE should copy all of the chunks in the WAVE, even those it chooses not to interpret.

There are no restrictions upon the order of the chunks within a WAVE file, with the exception that the Format chunk must precede the Data chunk. Some inflexibly written programs expect the Format chunk as the first chunk (after the RIFF header) although they shouldn't because the specification doesn't require this.

Here is a graphical overview of an example, minimal WAVE file. It consists of a single WAVE containing the 2 required chunks, a Format and a Data Chunk.



A Bastardized Standard

The WAVE format is sort of a bastardized standard that was concocted by too many "cooks" who didn't properly coordinate the addition of "ingredients" to the "soup". Unlike with the AIFF standard which was mostly designed by a small, coordinated group, the WAVE format has had all manner of much-too-independent, uncoordinated aberrations inflicted upon it. The net result is that there are far too many chunks that may be found in a WAVE file -- many of them duplicating the same information found in other chunks (but in an unnecessarily different way) simply because there have been too many programmers who took too many liberties with unilaterally adding their own additions to the WAVE format without properly coming to a concensus of what everyone else needed (and therefore it encouraged an "every man for himself" attitude toward adding things to this "standard"). One example is the Instrument chunk versus the Sampler chunk. Another example is the Note versus Label chunks in an Associated Data List. I don't even want to get into the totally irresponsible proliferation of compressed formats. (ie, It seems like everyone and his pet Dachshound has come up with some compressed version of storing wave data -- like we need 100 different ways to do that). Furthermore, there are lots of inconsistencies, for example how 8-bit data is unsigned, but 16-bit data is signed.

I've attempted to document only those aspects that you're very likely to encounter in a WAVE file. I suggest that you concentrate upon these and refuse to support the work of programmers who feel the need to deviate from a standard with inconsistent, proprietary, self-serving, unnecessary extensions. Please do your part to rein in half-assed programming.

Sample Points and Sample Frames

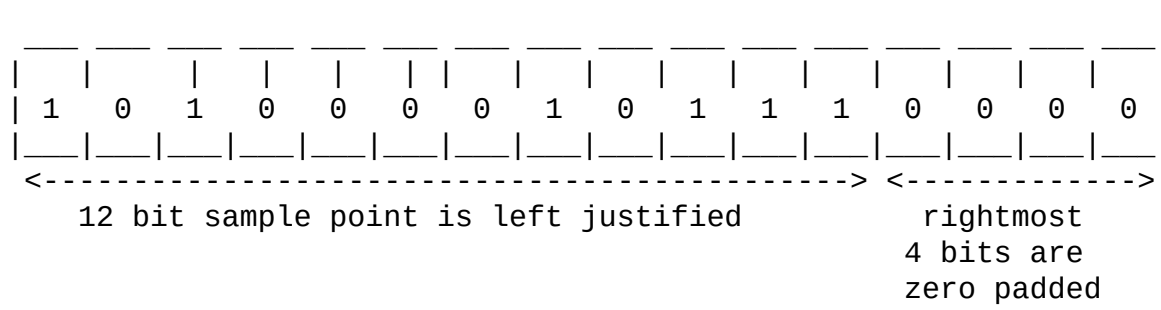
A large part of interpreting WAVE files revolves around the two concepts of sample points and sample frames.

A sample point is a value representing a sample of a sound at a given moment in time. For waveforms with greater than 8-bit resolution, each sample point is stored as a linear, 2's-complement value which may be from 9 to 32 bits wide (as determined by

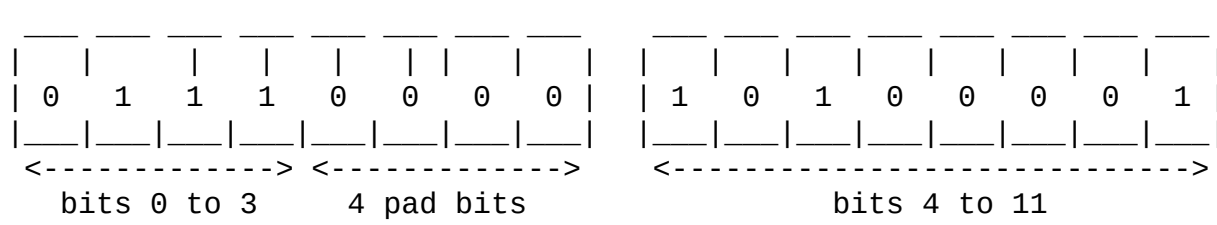
the wBitsPerSample field in the Format Chunk, assuming PCM format -- an uncompressed format). For example, each sample point of a 16-bit waveform would be a 16-bit word (ie, two 8-bit bytes) where 32767 (0x7FFF) is the highest value and -32768 (0x8000) is the lowest value. For 8-bit (or less) waveforms, each sample point is a linear, unsigned byte where 255 is the highest value and 0 is the lowest value. Obviously, this signed/unsigned sample point discrepancy between 8-bit and larger resolution waveforms was one of those "oops" scenarios where some Microsoft employee decided to change the sign sometime after 8-bit wave files were common but 16-bit wave files hadn't yet appeared.

Because most CPU's read and write operations deal with 8-bit bytes, it was decided that a sample point should be rounded up to a size which is a multiple of 8 when stored in a WAVE. This makes the WAVE easier to read into memory. If your ADC produces a sample point from 1 to 8 bits wide, a sample point should be stored in a WAVE as an 8-bit byte (ie, unsigned char). If your ADC produces a sample point from 9 to 16 bits wide, a sample point should be stored in a WAVE as a 16-bit word (ie, signed short). If your ADC produces a sample point from 17 to 24 bits wide, a sample point should be stored in a WAVE as three bytes. If your ADC produces a sample point from 25 to 32 bits wide, a sample point should be stored in a WAVE as a 32-bit doubleword (ie, signed long). Etc.

Furthermore, the data bits should be left-justified, with any remaining (ie, pad) bits zeroed. For example, consider the case of a 12-bit sample point. It has 12 bits, so the sample point must be saved as a 16-bit word. Those 12 bits should be left-justified so that they become bits 4 to 15 inclusive, and bits 0 to 3 should be set to zero. Shown below is how a 12-bit sample point with a value of binary 101000010111 is formatted left-justified as a 16-bit word.

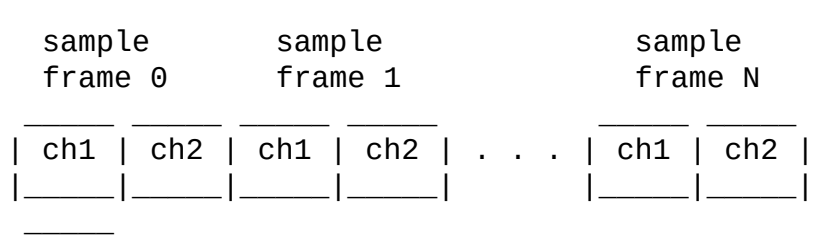


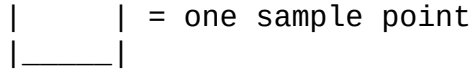
But note that, because the WAVE format uses Intel little endian byte order, the LSB is stored first in the wave file as so:



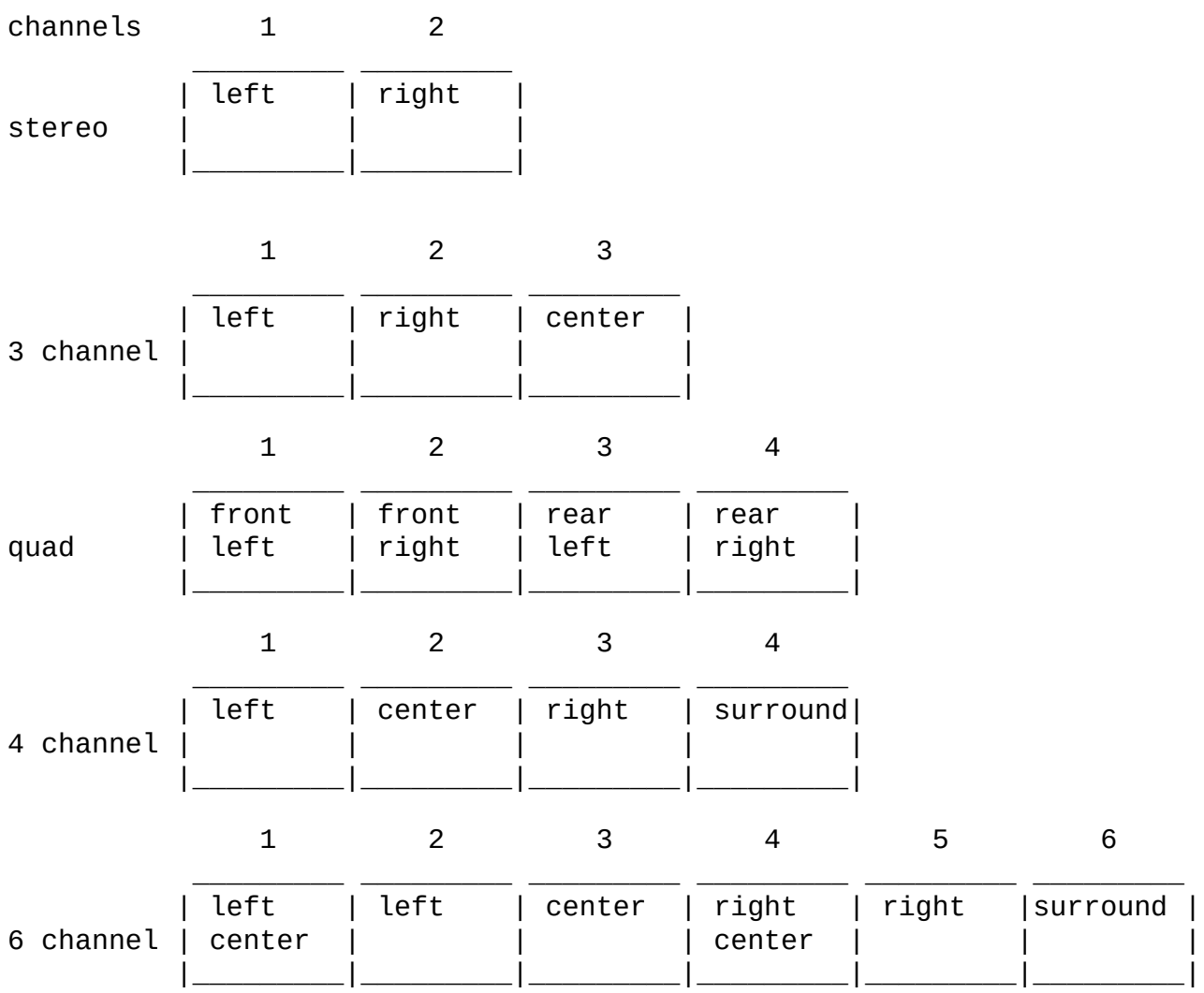
For multichannel sounds (for example, a stereo waveform), single sample points from each channel are interleaved. For example, assume a stereo (ie, 2 channel) waveform. Instead of storing all of the sample points for the left channel first, and then storing all of the sample points for the right channel next, you "mix" the two channels' sample points together. You would store the first sample point of the left channel. Next, you would store the first sample point of the right channel. Next, you would store the second sample point of the left channel. Next, you would store the second sample point of the right channel, and so on, alternating between storing the next sample point of each channel. This is what is meant by interleaved data; you store the next sample point of each of the channels in turn, so that the sample points that are meant to be "played" (ie, sent to a DAC) simultaneously are stored contiguously.

The sample points that are meant to be "played" (ie, sent to a DAC) simultaneously are collectively called a **sample frame**. In the example of our stereo waveform, every two sample points makes up another sample frame. This is illustrated below for that stereo example.





For a monophonic waveform, a sample frame is merely a single sample point (ie, there's nothing to interleave). For multichannel waveforms, you should follow the conventions shown below for which order to store channels within the sample frame. (ie, Below, a single sample frame is displayed for each example of a multichannel waveform).



The sample points within a sample frame are packed together; there are no unused bytes between them. Likewise, the sample frames are packed together with no pad bytes.

Note that the above discussion outlines the format of data within an uncompressed data chunk. There are some techniques of storing compressed data in a data chunk. Obviously, that data would need to be uncompressed, and then it will adhere to the above layout.

Format chunk

The Format (fmt) chunk describes fundamental parameters of the waveform data such as sample rate, bit resolution, and how many channels of digital audio are stored in the WAVE.

```
#define FormatID 'fmt '   /* chunkID for Format Chunk. NOTE: There is a space at the end of this ID. */

typedef struct {
  ID              chunkID;
  long            chunkSize;
```

```
  short          wFormatTag;
  unsigned short wChannels;
  unsigned long  dwSamplesPerSec;
  unsigned long  dwAvgBytesPerSec;
  unsigned short wBlockAlign;
  unsigned short wBitsPerSample;
/* Note: there may be additional fields here, depending upon wFormatTag. */
} FormatChunk;
```

The ID is always "**fmt** ". The chunkSize field is the number of bytes in the chunk. This does not include the 8 bytes used by ID and Size fields. For the Format Chunk, chunkSize may vary according to what "format" of WAVE file is specified (ie, depends upon the value of wFormatTag).

WAVE data may be stored without compression, in which case the sample points are stored as described in **Sample Points and Sample Frames**. Alternately, different forms of compression may be used when storing the sound data in the Data chunk. With compression, each sample point may take a differing number of bytes to store. The wFormatTag indicates whether compression is used when storing the data.

If compression is used (ie, WFormatTag is some value other than 1), then there will be additional fields appended to the Format chunk which give needed information for a program wishing to retrieve and decompress that stored data. The first such additional field will be an unsigned short that indicates how many more bytes have been appended (after this unsigned short). Furthermore, compressed formats must have a Fact chunk which contains an unsigned long indicating the size (in sample points) of the waveform after it has been decompressed. There are (too) many compressed formats. Details about them can be gotten from Microsoft's web site.

If no compression is used (ie, wFormatTag = 1), then there are no further fields.

The wChannels field contains the number of audio channels for the sound. A value of 1 means monophonic sound, 2 means stereo, 4 means four channel sound, etc. Any number of audio channels may be represented. For multichannel sounds, single sample points from each channel are interleaved. A set of interleaved sample points is called a sample frame.

The actual waveform data is stored in another chunk, the Data Chunk, which will be described later.

The dwSamplesPerSec field is the sample rate at which the sound is to be played back in sample frames per second (ie, Hertz). The 3 standard MPC rates are 11025, 22050, and 44100 KHz, although other rates may be used.

The dwAvgBytesPerSec field indicates how many bytes play every second. dwAvgBytesPerSec may be used by an application to estimate what size RAM buffer is needed to properly playback the WAVE without latency problems. Its value should be equal to the following formula rounded up to the next whole number:

dwSamplesPerSec * wBlockAlign

The wBlockAlign field should be equal to the following formula, rounded to the next whole number:

wChannels * (wBitsPerSample / 8)

Essentially, wBlockAlign is the size of a sample frame, in terms of bytes. (eg, A sample frame for a 16-bit mono wave is 2 bytes. A sample frame for a 16-bit stereo wave is 4 bytes. Etc).

The wBitsPerSample field indicates the bit resolution of a sample point (ie, a 16-bit waveform would have wBitsPerSample = 16).

One, and only one, Format Chunk is required in every WAVE.

Data chunk

The Data (data) chunk contains the actual sample frames (ie, all channels of waveform data).

```
#define DataID 'data'  /* chunk ID for data Chunk */

typedef struct {
  ID             chunkID;
  long           chunkSize;

  unsigned char  waveformData[];
} DataChunk;
```

The ID is always **data**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size (ie, chunkSize is the number of remaining bytes in the chunk after the chunkSize field, not counting any trailing pad byte).

Remember that the bit resolution, and other information is gotten from the Format chunk.

The following discussion assumes uncompressed data.

The waveformData array contains the actual waveform data. The data is arranged into what are called *sample frames*. For more information on the arrangment of data, see "Sample Points and Sample Frames".

You can determine how many bytes of actual waveform data there is from the Data chunk's chunkSize field. The number of sample frames in waveformData is determined by dividing this chunkSize by the Format chunk's wBlockAlign.

The Data Chunk is required. One, and only one, Data Chunk may appear in a WAVE.

Another way of storing waveform data

So, you're thinking "This WAVE format isn't that bad. It seems to make sense and there aren't all that many inconsistencies, duplications, and inefficiencies". You fool! We're just getting started with our first excursion into unnecessary inconsistencies, duplications, and inefficiency.

Sure, countless brain-damaged programmers have inflicted literally dozens of compressed data formats upon the Data chunk, but apparently someone felt that even this wasn't enough to make your life difficult in trying to support WAVE files. No, some half-wit decided that it would be a good idea to screw around with storing waveform data in something other than one Data chunk. NOOOOOOOOOOOOO!!!!!!

For some god-forsaken reason, someone came up with the idea of using an imbedded IFF List inside of the WAVE file. NOOOOOOOOOOOOOOOO!!!!!!!! And this "Wave List" would contain multiple 'data' and 'slnt' chunks. NOOOOOOOOOOOOOOO!!!! The Type ID for this List is 'wavl'.

I strongly suggest that you refuse to support any WAVE file that exhibits this Wave List nonsense. There's no need for it, and hopefully, the misguided programmer who conjured it up will be embarrassed into hanging his head in shame when nobody agrees to support his foolishness. Just say "NOOOOOOOOOOOOO!!!!"

Cue chunk

The Cue chunk contains one or more "cue points" or "markers". Each cue point references a specific offset within the

waveformData array, and has its own CuePoint structure within this chunk.

In conjunction with the Playlist chunk, the Cue chunk can be used to store looping information.

CuePoint Structure

```
typedef struct {
  long    dwIdentifier;
  long    dwPosition;
  ID      fccChunk;
  long    dwChunkStart;
  long    dwBlockStart;
  long    dwSampleOffset;
} CuePoint;
```

The dwIdentifier field contains a unique number (ie, different than the ID number of any other CuePoint structure). This is used to associate a CuePoint structure with other structures used in other chunks which will be described later.

The dwPosition field specifies the position of the cue point within the "play order" (as determined by the Playlist chunk. See that chunk for a discussion of the play order).

The fccChunk field specifies the chunk ID of the Data or Wave List chunk which actually contains the waveform data to which this CuePoint refers. If there is only one Data chunk in the file, then this field is set to the ID 'data'. On the other hand, if the file contains a Wave List (which can contain both 'data' and 'slnt' chunks), then fccChunk will specify 'data' or 'slnt' depending upon in which type of chunk the referenced waveform data is found.

The dwChunkStart and dwBlockStart fields are set to 0 for an uncompressed WAVE file that contains one 'data' chunk. These fields are used only for WAVE files that contain a Wave List (with multiple 'data' and 'slnt' chunks), or for a compressed file containing a 'data' chunk. (Actually, in the latter case, dwChunkStart is also set to 0, and only dwBlockStart is used). Again, I want to emphasize that you can avoid all of this unnecessary crap if you avoid hassling with compressed files, or Wave Lists, and instead stick to the sensible basics.

The dwChunkStart field specifies the byte offset of the start of the 'data' or 'slnt' chunk which actually contains the waveform data to which this CuePoint refers. This offset is relative to the start of the first chunk within the Wave List. (ie, It's the byte offset, within the Wave List, of where the 'data' or 'slnt' chunk of interest appears. The first chunk within the List would be at an offset of 0).

The dwBlockStart field specifies the byte offset of the start of the block containing the position. This offset is relative to the start of the waveform data within the 'data' or 'slnt' chunk.

The dwSampleOffset field specifies the sample offset of the cue point relative to the start of the block. In an uncompressed file, this equates to simply being the offset within the waveformData array. Unfortunately, the WAVE documentation is much too ambiguous, and doesn't define what it means by the term "sample offset". This could mean a byte offset, or it could mean counting the sample points (for example, in a 16-bit wave, every 2 bytes would be 1 sample point), or it could even mean sample frames (as the loop offsets in AIFF are specified). Who knows? The guy who conjured up the Cue chunk certainly isn't saying. I'm assuming that it's a byte offset, like the above 2 fields.

Cue Chunk

```
#define CueID 'cue '  /* chunk ID for Cue Chunk */

typedef struct {
  ID        chunkID;
  long      chunkSize;

  long      dwCuePoints;
  CuePoint  points[];
```

```
} CueChunk;
```

The ID is always **cue** . chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields.

The dwCuePoints field is the number of CuePoint structures in the Cue Chunk. If dwCuePoints is not 0, it is followed by that many CuePoint structures, one after the other. Because all fields in a CuePoint structure are an even number of bytes, the length of any CuePoint will always be even. Thus, CuePoints are packed together with no unused bytes between them. The CuePoints need not be placed in any particular order.

The Cue chunk is optional. No more than one Cue chunk can appear in a WAVE.

Playlist chunk

The Playlist (plst) chunk specifies a play order for a series of cue points. The Cue chunk contains all of the cue points, but the Playlist chunk determines how those cue points are used when playing back the waveform (ie, which cue points represent looped sections, and in what order those loops are "played"). The Playlist chunk contains one or more Segment structures, each of which identifies a looped section of the waveform (in conjunction with the CuePoint structure with which it is associated).

Segment Structure

```
typedef struct {
  long    dwIdentifier;
  long    dwLength;
  long    dwRepeats;
} Segment;
```

The dwIdentifier field contains a unique number (ie, different than the ID number of any other Segment structure). This field should correspond with the dwIndentifier field of some CuePoint stored in the Cue chunk. In other words, this Segment structure contains the looping information associated with that CuePoint structure with the same ID number.

The dwLength field specifies the length of the section in samples (ie, the length of the looped section). Note that the start position of the loop would be the dwSampleOffset of the referenced CuePoint structure in the Cue chunk. (Or, you may need to hassle with the dwChunkStart and dwBlockStart fields as well if dealing with a Wave List or compressed data).

The dwRepeats field specifies the number of times to play the loop. I assume that a value of 1 means to repeat this loop once only, but the WAVE documentation is very incomplete and omits this important information. I have no idea how you would specify an infinitely repeating loop. Certainly, the person who conjured up the Playlist chunk appears to have no idea whatsoever. Due to the ambiguities, inconsistencies, inefficiencies, and omissions of the Cue and Playlist chunks, I very much recommend that you use the Sampler chunk (described later) to replace them.

Playlist chunk

```
#define PlaylistID 'plst'  /* chunk ID for Playlist Chunk */

typedef struct {
  ID        chunkID;
  long      chunkSize;

  long      dwSegments;
  Segment   Segments[];
} PlaylistChunk;
```

The ID is always **plst**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields.

The dwSegments field is the number of Segment structures in the Playlist Chunk. If dwSegments is not 0, it is followed by that many Segment structures, one after the other. Because all fields in a Segment structure are an even number of bytes, the length of any Segment will always be even. Thus, Segments are packed together with no unused bytes between them. The Segments need not be placed in any particular order.

Associated Data List

The Associated Data List contains text "labels" or "names" that are associated with the CuePoint structures in the Cue chunk. In other words, this list contains the text labels for those CuePoints.

Again, we're talking about another imbedded IFF List within the WAVE file. NOOOOOOOOOOOOO!!!! What's a List? A List is simply a "master chunk" that contains several "sub-chunks". Just like with any other chunk, the "master chunk" has an ID and chunkSize, but inside of this chunk are sub-chunks, each with its own ID and chunkSize. Of course, the chunkSize for the master chunk (ie, List) includes the size of all of these sub-chunks (including their ID and chunkSize fields).

The "Type ID" for the Associated Data List is "adtl". Remember that an IFF list header has 3 fields:

```
typedef struct {
  ID      listID;      /* 'list' */
  long    chunkSize;   /* includes the Type ID below */
  ID      typeID;     /* 'adtl' */
} ListHeader;
```

There are several sub-chunks that may be found inside of the Associated Data List. The ones that are important to WAVE format have IDs of "labl", "note", or "ltxt". Ignore the rest. Here are those 3 sub-chunks and their fields:

The Associated Data List is optional. The WAVE documentation doesn't specify if more than one can be contained in a WAVE file.

Label Chunk

```
#define LabelID 'labl'  /* chunk ID for Label Chunk */

typedef struct {
  ID      chunkID;
  long    chunkSize;

  long    dwIdentifier;
  char    dwText[];
} LabelChunk;
```

The ID is always **labl**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size (ie, chunkSize is the number of remaining bytes in the chunk after the chunkSize field, not counting any trailing pad byte).

The dwIdentifier field contains a unique number (ie, different than the ID number of any other Label chunk). This field should correspond with the dwIndentifier field of some CuePoint stored in the Cue chunk. In other words, this Label chunk contains the text label associated with that CuePoint structure with the same ID number.

The dwText array contains the text label. It should be a null-terminated string. (The null byte is included in the chunkSize, therefore the length of the string, including the null byte, is chunkSize - 4).

Note Chunk

```
#define NoteID 'note'  /* chunk ID for Note Chunk */

typedef struct {
  ID      chunkID;
  long    chunkSize;

  long    dwIdentifier;
  char    dwText[];
} NoteChunk;
```

The Note chunk, whose ID is **note**, is otherwise exactly the same as the Label chunk (ie, same fields). See what I mean about pointless duplication? But, in theory, a Note chunk contains a "comment" about a CuePoint, whereas the Label chunk is supposed to contain the actual CuePoint label. So, it's possible that you'll find both a Note and Label for a specific CuePoint, each containing different text.

Labeled Text Chunk

```
#define LabelTextID 'ltxt'  /* chunk ID for Labeled Text Chunk */

typedef struct {
  ID      chunkID;
  long    chunkSize;

  long    dwIdentifier;
  long    dwSampleLength;
  long    dwPurpose;
  short   wCountry;
  short   wLanguage;
  short   wDialect;
  short   wCodePage;
  char    dwText[];
} LabelTextChunk;
```

The ID is always **ltxt**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size (ie, chunkSize is the number of remaining bytes in the chunk after the chunkSize field, not counting any trailing pad byte).

The dwIdentifier field is the same as the Label chunk.

The dwSampleLength field specifies the number of sample points in the segment of waveform data. In other words, a Labeled Text chunk contains a label for a **section** of the waveform data, not just a specific point, for example the looped section of a waveform.

The dwPurpose field specifies the type or purpose of the text. For example, dwPurpose can contain an ID like "scrp" for script text or "capt" for close-caption text. How is this related to waveform data? Well, it isn't really. It's just that Associated Data Lists are used in other file formats, so they contain generic fields that sometimes don't have much relevance to waveform data.

The wCountry, wLanguage, and wCodePage fields specify the country code, language/dialect, and code page for the text. An application typically queries these values from the operating system.

Sampler Chunk

The Sampler (smpl) Chunk defines basic parameters that an instrument, such as a MIDI sampler, could use to play the waveform data. Most importantly, it includes information about looping the waveform (ie, during playback, to "sustain" the waveform). Of course, as you've come to expect from the WAVE file format, it duplicates some of the information that can be

found in the Cue and Playlist chunks, but fortunately, in a more sensible, consistent, better-documented way.

```
#define SamplerID 'smpl'  /* chunk ID for Sampler Chunk */

typedef struct {
  ID              chunkID;
  long            chunkSize;

  long            dwManufacturer;
  long            dwProduct;
  long            dwSamplePeriod;
  long            dwMIDIUnityNote;
  long            dwMIDIPitchFraction;
  long            dwSMPTEFormat;
  long            dwSMPTEOffset;
  long            cSampleLoops;
  long            cbSamplerData;
  struct SampleLoop Loops[];
} SamplerChunk;
```

The ID is always **smpl**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size (ie, chunkSize is the number of remaining bytes in the chunk after the chunkSize field, not counting any trailing pad byte).

The dwManufacturer field contains the MMA Manufacturer code for the intended sampler. Each manufacturer of MIDI products has his own ID assigned to him by the MIDI Manufacturer's Association. See the MIDI Specification (under System Exclusive) for a listing of current Manufacturer IDs. The high byte of dwManufacturer indicates the number of low order bytes (1 or 3) that are valid for the manufacturer code. For example, this value will be 0x01000013 for Digidesign (the MMA Manufacturer code is one byte, 0x13); whereas 0x03000041 identifies Microsoft (the MMA Manufacturer code is three bytes, 0x00 0x00 0x41). If the WAVE is not intended for a specific manufacturer, then this field should be set to 0.

The dwProduct field contains the Product code (ie, model ID) of the intended sampler for the dwManufacturer. Contact the manufacturer of the sampler to ascertain the sampler's model ID. If the WAVE is not intended for a specific manufacturer's product, then this field should be set to 0.

The dwSamplePeriod field specifies the period of one sample in nanoseconds (normally 1/nSamplesPerSec from the Format chunk. But note that this field allows finer tuning than nSamplesPerSec). For example, 44.1 KHz would be specified as 22675 (0x00005893).

The dwMIDIUnityNote field is the MIDI note number at which the instrument plays back the waveform data without pitch modification (ie, at the same sample rate that was used when the waveform was created). This value ranges 0 through 127, inclusive. Middle C is 60.

The dwMIDIPitchFraction field specifies the fraction of a semitone up from the specified dwMIDIUnityNote. A value of 0x80000000 is 1/2 semitone (50 cents); a value of 0x00000000 represents no fine tuning between semitones.

The dwSMPTEFormat field specifies the SMPTE time format used in the dwSMPTEOffset field. Possible values are:

```
0  = no SMPTE offset (dwSMPTEOffset should also be 0)
24 = 24 frames per second
25 = 25 frames per second
29 = 30 frames per second with frame dropping ('30 drop')
30 = 30 frames per second
```

The dwSMPTEOffset field specifies a time offset for the sample if it is to be syncronized or calibrated according to a start time other than 0. The format of this value is 0xhhmmssff. hh is a signed Hours value [-23..23]. mm is an unsigned Minutes value [0..59]. ss is unsigned Seconds value [0..59]. ff is an unsigned value [0..(- 1)].

The cSampleLoops field is the number (count) of SampleLoop structures that are appended to this chunk. These structures immediately follow the cbSamplerData field. This field will be 0 if there are no SampleLoop structures.

The cbSamplerData field specifies the size (in bytes) of any optional fields that an application wishes to append to this chunk. An application which needed to save additional information (ie, beyond the above fields) may append additional fields to the end of this chunk, after all of the SampleLoop structures. These additional fields are also reflected in the ChunkSize, and remember that the chunk should be padded out to an even number of bytes. The cbSamplerData field will be 0 if no additional information is appended to the chunk.

What follows the above fields are any SampleLoop structures. Each SampleLoop structure defines one loop (ie, the start and end points of the loop, and how many times it plays). What follows any SampleLoop structures are any additional, proprietary sampler information that an application chooses to store.

SampleLoop Structure

```
typedef struct {
  long  dwIdentifier;
  long  dwType;
  long  dwStart;
  long  dwEnd;
  long  dwFraction;
  long  dwPlayCount;
} SampleLoop;
```

The dwIdentifier field contains a unique number (ie, different than the ID number of any other SampleLoop structure). This field may correspond with the dwIdentifier field of some CuePoint stored in the Cue chunk. In other words, the CuePoint structure which has the same ID number would be considered to be describing the same loop as this SampleLoop structure. Furthermore, this field corresponds to the dwIndentifier field of any label stored in the Associated Data List. In other words, the text string (within some chunk in the Associated Data List) which has the same ID number would be considered to be this loop's "name" or "label".

The dwType field is the loop type (ie, how the loop plays back) as so:

0	Loop forward (normal)
1	Alternating loop (forward/backward)
2	Loop backward
3 - 31	reserved for future standard types
32 - ?	sampler specific types (manufacturer defined)

The dwStart field specifies the startpoint of the loop. In other words, it's the byte offset from the start of waveformData[], where an offset of 0 would be at the start of the waveformData[] array (ie, the loop start is at the very first sample point).

The dwEnd field specifies the endpoint of the loop (ie, a byte offset).

The dwFraction field allows fine-tuning for loop fractional areas between samples. Values range from 0x00000000 to 0xFFFFFFFF. A value of 0x80000000 represents 1/2 of a sample length.

The dwPlayCount field is the number of times to play the loop. A value of 0 specifies an infinite sustain loop (ie, the wave keeps looping until some external force interrupts playback, such as the musician releasing the key that triggered that wave's playback).

The Sampler Chunk is optional. I don't know as if there is any limit of one per WAVE file. I don't see why there should be such a limit, since after all, an application may need to deal with several MIDI samplers.

Instrument chunk

The Instrument Chunk contains some of the same type of information as the Sampler chunk. So what else is new?

```
#define InstrumentID 'inst'  /* chunkID for Instruments Chunk */

typedef struct {
  ID     chunkID;
  long   chunkSize;

  unsigned char UnshiftedNote;
  char          FineTune;
  char          Gain;
  unsigned char LowNote;
  unsigned char HighNote;
  unsigned char LowVelocity;
  unsigned char HighVelocity;
} InstrumentChunk;
```

The ID is always **inst**. chunkSize should always be 7 since there are no fields of variable length.

The UnshiftedNote field is the same as the Sampler chunk's dwMIDIUnityNote field.

The FineTune field determines how much the instrument should alter the pitch of the sound when it is played back. Units are in cents (1/100 of a semitone) and range from -50 to +50. Negative numbers mean that the pitch of the sound should be lowered, while positive numbers mean that it should be raised. While not the same measurement is used, this field serves the same purpose as the Sampler chunk's dwFraction field.

The Gain field is the amount by which to change the gain of the sound when it is played. Units are decibels. For example, 0db means no change, 6db means double the value of each sample point (ie, every additional 6db doubles the gain), while -6db means halve the value of each sample point.

The LowNote and HighNote fields specify the suggested MIDI note range on a keyboard for playback of the waveform data. The waveform data should be played if the instrument is requested to play a note between the low and high note numbers, inclusive. The UnshiftedNote does not have to be within this range.

The LowVelocity and HighVelocity fields specify the suggested range of MIDI velocities for playback of the waveform data. The waveform data should be played if the note-on velocity is between low and high velocity, inclusive. The range is 1 (lowest velocity) through 127 (highest velocity), inclusive.

The Instrument Chunk is optional. No more than 1 Instrument Chunk can appear in one WAVE.

Audio Interchange File Format (or AIFF) is a file format for storing digital audio (waveform) data. It supports a variety of bit resolutions, sample rates, and channels of audio. This format is very popular upon Apple platforms, and is widely used in professional programs that process digital audio waveforms.

This format uses the Electronic Arts Interchange File Format method for storing data in "chunks". You should read the article About Interchange File Format before proceeding.

Data Types

A C-like language will be used to describe the data structures in the file. A few extra data types that are not part of standard C, but which will be used in this document, are:

extended	80 bit IEEE Standard 754 floating point number (Standard Apple Numeric Environment [SANE] data type Extended). This would be a 10 byte field.
pstring	Pascal-style string, a one-byte count followed by that many text bytes. The total number of bytes in this data type should be even. A pad byte can be added to the end of the text to accomplish this. This pad byte is not reflected in the count.
ID	A chunk ID (ie, 4 ASCII bytes) as described in About Interchange File Format.

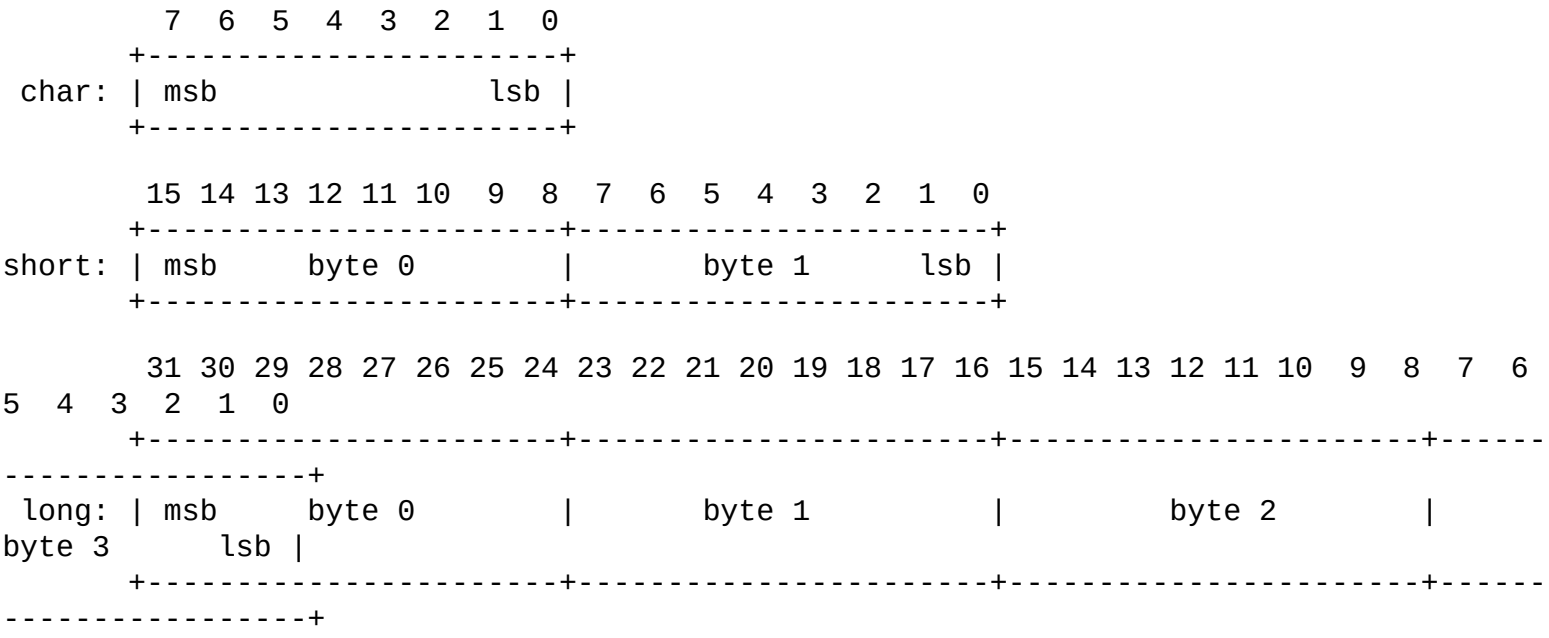
Also note that when you see an array with no size specification (e.g., char ckData[];), this indicates a variable-sized array in our C-like language. This differs from standard C arrays.

Constants

Decimal values are referred to as a string of digits, for example 123, 0, 100 are all decimal numbers. Hexadecimal values are preceded by a 0x - e.g., 0x0A, 0x1, 0x64.

Data Organization

All data is stored in Motorola 68000 (ie, big endian) format. The bytes of multiple-byte values are stored with the high-order (ie, most significant) bytes first. Data bits are as follows (ie, shown with bit numbers on top):



File Structure

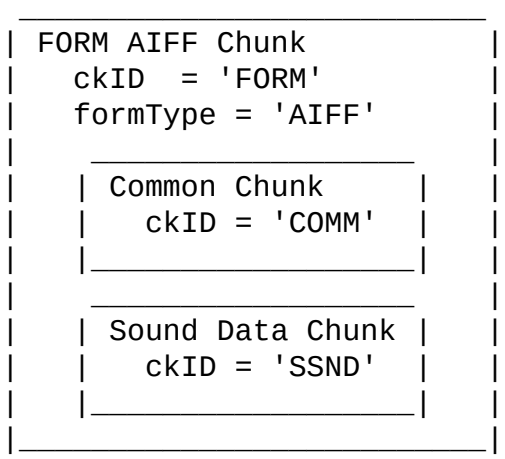
An Audio IFF file is a collection of a number of different types of chunks. There is a required Common Chunk which contains important parameters describing the waveform, such as its length and sample rate. The Sound Data chunk, which contains the

actual waveform data, is also required if the waveform data has a length greater than 0 (ie, there actually is waveform data in the FORM). All other chunks are optional. Among the other optional chunks are ones which define markers, list instrument parameters, store application-specific information, etc. All of these chunks are described in detail in the following sections of this document.

All applications that use FORM AIFF must be able to read the 2 required chunks and can choose to selectively ignore the optional chunks. A program that copies a FORM AIFF should copy all of the chunks in the FORM AIFF, even those it chooses not to interpret.

There are no restrictions upon the order of the chunks within a FORM AIFF.

Here is a graphical overview of an example, minimal AIFF file. It consists of a single FORM AIFF containing the 2 required chunks, a Common Chunk and a Sound Data Chunk.



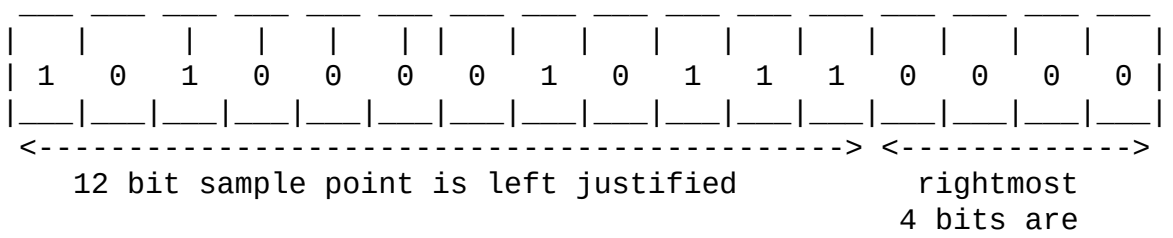
Sample Points and Sample Frames

A large part of interpreting Audio IFF files revolves around the two concepts of sample points and sample frames.

A sample point is a value representing a sample of a sound at a given moment in time. Each sample point is stored as a linear, 2's-complement value which may be from 1 to 32 bits wide (as determined by the sampleSize field in the Common Chunk). For example, each sample point of an 8-bit waveform would be an 8-bit byte (ie, a signed char).

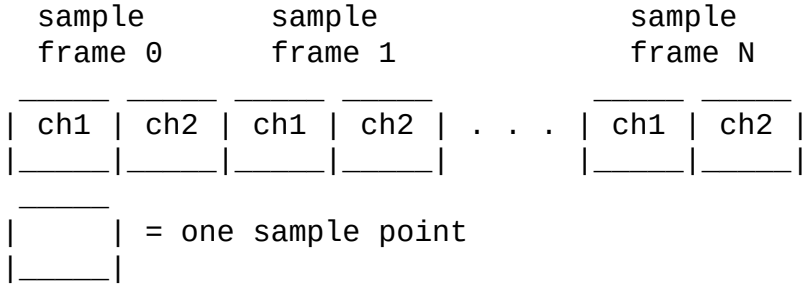
Because most CPU's read and write operations deal with 8-bit bytes, it was decided that a sample point should be rounded up to a size which is a multiple of 8 when stored in an AIFF. This makes the AIFF easier to read into memory. If your ADC produces a sample point from 1 to 8 bits wide, a sample point should be stored in an AIFF as an 8-bit byte (ie, signed char). If your ADC produces a sample point from 9 to 16 bits wide, a sample point should be stored in an AIFF as a 16-bit word (ie, signed short). If your ADC produces a sample point from 17 to 24 bits wide, a sample point should be stored in an AIFF as three bytes. If your ADC produces a sample point from 25 to 32 bits wide, a sample point should be stored in an AIFF as a 32-bit doubleword (ie, signed long). Etc.

Furthermore, the data bits should be left-justified, with any remaining (ie, pad) bits zeroed. For example, consider the case of a 12-bit sample point. It has 12 bits, so the sample point must be saved as a 16-bit word. Those 12 bits should be left-justified so that they become bits 4 to 15 inclusive, and bits 0 to 3 should be set to zero. Shown below is how a 12-bit sample point with a value of binary 101000010111 is stored left-justified as two bytes (ie, a 16-bit word).

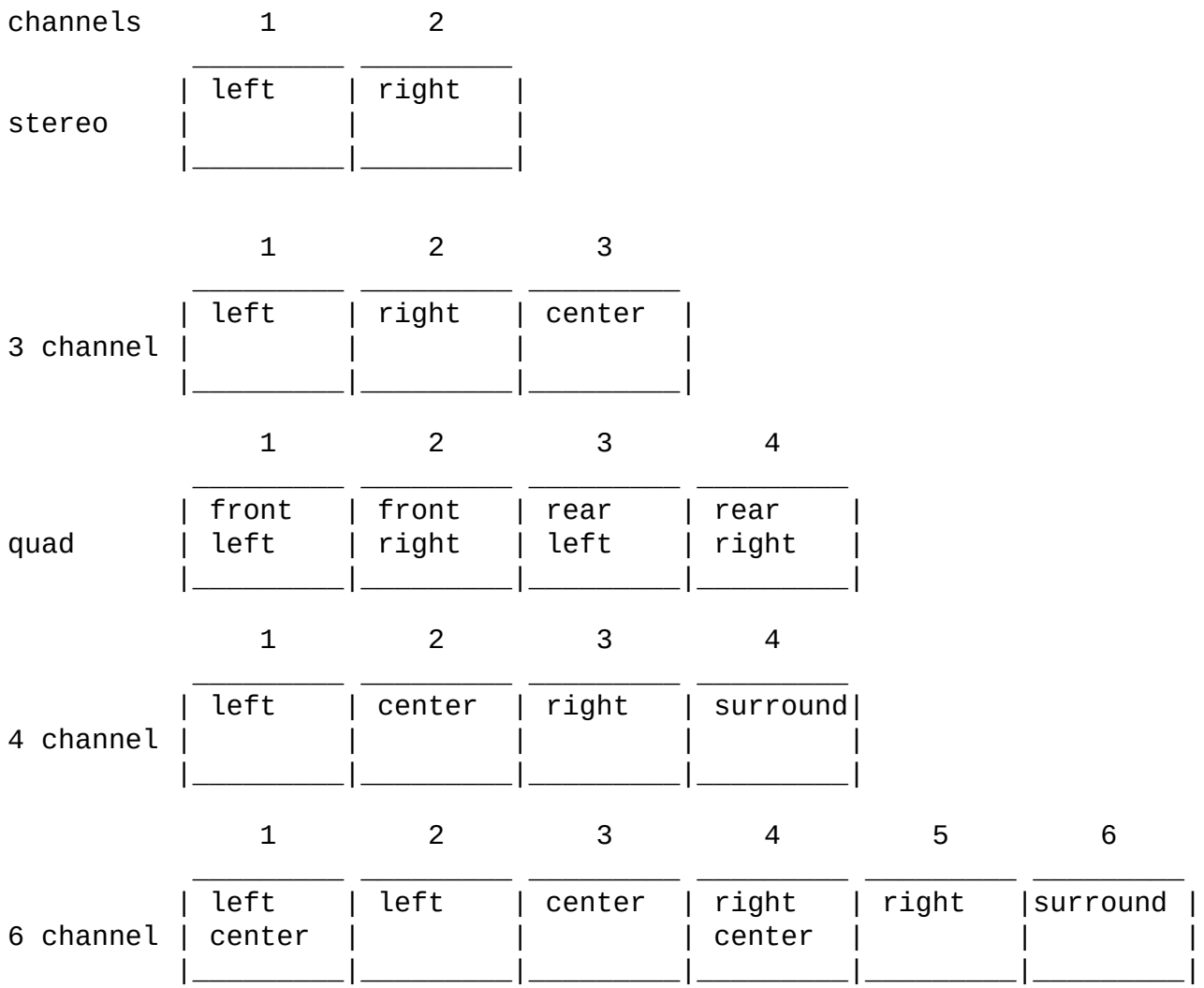


For multichannel sounds (for example, a stereo waveform), single sample points from each channel are interleaved. For example, assume a stereo (ie, 2 channel) waveform. Instead of storing all of the sample points for the left channel first, and then storing all of the sample points for the right channel next, you "mix" the two channels' sample points together. You would store the first sample point of the left channel. Next, you would store the first sample point of the right channel. Next, you would store the second sample point of the left channel. Next, you would store the second sample point of the right channel, and so on, alternating between storing the next sample point of each channel. This is what is meant by interleaved data; you store the next sample point of each of the channels in turn, so that the sample points that are meant to be "played" (ie, sent to a DAC) simultaneously are stored contiguously.

The sample points that are meant to be "played" (ie, sent to a DAC) simultaneously are collectively called a **sample frame**. In the example of our stereo waveform, every two sample points makes up another sample frame. This is illustrated below for that stereo example.



For a monophonic waveform, a sample frame is merely a single sample point (ie, there's nothing to interleave). For multichannel waveforms, you should follow the conventions shown below for which order to store channels within the sample frame. (ie, Below, a single sample frame is displayed for each example of a multichannel waveform).



The sample points within a sample frame are packed together; there are no unused bytes between them. Likewise, the sample frames are packed together with no pad bytes.

Common chunk

The Common Chunk describes fundamental parameters of the waveform data such as sample rate, bit resolution, and how many channels of digital audio are stored in the FORM AIFF.

```
#define CommonID 'COMM'   /* chunkID for Common Chunk */

typedef struct {
  ID             chunkID;
  long           chunkSize;

  short          numChannels;
  unsigned long  numSampleFrames;
  short          sampleSize;
  extended       sampleRate;
} CommonChunk;
```

The ID is always **COMM**. The chunkSize field is the number of bytes in the chunk. This does not include the 8 bytes used by ID and Size fields. For the Common Chunk, chunkSize should always 18 since there are no fields of variable length (but to maintain compatibility with possible future extensions, if the chunkSize is > 18, you should always treat those extra bytes as pad bytes).

The numChannels field contains the number of audio channels for the sound. A value of 1 means monophonic sound, 2 means stereo, 4 means four channel sound, etc. Any number of audio channels may be represented. For multichannel sounds, single sample points from each channel are interleaved. A set of interleaved sample points is called a sample frame.

The actual waveform data is stored in another chunk, the Sound Data Chunk, which will be described later.

The numSampleFrames field contains the number of sample frames. This is not necessarily the same as the number of bytes nor the number of sample points in the Sound Data Chunk (ie, it won't be unless you're dealing with a mono waveform). The total number of sample points in the file is numSampleFrames times numChannels.

The sampleSize is the number of bits in each sample point. It can be any number from 1 to 32.

The sampleRate field is the sample rate at which the sound is to be played back in sample frames per second.

One, and only one, Common Chunk is required in every FORM AIFF.

Sound Data chunk

The Sound Data Chunk contains the actual sample frames (ie, all channels of waveform data).

```
#define SoundDataID 'SSND'  /* chunk ID for Sound Data Chunk */

typedef struct {
  ID             chunkID;
  long           chunkSize;

  unsigned long  offset;
  unsigned long  blockSize;
```

```
  unsigned char  WaveformData[];
}  SoundDataChunk;
```

The ID is always **SSND**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size (ie, chunkSize is the number of remaining bytes in the chunk after the chunkSize field, not counting any trailing pad byte).

You can determine how many bytes of actual waveform data there is by subtracting 8 from the chunkSize. Remember that the number of sample frames, bit resolution, and other information is gotten from the Common Chunk.

The offset field determines where the first sample frame in the WaveformData starts. The offset is in bytes. Most applications won't use offset and should set it to zero. Use for a non-zero offset is explained in "Block-Aligning Waveform Data".

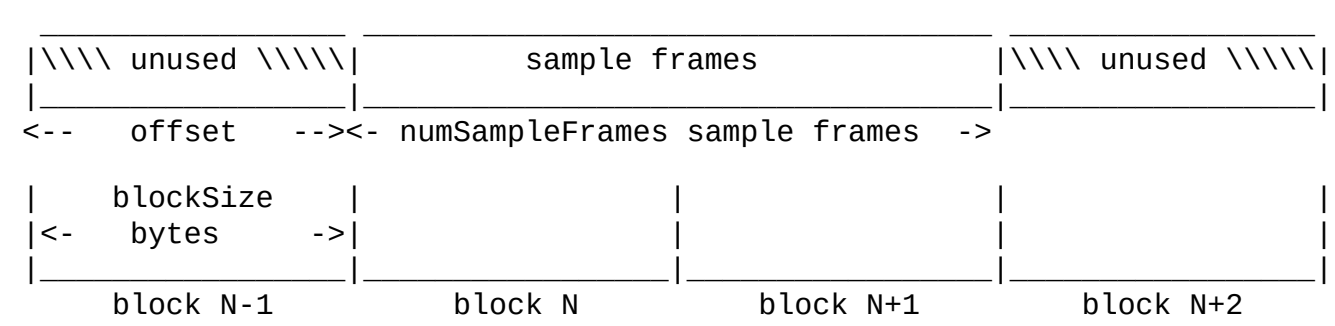
The blockSize is used in conjunction with offset for block-aligning waveform data. It contains the size in bytes of the blocks that waveform data is aligned to. As with offset, most applications won't use blockSize and should set it to zero. More information on blockSize is in "Block-Aligning Waveform Data".

The WaveformData array contains the actual waveform data. The data is arranged into what are called *sample frames* The number of sample frames in WaveformData is determined by the numSampleFrames field in the Common Chunk. For more information, see "Sample Points and Sample Frames".

The Sound Data Chunk is required unless the numSampleFrames field in the Common Chunk is zero. One, and only one, Sound Data Chunk may appear in a FORM AIFF.

Block-Aligning Waveform Data

There may be some applications that, to ensure real time recording and playback of audio, wish to align waveform data into fixed-size blocks. This alignment can be accomplished with the offset and blockSize parameters of the Sound Data Chunk, as shown below.



Above, the first sample frame starts at the beginning of block N. This is accomplished by skipping the first offset bytes (ie, some stored pad bytes) of the WaveformData. Note that there may also be pad bytes stored at the end of WaveformData to pad it out so that it ends upon a block boundary.

The blockSize specifies the size in bytes of the block to which you would align the waveform data. A blockSize of 0 indicates that the waveform data does not need to be block-aligned. Applications that don't care about block alignment should set the blockSize and offset to 0 when creating AIFF files. Applications that write block-aligned waveform data should set blockSize to the appropriate block size. Applications that modify an existing AIFF file should try to preserve alignment of the waveform data, although this is not required. If an application does not preserve alignment, it should set the blockSize and offset to 0. If an application needs to realign waveform data to a different sized block, it should update blockSize and offset accordingly.

Marker chunk

The Marker Chunk contains markers that point to positions in the waveform data. Markers can be used for whatever purposes an

application desires. The Instrument Chunk, defined later in this document, uses markers to mark loop beginning and end points.

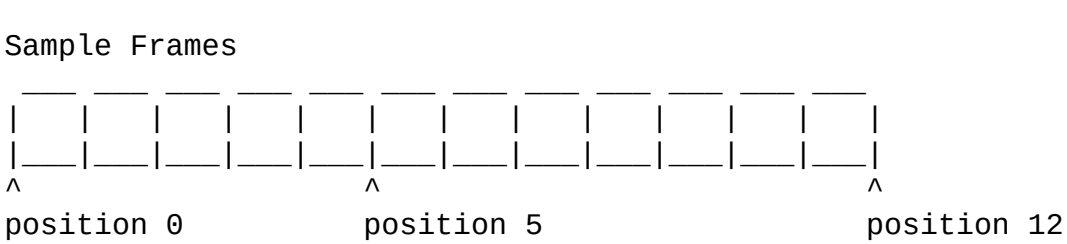
A marker structure is as follows:

```
typedef short  MarkerId;

typedef struct {
  MarkerID       id;
  unsigned long  position;
  pstring        markerName;
} Marker;
```

The id is a number that uniquely identifies that marker within an AIFF. The id can be any positive non-zero integer, as long as no other marker within the same FORM AIFF has the same id.

The marker's position in the WaveformData is determined by the position field. Markers conceptually fall between two sample frames. A marker that falls before the first sample frame in the waveform data is at position 0, while a marker that falls between the first and second sample frame in the waveform data is at position 1. Therefore, the units for position are sample frames, not bytes nor sample points.



The markerName field is a Pascal-style text string containing the name of the mark.

Note: Some "EA IFF 85" files store strings as C-strings (text bytes followed by a null terminating character) instead of Pascal-style strings. Audio IFF uses pstrings because they are more efficiently skipped over when scanning through chunks. Using pstrings, a program can skip over a string by adding the string count to the address of the first character. C strings require that each character in the string be examined for the null terminator.

Marker chunk format

The format for the data within a Marker Chunk is shown below.

```
#define MarkerID 'MARK'  /* chunkID for Marker Chunk */

typedef  struct {
  ID             chunkID;
  long           chunkSize;

  unsigned short  numMarkers;
  Marker          Markers[];
} MarkerChunk;
```

The ID is always **MARK**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields.

The numMarkers field is the number of marker structures in the Marker Chunk. If numMarkers is not 0, it is followed by that many marker structures, one after the other. Because all fields in a marker structure are an even number of bytes, the length of any marker will always be even. Thus, markers are packed together with no unused bytes between them. The markers need not be placed in any particular order.

The Marker Chunk is optional. No more than one Marker Chunk can appear in a FORM AIFF.

Instrument chunk

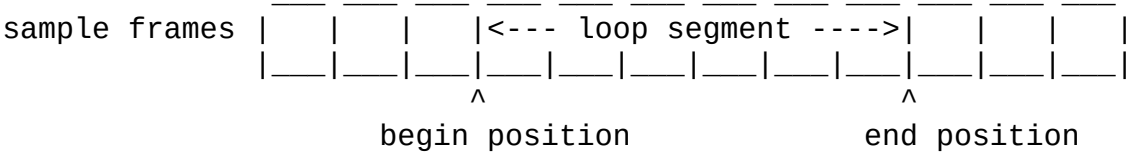
The Instrument Chunk defines basic parameters that an instrument, such as a MIDI sampler, could use to play the waveform data.

Looping

Waveform data can be looped, allowing a portion of the waveform to be repeated in order to lengthen the sound. The structure below describes a loop.

```
typedef struct {
  short     PlayMode;
  MarkerId  beginLoop;
  MarkerId  endLoop;
} Loop;
```

A loop is marked with two points, a begin position and an end position. There are two ways to play a loop, forward looping and forward/backward looping. In the case of forward looping, playback begins at the beginning of the waveform, continues past the begin position and continues to the end position, at which point playback starts again at the begin position. The segment between the begin and end positions, called the loop segment, is played repeatedly until interrupted by some action, such as a musician releasing a key on a musical controller.



With forward/backward looping, the loop segment is first played from the begin position to the end position, and then played backwards from the end position to the begin position. This flip-flop pattern is repeated over and over again until interrupted.

The playMode specifies which type of looping is to be performed:

```
#define NoLooping              0
#define ForwardLooping         1
#define ForwardBackwardLooping 2
```

If NoLooping is specified, then the loop points are ignored during playback.

The beginLoop is a marker id that marks the begin position of the loop segment.

The endLoop marks the end position of a loop. The begin position must be less than the end position. If this is not the case, then the loop segment has 0 or negative length and no looping takes place.

Instrument chunk format

The format of the data within an Instrument Chunk is described below.

```
#define InstrumentID 'INST'  /*chunkID for Instruments Chunk */

typedef struct {
  ID     chunkID;
  long   chunkSize;
```

```
  char   baseNote;
  char   detune;
  char   lowNote;
  char   highNote;
  char   lowvelocity;
  char   highvelocity;
  short  gain;
  Loop   sustainLoop;
  Loop   releaseLoop;
} InstrumentChunk;
```

The ID is always **INST**. chunkSize should always be 20 since there are no fields of variable length.

The baseNote is the note number at which the instrument plays back the waveform data without pitch modification (ie, at the same sample rate that was used when the waveform was created). Units are MIDI note numbers, and are in the range 0 through 127. Middle C is 60.

The detune field determines how much the instrument should alter the pitch of the sound when it is played back. Units are in cents (1/100 of a semitone) and range from -50 to +50. Negative numbers mean that the pitch of the sound should be lowered, while positive numbers mean that it should be raised.

The lowNote and highNote fields specify the suggested note range on a keyboard for playback of the waveform data. The waveform data should be played if the instrument is requested to play a note between the low and high note numbers, inclusive. The base note does not have to be within this range. Units for lowNote and highNote are MIDI note values.

The lowVelocity and highVelocity fields specify the suggested range of velocities for playback of the waveform data. The waveform data should be played if the note-on velocity is between low and high velocity, inclusive. Units are MIDI velocity values, 1 (lowest velocity) through 127 (highest velocity).

The gain is the amount by which to change the gain of the sound when it is played. Units are decibels. For example, 0db means no change, 6db means double the value of each sample point (ie, every additional 6db doubles the gain), while -6db means halve the value of each sample point.

The sustainLoop field specifies a loop that is to be played when an instrument is sustaining a sound.

The releaseLoop field specifies a loop that is to be played when an instrument is in the release phase of playing back a sound. The release phase usually occurs after a key on an instrument is released.

The Instrument Chunk is optional. No more than 1 Instrument Chunk can appear in one FORM AIFF.

MIDI Data chunk

The MIDI Data Chunk can be used to store MIDI data.

The primary purpose of this chunk is to store MIDI System Exclusive messages, although other types of MIDI data can be stored in the chunk as well. As more instruments come to market, they will likely have parameters that have not been included in the AIFF specification. The Sys Ex messages for these instruments may contain many parameters that are not included in the Instrument Chunk. For example, a new MIDI sampler may support more than the two loops per waveform. These loops will likely be represented in the Sys Ex message for the new sampler. This message can be stored in the MIDI Data Chunk (ie, so you have some place to store these extra loop points that may not be used by other instruments).

```
#define MIDIDataID 'MIDI' /* chunkID for MIDI Data Chunk */
```

```
typedef struct {
  ID             chunkID;
  long           chunkSize;

  unsigned char  MIDIdata[];
} MIDIDataChunk;
```

The ID is always **MIDI**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size.

The MIDIData field contains a stream of MIDI data. There should be as many bytes as chunkSize specifies, plus perhaps a pad byte if needed.

The MIDI Data Chunk is optional. Any number of these chunks may exist in one FORM AIFF. If MIDI System Exclusive messages for several instruments are to be stored in a FORM AIFF, it is better to use one MIDI Data Chunk per instrument than one big MIDI Data Chunk for all of the instruments.

Audio Recording chunk

The Audio Recording Chunk contains information pertinent to audio recording devices.

```
#define AudioRecording ID 'AESD'   /* chunkID for Audio Recording Chunk. */

typedef struct {
  ID             chunkID
  long           chunkSize;

  unsigned char  AESChannelStatusData[24];
} AudioRecordingChunk;
```

The ID is always **AESD**. chunkSize should always be 24 since there are no fields of variable length.

The 24 bytes of AESCChannelStatusData are specified in the "AES Recommended Practice for Digital Audio Engineering - Serial Transmission Format for Linearly Represented Digital Audio Data", transmission of digital audio between audio devices. This information is duplicated in the Audio Recording Chunk for convenience. Of general interest would be bits 2, 3, and 4 of byte 0, which describe recording emphasis.

The Audio Recording Chunk is optional. No more than 1 Audio Recording Chunk may appear in one FORM AIFF.

Application Specific chunk

The Application Specific Chunk can be used for any purposes whatsoever by developers and application authors. For example, an application that edits sounds might want to use this chunk to store editor state parameters such as magnification levels, last cursor position, etc.

```
#define ApplicationSpecificID 'APPL' /* chunkID for Application Specific Chunk. */

typedef struct {
  ID       chunkID;
  long     chunkSize;

  char     applicationSignature[4];
  char     data[];
```

```
} ApplicationSpecificChunk;
```

The ID is always **APPL**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size.

The applicationSignature field is used by applications which run on platforms from Apple Computer, Inc. For the Apple II, this field should be set to 'pdos'. For the Mac, this field should be set to the application's four character signature as registered with Apple Technical Support.

The data field is the data specific to the application. The application determines how many bytes are stored here, and what their purpose are. A trailing pad byte must follow if that is needed in order to make the chunk an even size.

The Application Specific Chunk is optional. Any number of these chunks may exist in a one FORM AIFF.

Comments chunk

The Comments Chunk is used to store comments in the FORM AIFF. Standard IFF has an Annotation Chunk that can also be used for comments, but this new Comments Chunk has two fields (per comment) not found in the Standard IFF chunk. They are a time-stamp for the comment and a link to a marker.

Comment structure

A Comment structure consists of a time stamp, marker id, and a text count followed by text.

```
typedef struct {
  unsigned long   timeStamp;
  MarkerID        marker;
  unsigned short  count;
  char            text[];
} Comment;
```

The timeStamp indicates when the comment was created. On the Amiga, units are the number of seconds since January 1, 1978. On the Mac, units are the number of seconds since January 1, 1904.

A comment can be linked to a marker. This allows applications to store long descriptions of markers as a comment. If the comment is referring to a marker, then the marker field is the ID of that marker. Otherwise, marker is 0, indicating that this comment is not linked to any marker.

The count is the length of the text that makes up the comment. This is a 16-bit quantity, allowing much longer comments than would be available with a pstring. This count does not include any possible pad byte needed to make the comment an even number of bytes in length.

The text field contains the comment itself, followed by a pad byte if needed to make the text field an even number of bytes.

Comments chunk format

```
#define CommentID 'COMT'  /* chunkID for Comments Chunk */

typedef struct {
  ID              chunkID;
  long            chunkSize;

  unsigned short  numComments;
```

```
  char            comments[];
}CommentsChunk;
```

The ID is always **COMT**. chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields.

The numComments field contains the number of Comment structures in the chunk. This is followed by the Comment structures, one after the other. Comment structures are always even numbers of bytes in length, so there is no padding needed between structures.

The Comments Chunk is optional. No more than 1 Comments Chunk may appear in one FORM AIFF.

Text Chunks, Name, Author, Copyright, Annotation

These four optional chunks are included in the definition of every Standard IFF file.

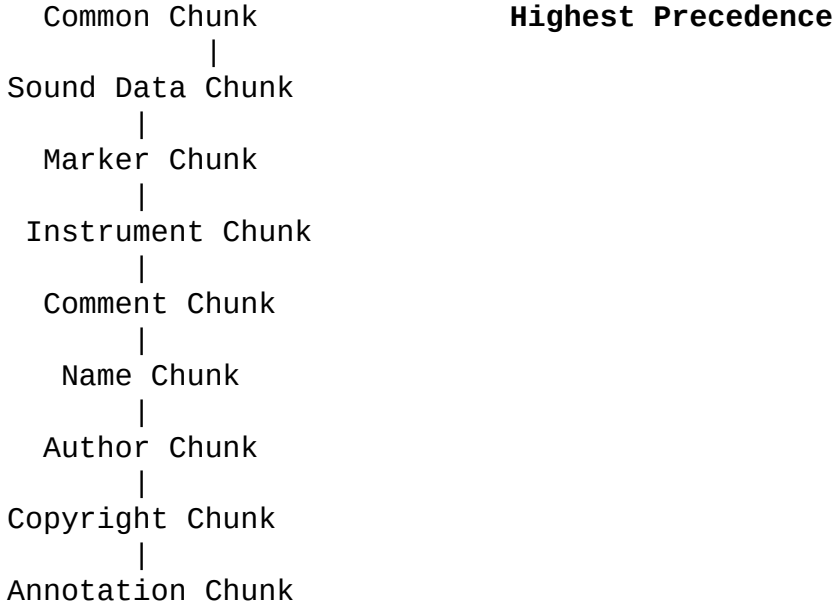
```
#define NameID 'NAME'   /* chunkID for Name Chunk */
#define NameID 'AUTH'   /* chunkID for Author Chunk */
#define NameID '(c) '   /* chunkID for Copyright Chunk */
#define NameID 'ANNO'   /* chunkID for Annotation Chunk */

typedef struct {
  ID      chunkID;
  long    chunkSize;
  char    text[];
}TextChunk;
```

chunkSize is the number of bytes in the chunk, not counting the 8 bytes used by ID and Size fields nor any possible pad byte needed to make the chunk an even size.

Chunk Precedence

Several of the local chunks for FORM AIFF may contain duplicate information. For example, the Instrument Chunk defines loop points and some MIDI Sys Ex data in the MIDI Data Chunk may also define loop points. What happens if these loop points are different? How is an application supposed to loop the sound? Such conflicts are resolved by defining a precedence for chunks. This precedence is illustrated below.



```
            |
  Audio Recording Chunk
            |
     MIDI Data Chunk
            |
Application Specific Chunk      Lowest Precedence
```

The Common Chunk has the highest precedence, while the Application Specific Chunk has the lowest. Information in the Common Chunk always takes precedence over conflicting information in any other chunk. The Application Specific Chunk always loses in conflicts with other chunks. By looking at the chunk hierarchy, for example, one sees that the loop points in the Instrument Chunk take precedence over conflicting loop points found in the MIDI Data Chunk.

It is the responsibility of applications that write data into the lower precedence chunks to make sure that the higher precedence chunks are updated accordingly (ie, so that conflicts tend not to exist).

Errata

The Apple IIGS Sampled Instrument Format also defines a chunk with ID of "INST," which is not the same as the AIFF Instrument Chunk. A good way to tell the two chunks apart in generic IFF-style readers is by the chunkSize fields. The AIFF Instrument Chunk's chunkSize field is always 20, whereas the Apple IIGS Sampled Instrument Format Instrument Chunk's chunkSize field, for structural reasons, can never be 20.

Storage of AIFF on Apple and Other Platforms

On a Macintosh, the FORM AIFF, is stored in the data fork of an Audio IFF file. The Macintosh file type of an Audio IFF file is 'AIFF'. This is the same as the formType of the FORM AIFF. Macintosh applications should not store any information in Audio IFF file's resource fork, as this information may not be preserved by all applications. Applications can use the Application Specific Chunk, defined later in this document, to store extra information specific to their application.

Audio IFF files may be identified in other Apple file systems as well. On a Macintosh under MFS or HFS, the FORM AIFF is stored in the data fork of a file with file type "AIFF." This is the same as the formType of the FORM AIFF.

On an operating system such as MS-DOS or UNIX, where it is customary to use a file name extension, it is recommended that Audio IFF file names use ".AIF" for the extension.

Referring to Audio IFF

The official name is "Audio Interchange File Format". If an application needs to present the name of this format to a user, such as in a "Save As..." dialog box, the name can be abbreviated to Audio IFF. Referring to Audio IFF files by a four-letter abbreviation (ie, "AIFF") at the user-level should be avoided.

Converting Extended data to a unsigned long

The sample rate field in a Common Chunk is expressed as an 80 bit IEEE Standard 754 floating point number. This isn't a very useful format for computer software and audio hardware that can't directly deal with such floating point values. For this reason, you may wish to use the following ConvertFloat() hack to convert the floating point value to an unsigned long. This function doesn't handle very large floating point values, but certainly larger than you would ever expect a typical sampling rate to be. This function assumes that the passed *buffer* arg is a pointer to the 10 byte array which already contains the 80-bit floating point value. It returns the value (ie, sample rate in Hertz) as an unsigned long. Note that the FlipLong() function is only of use to Intel CPU's which have to deal with AIFF's Big Endian order. If not compiling for an Intel CPU, comment out the INTEL_CPU define.

```
#define INTEL_CPU
```

```
#ifdef INTEL_CPU
/* ************************** FlipLong() ****************************
 * Converts a long in "Big Endian" format (ie, Motorola 68000) to Intel
 * reverse-byte format, or vice versa if originally in Big Endian.
 ****************************************************************** */

void FlipLong(unsigned char * ptr)
{
   register unsigned char val;

   /* Swap 1st and 4th bytes */
   val = *(ptr);
   *(ptr) = *(ptr+3);
   *(ptr+3) = val;

   /* Swap 2nd and 3rd bytes */
   ptr += 1;
   val = *(ptr);
   *(ptr) = *(ptr+1);
   *(ptr+1) = val;
}
#endif

/* ************************ FetchLong() ****************************
 * Fools the compiler into fetching a long from a char array.
 ****************************************************************** */

unsigned long FetchLong(unsigned long * ptr)
{
   return(*ptr);
}

/* ************************ ConvertFloat() ****************************
 * Converts an 80 bit IEEE Standard 754 floating point number to an unsigned
 * long.
 ****************************************************************** */

unsigned long ConvertFloat(unsigned char * buffer)
{
   unsigned long mantissa;
   unsigned long last = 0;
   unsigned char exp;

#ifdef INTEL_CPU
   FlipLong((unsigned long *)(buffer+2));
#endif

   mantissa = FetchLong((unsigned long *)(buffer+2));
   exp = 30 - *(buffer+1);
   while (exp--)
   {
     last = mantissa;
     mantissa >>= 1;
   }
   if (last & 0x00000001) mantissa++;
   return(mantissa);
}
```

Of course, you may need a complementary routine to take a sample rate as an unsigned long, and put it into a buffer formatted as that 80-bit floating point value.

```
/* ************************ StoreLong() ****************************
 * Fools the compiler into storing a long into a char array.
 ***************************************************************** */

void StoreLong(unsigned long val, unsigned long * ptr)
{
   *ptr = val;
}

/* ************************* StoreFloat() ****************************
 * Converts an unsigned long to 80 bit IEEE Standard 754 floating point
 * number.
 ***************************************************************** */

void StoreFloat(unsigned char * buffer, unsigned long value)
{
   unsigned long exp;
   unsigned char i;

   memset(buffer, 0, 10);

   exp = value;
   exp >>= 1;
   for (i=0; i<32; i++)
   {
      exp >>= 1;
      if (!exp) break;
   }
   *(buffer+1) = i;

   for (i=32; i; i--)
   {
      if (value & 0x80000000) break;
      value <<= 1;
   }

   StoreLong(value, (unsigned long *)(buffer+2));

#ifdef INTEL_CPU
   FlipLong((unsigned long *)(buffer+2));
#endif
}
```

Multi-sampling

Many MIDI samplers allow splitting up the MIDI note range into smaller ranges (for example by octaves) and assigning a different waveform to play over each range. If you wanted to store all of those waveforms into a single data file, what you would do is create an IFF LIST (or perhaps CAT if you wanted to store FORMs other than AIFF in it) and then include an embedded FORM AIFF for each one of the waveforms. (ie, Each waveform would be in a separate FORM AIFF, and all of these FORM AIFFs would be in a single LIST file). See About Interchange File Format for details about LISTs.

Unlike with some other operating systems, in Windows, a program should refrain from directly reading and writing hardware ports on a sound card. Whenever possible, a program should instead call functions in the Windows operating system, which will do the actual hardware reading and writing for you (in conjunction with the sound card's Windows device driver). For writing data to the card, you pass that data to an operating system function that sends that data to the card's device driver, which in turn writes that data to the card. For reading data from the card, you call an operating system function that causes the driver to read some data from the card, perhaps placing that data into a buffer whose address you've specified. In this way, you'll create a Windows program that will:

1. continue to operate under versions of Windows which restrict access to hardware, in order to implement "crash protection" (for example, Windows NT/2000/XP). Typically, such versions allow drivers to operate in special "modes" that allow hardware access, but restrict programs from the same access.

2. work with a wide range of sound cards and MIDI interfaces. It's the driver that has the hardware specific code in it, not your program. The drivers for various hardware all look the same to the program and operating system. (ie, Every driver has the same set of functions in it, which are called by the operating system using one "standard" method for all. Therefore, your program automatically supports all such drivers and their hardware by virtue of calling Windows operating system functions).

So, is there a "standard" that all Windows sound/MIDI drivers are written to follow (ie, such that each driver has the same set of functions, to be called using the same method)? Yes. This is called Windows "Media Control Interface" (ie, MCI). The Windows MIDI Mapper, the MCI Sequencer Device (driver), and the MCI Wave Audio Device (driver) are 3 parts of MCI that are particularly relevant to sound/MIDI cards.

When a user buys a sound/MIDI card, he installs the Windows MCI driver that ships with it (for example, using Windows Control Panel's *Add new Hardware*. See the explanation of installing a driver under Win95 for more details). Virtually every PC sound/MIDI card ships with a Windows MCI driver for that card. You should never have to write a Windows driver for a card unless you're the manufacturer of that card. As a program developer, you don't care what particular brand/model card is being used (as long as it supports what you need to do). You don't even need to know the name of the driver, because you call it indirectly via functions in the Windows operating system.

Note: Windows programs that call these MIDI and digital audio functions in the Windows operating system should include the header MMSYSTEM.H (and MMREG.H for Win32), and also be linked with the library WINMM.LIB (or MMSYSTEM.DLL if a Win3.1 program).

Windows' lists of devices

In any computer, there can be more than one installed card capable of inputting or outputting MIDI data. Likewise, there can be more than one installed card capable of recording/playing digital audio data. (Needless to say, each card will have its own driver installed). For this reason, Windows

maintains a list of all of the installed devices that support MIDI and Digital Audio. The list of devices that support MIDI is a separate list from the list of devices that support Digital Audio. So, if one card happens to support both Digital Audio and MIDI, a name for it will appear in both lists. In fact, Windows differentiates between devices that support MIDI input and devices that support MIDI output, and maintains separate lists for each. So, if one card happens to support both MIDI input and output, a name for it will appear in both lists. Likewise, Windows maintains separate lists for devices that record Digital Audio as well as devices that play Digital Audio.

Therefore, Windows maintains 4 lists:

1. Devices that can input (or create) MIDI data.

2. Devices that can output MIDI data (or play it upon some sort of built-in sound module).

3. Devices that can record (or create) Digital Audio data.

4. Devices that can output (ie, usually play) Digital Audio data.

For a card that can input/output MIDI data, as well as record/play Digital Audio data, a name for it will appear in all 4 lists.

What name for a device appears in each list? Well, that is entirely up to the card's device driver. It is the card's device driver that tells Windows into which lists Windows should place the card's name, and what name to use for each list. And in fact, if the card has several different components that can logically fit into one list, then the device driver may tell Windows to put several component names into that list.

Let's take an example for illustration: A Creative Labs' AWE32 sound card. This card has the following components:

1. An ADC that can record stereo Digital Audio.

2. A DAC that can play stereo Digital Audio.

3. A MIDI In port that can input MIDI data.

4. A MIDI Out port that can output MIDI data.

5. An FM Synth sound module that can play MIDI data.

6. A Wavetable Synth sound module that can play MIDI data.

Notice that the last 3 components can all fit into the list of devices that support outputting or playing MIDI data. So, the AWE32 driver is going to tell Windows to put 3 names into that one list. For

example, the driver may tell Windows to put the name "AWE32 MIDI Out" into the list. The driver may tell Windows to also put the name "AWE32 FM Synth" into that same list. Finally, the driver may tell Windows to also put the name "AWE32 Wavetable Synth" into that same list. Although these 3 components are all on the same card, to Windows, they represent 3 individual devices that are capable of outputting/playing MIDI data, and Windows treats them as if they really are 3 separate devices. (Incidentally, there is no particular rules about naming the components. I just arbitrarily picked the above 3 names, although they do make sense. A manufacturer can put any names he wants into his driver for inclusion into those Windows lists).

Of course, the AWE32 driver will also tell Windows to put the name "AWE32 Digital Audio In" into the list of devices that support recording Digital Audio. The driver will also tell Windows to put the name "AWE32 Digital Audio Out" into the list of devices that support playing Digital Audio. Finally, the driver will also tell Windows to put the name "AWE32 MIDI In" into the list of devices that support recording or creating MIDI data.

Windows assigns a **Device ID** (ie, merely an unsigned long numeric value) to each name in its lists. The first device in each list gets a value of 0, and the subsequent devices have increasing values for their IDs. (ie, The second device in a list has a Device ID of 1, the third device has an ID of 2, etc).

So in our above example, the 3 components that the AWE32 device driver told Windows to add to the list of devices that support playing MIDI data ("AWE32 MIDI Out", "AWE32 FM Synth", and "AWE32 Wavetable Synth") will have Device IDs of 0, 1, and 2 respectively (assuming that they were added to the list in that order and are the first 3 names to be added to the list).

In fact, drivers for MIDI interfaces that have multiple MIDI Outputs typically add a separate name to the list for each MIDI Output, for example "MIDI Out #1 for Brand X card", "MIDI Out #2 for Brand X card", etc.

The one caveat to this is the "MIDI Mapper" device driver which is part of Windows. This doesn't appear in the list of devices that support outputting MIDI data, although the MIDI Mapper is indeed a device to which your program can output MIDI data. MIDI Mapper always has a Device ID of -1.

For a discussion of writing a program that properly shares MIDI inputs and outputs, see Managing MIDI Ports.

Low level API

The "low level" Windows functions are called that because they require that your program does a lot of the work of recording/playing MIDI and Digital Audio data, short of actually manipulating the sound card hardware.

For example, if you want to play a MIDI file (ie, play a musical sequence), then you have to load the MIDI data contained within that file, use a (Windows software) timer to determine when it is time to output each MIDI message, and when that time occurs, pass that message (ie, its data bytes) to a

Windows function that actually outputs the bytes.

If you want to record a MIDI file, then you have to tell Windows to notify you when the card's driver has input the next MIDI message and to request Windows to pass you those MIDI data bytes, use a (Windows software) timer to timestamp that MIDI message, and write the bytes to disk (typically in MIDI File Format).

Because your program does most of the work, you have a lot of control over the playback/recording process, such as the ability to vary tempo, or perform filtering/mixing/transposing/etc of data on-the-fly.

For a more indepth discussion of writing a program that uses the Low level MIDI functions, see Low level MIDI API.

There are also low level functions for Digital Audio. Again, these require that you do a lot of work to record/play the digital audio data.

For example, if you want to play a WAVE file (ie, play digital audio), then you have to load the digital audio data contained within that file in "blocks" (ie, usually filling a buffer of 4K of data at a time), and feed it to a Windows function that actually outputs the data. In order to get a smooth, continuous playback of digital audio, you typically have to do double-buffering (ie, while the card is playing back one buffer of 4K data, you need to be simultaneously filling a second buffer with the next 4K of data).

If you want to record a WAVE file, then Windows has to feed you a continuous stream of those blocks of data, and you must write each block to disk (typically in WAVE File Format). In order to get a smooth, continuous recording of digital audio, you typically have to do double-buffering (ie, while the card is recording one buffer of 4K data, you need to be simultaneously writing the buffer containing the previous 4K of data to disk).

For a more indepth discussion of writing a program that uses the Low level Digital Audio functions, see Low level Digital Audio API.

High level API

The "high level" Windows functions are called that because they relieve your program of a lot of the burden of recording/playing MIDI and Digital Audio data.

For example, if you want to play a MIDI file (ie, play a musical sequence), then you merely call one Windows function, specifying the name of the MIDI file to be played back. Windows will open the MIDI file and load the MIDI data contained within that file, use a (Windows software) timer to determine when it is time to output each MIDI message, and actually output the bytes. In other words, Windows plays the MIDI file in the background while your program can go on to do other things simultaneously.

If you want to record a MIDI file, then you merely call one Windows function, specifying the name of the MIDI file to be created. Again, Windows does all of the work of reading the incoming MIDI data, time-stamping each message, and writing the data to disk in MIDI File Format. (NOTE: At this time, Windows does not offer High level API support for MIDI recording).

But because your program doesn't do most of the work, you lose a lot of control over the playback/recording process, such as being able to perform filtering of data on-the-fly, etc. You're pretty much limited to being able to stop, start, pause, rewind, and fast-forward the playback.

For a more indepth discussion of writing a program that uses the High level MIDI functions, see High level MIDI API.

There is also a high level API for playing/recording WAVE files. Just like with the High level MIDI API, Windows does a lot of the work (ie, completely loads and plays the WAVE file in the background), but again, you lose a lot of control over the actual playback (for example being able to do on-the-fly mixing of digital audio -- to implement virtual tracks. For a more indepth discussion of virtual tracks, see the article Digital Audio on a computer).

For a more indepth discussion of writing a program that uses the High level Digital Audio functions, see High level Digital Audio API.

A new, "inbetween" MIDI API

A further option for MIDI sequencing (ie, the playback of musical excerpts using MIDI events) under Win95/98/ME/2000/XP and WinNT (4.X and above only) is to use the Stream API. This allows you to store a MIDI message in a special structure that contains a field that specifies the amount of time to delay before outputting the MIDI message. Windows does the actual timing of the MIDI playback. Unlike with the High level MIDI API, you still load the MIDI data and feed each message (or more likely a block of messages) to Windows (ie, so you have more control over the playback on a per-message basis), but you no longer have to time the playback. In other words, the Stream API is useful for playing back musical sequences where you still have a lot of control over individual events and can therefore perform various filtering and modification of MIDI data on-the-fly, but don't want to deal with timing issues (which is particularly aggravating under Win95/98/ME due to the timing discrepancies of 32-bit MultiMedia Timer callbacks).

Currently, the Stream API supports playback only (not recording -- ie, it can't supply you with time-stamped incoming MIDI messages). So, if you need to do MIDI recording, you'll have to use the Low level MIDI API and a MultiMedia Timer, or use DirectMusic.

For a more indepth discussion of writing a program that uses the MIDI Stream functions, see MIDI Stream API.

A new, fancier Digital Audio API

Many game programs use lots of digital audio for sound effects, and various waveforms often have to be mixed on-the-fly. For this purpose, Microsoft devised a new set of functions called "DirectSound" (ie, part of DirectX) under Win95/98/ME/2000/XP and WinNT (4.X and above only). These functions let the operating system take care of mixing several waveforms to a sound card's stereo DAC.

DirectX versions prior to 6.1 support playback, but not recording.

A card's device driver should have extra support to work well with the DirectSound API (although older drivers not directly supporting DirectSound may work with this API albeit very inefficiently and/or with some features disabled).

For a more indepth discussion of writing a program that uses the DirectSound functions, see DirectSound API.

A new, less archaic MIDI API

Many game programs also use MIDI for background music. For this purpose, Microsoft devised a new set of functions called "DirectMusic" (ie, part of DirectX) under Win95/98/ME/2000/XP and WinNT (4.X and above only). These functions are a bit newer than the older low level MIDI API, and address some of the oddities/discrepancies that plague the legacy MIDI API.

DirectX versions prior to 6.0 do not include DirectMusic, and versions prior to 6.1 support only playback.

A card's device driver should have extra support to work well with the DirectMusic API (although older drivers not directly supporting DirectMusic may work with this API albeit very inefficiently and/or with some features disabled).

For a more indepth discussion of writing a program that uses the DirectMusic functions, see DirectMusic API.

Miscellaneous Functions (ie, the Mixer API, and Aux API)

The Mixer API has functions to get a listing of all of the various inputs and outputs on a particular card (ie, how many Aux inputs, Line inputs, Mic inputs, Line outputs, etc, it has), and to adjust all of the volumes of these various components, as well as perform other operations such as muting inputs. This is a new API added to Win95/98/ME/2000/XP and WinNT (4.X and above only).

A card's device driver needs extra support to work with the Mixer API. Not all Windows drivers have this support. Win3.1 drivers do not.

For a more indepth discussion of writing a program that uses the Mixer functions, see Mixer API.

Older drivers should at least support the Aux API. Just like with the lists for devices that support MIDI input and output, and Digital Audio playback and recording, Windows maintains a list of all devices that have Aux outputs. (And again, a card's device driver instructs Windows to put some name into this list). The Aux functions allow a program to control the volume (and perhaps other settings) of each Aux output. Typically, audio from a CDROM or the audio output of a voice modem is connected to a sound card's Aux output. (ie, So, it's the volume of these that you're affecting).

For a more indepth discussion of writing a program that uses the Aux functions, see Aux API.

MultiMedia File functions

When programs use the low level MIDI and Digital Audio API, they typically have to open, read, and write MIDI and WAVE File Format files. To help parse Interchange File Format types of files (ie, files whose data is arranged into nested chunks), Windows has a MultiMedia File (ie, Mmio) API.

For a more indepth discussion of writing a program that uses the MultiMedia File API, see MultiMedia File API.

Misc Topics

The following online Windows book is available for download. It gives more information about the Windows MIDI and digital audio functions (as well as Aux API and MultiMedia Timer API). But note that this book dates back to Windows 3.1, so some newer features, APIs, and notes may be omitted. Use the tutorials on this site for more uptodate information.

Windows MultiMedia System Book

For a more indepth discussion of the history of why and how the Windows MIDI and digital audio functions came to be, read the introduction to the article Audio cards and MIDI Interfaces for a computer.

This article answers questions about MIDI and audio setup under Windows 95/98/ME/2000/XP. For more general questions/answers about MIDI/audio/computer setups, and trouble-shooting, read the FAQ "MIDI connections and computer setup".

If you're having a problem with an internal IBM PC card not working, then you should first check for any hardware conflicts in your system. Read the article "Resolving hardware conflicts" for more information.

Questions in this FAQ are:

How do I install/setup an audio/MIDI driver?

Nowadays, most driver packages ship with a program you can run which installs the driver for you. Typically, this program is named ***setup.exe***. There will often be numerous other files (in the package) along with setup.exe, but these are used only by setup.exe to install driver support. (Once the driver support is installed, you can delete setup.exe and all other files in the package, or save them if you need to reinstall). If the driver package is shipped on a CDROM, the CDROM will typically have a file named ***autorun.inf*** on it. All autorun.inf does is automatically run setup.exe when you insert the CDROM into the drive. (This is a function of Windows' autorun capability. Unless you have this feature disabled, Windows will "autorun" any CDROM with an autorun.inf file on it).

Setup.exe will automatically detect what operating system is on your computer, and install the correct driver. (ie, Many maunfacturers will ship one CDROM containing drivers for several versions of Windows. But not every driver is compatible with all versions of Windows, so the correct one must be installed for your system).

If the driver does not ship with its own setup program, then you will have to manually install the driver yourself. (Some manufacturers farm out their driver development to programmers who don't know how to, or don't take the time to, write a self-installing driver package).

To manually install a driver, the driver should have a .INF file (ie, created by the manufacturer and shipped with the driver). The INF file (ie, the filename ends with a .INF extension) is really just a text

file that tells Windows what kind of driver is being installed (ie, digital audio, MIDI, video, printer, etc). It also contains the text that Windows uses when it displays the "name" of the driver (ie, not the driver filename, but rather, a more meaningful name which I'll call the "device name") in various displays, such as in the Control Panel's MultiMedia notebook. The INF also lists what the driver filenames are so that Windows knows what files to copy off of the install disk. You can view this INF file in a text editor (and even edit the device name). For example, the Roland RAP-10 INF indicates that the driver supports digital audio, MIDI, and LINE IN (ie, aux). Windows treats these 3 "sections" of the card as if they were 3 separate devices. The device name for each of these sections of my RAP-10 is "Roland Audio Producer". Windows ships with several audio/MIDI drivers for various cards and the INF files (and drivers) for those are included with Windows.

Note: Before installing new MIDI/Audio drivers, I recommend that you first manually remove any currently installed drivers that the new drivers are meant to replace, unless the new drivers are written to be installed over those old drivers. Usually, if you're updating new drivers over old drivers written by the same manufacturer for the same card, then there's no need to remove the old drivers first. But, if you had also changed your sound card, or gotten drivers from a different source, then I recommend removing the old drivers. For example, here's how I would go about changing to a different sound card and installing new drivers:

1. Boot up your system with its current setup (ie, with your current sound card and drivers).
2. Remove the current drivers. See "How do I remove an audio/MIDI driver?".
3. ShutDown Windows and turn off the computer.
4. Install the new sound card.
5. Boot up the system. If Windows doesn't autodetect the new card (which Windows won't if the card's drivers weren't included with Windows itself, or if the card isn't Plug-and-Play), then proceed to manually install the new drivers.

Here are the steps to install a MIDI/audio driver:

1. Start the Control Panel's **Add New Hardware** utility.
2. Click **Next** to skip the intro screen.
3. Answer **No** to Windows asking to search your hardware, and click **Next**.
4. Move down the list until you see **Sound, video, and game controllers**. Click on (ie, highlight) that item, and click *Next*. Windows will now present a dialog that lists all of the audio/MIDI drivers included with Windows. In the box on the right is listed all of the manufacturers whose drivers are included with Windows. When you select a manufacturer name, the box on the left updates to display those included drivers for that manufacturer's products.
5. If the driver that you want to install is among Window's included drivers, and you wish to use that, simply highlight the driver name, and select OK. Now skip to the last step.

   On the other hand, if you prefer to use drivers other than the included ones (ie, you have updated drivers), or the driver you need isn't included with Windows, then click the **Have Disk** button instead.
6. Type in the directory where the driver INF file can be found, and click **OK**.
7. If Windows finds the INF file, it will present a window containing the device name. Highlight

that name, and click **OK**.

Note: If the driver package contains drivers for various versions of windows, or various models of cards, then there may be numerous INF files in the package. You may have to search around, selecting various INF files, until you find one that lists the desired model.

8. Windows will then install the driver, and run whatever setup routine is contained within the driver. Typically, the setup routine will open a window and allow you to specify such things as port address, IRQ number, and DMA channels used for digital audio (if supported). Each driver has its own proprietary setup routine, so you'll have to deal with whatever prompt/window pops up at this point.

After you've installed the driver, you can now configure Windows' (and ultimately, all Windows applications') use of the driver. For example, do you want all Windows programs to send their MIDI output to this driver, and get their MIDI input from this driver? Do you want all programs to use the card's ADC (assuming it has such) to record digital audio (usually stored in a WAVE file)? Do you want all programs to use the card's DAC to play digital audio? You use the Control Panel's **MultiMedia** notebook to configure the card's use. Open the MultiMedia notebook. It has several pages that you can flip to (ie, the tabs are at the top of the notebook).

The first page is called **Audio**. This page concerns playing and recording digital audio. The page is divided in half, with the *Playback* settings on top, and the *Recording* settings below. If your card has a Digital to Analog Converter (ie, for playback) and Analog to Digital Converter (ie, for recording), then your card's device name should be listed in both the dropdown list in the Playback section of the page as well as the dropdown list in the Recording section. Select the device name so that it appears as the "Preferred Device" for both Playback and Recording. Now all Windows apps (as well as Windows itself when it plays "system sounds") will use your card for digital audio playing and recording. For cards that support various sample rates and bit resolutions, you should be able to select a "Preferred quality" for recording. You can also set default values for playback and recording volume. The "Use preferred devices only" setting is only effective if you have more than one digital audio device (and driver) installed. When selected, Windows forces all programs to use just the one "preferred device". When not selected, if some program has the preferred device in use, and another program wants to do something with digital audio, Windows will automatically have the second program use some other digital audio device in your system. Note that it's perfectly OK to have different cards chosen for playback and recording, which is one way to get around the problem of not having a full duplex sound card (ie, one that can record digital audio at the same time that it is playing back previously recorded digital audio).

The above settings don't mean that all programs must use the "preferred device" for digital audio, with those volume and rate/resolution settings. Rather, these are default values (for programs that don't offer ways to change such settings). On the other hand, a Windows program could query all of the digital audio devices installed on your system assuming that you had more than one installed), and let you choose to use any and all devices, maybe even simultaneously for many tracks of digital audio. The program could provide controls with which you adjust volume and rate/resolution, and maybe even additional parameters. But such a program would have to be written to use Windows' MCI API in a fancy manner. Your average CD-ROM Encyclopedia, and other titles that aren't geared for fancy audio work, generally use the preferred device and default settings verbatim.

There is another page in the MultiMedia notebook for **MIDI**. This page concerns MIDI input and output. If your card has MIDI input and output ports, then your card's device name should appear in the list below "Single Instrument". If you want to set all 16 MIDI channels to come in and go out through that card's MIDI IN and OUT, then select "Single Instrument" (ie, make sure that the dot is highlighted), and then highlight the device name (ie, so that it also appears in the box beneath "Single Instrument").

Note: In Windows NT/2000/XP, the MIDI page is missing. Instead, only the one, preferred instrument for MIDI playback (ie, output) is listed under the "Audio" page. There is no facility for configuring any patch remapping via the MultiMedia notebook.

On the other hand, if you have more than one MIDI device installed, and you wish to divide up the 16 MIDI channels between those devices, then select "Custom configuration" instead. For example, assume that you have two cards, an MPU-401 and a Monterrey. You've installed drivers for both, and so there are 2 device names in the MIDI device list. You want to have all MIDI events on channels 1 to 8 go through the MPU-401, and all MIDI events on channels 9 to 16 go through the Monterrey. Click the **Configure** button, and a window will pop open listing all 16 MIDI channel numbers. Now follow these steps to setup MIDI channel 1.

1. Highlight number 1.
2. Click the **Change** button. Another window will pop open containing a dropdown list with all of the installed device names. (In this example, that's the MPU-401 and Monterrey device names).
3. Select the desired device name, and click *OK*. (In this example, we want the MPU-401).

You've now set MIDI channel 1 to go through the MPU-401. Repeat the steps with channels 2 to 8. Repeat the steps for channels 9 to 16, but select the Monterrey device name instead. (Note that you can select several channels simultaneously for changing, by holding the *SHIFT* key while highlighting channel numbers).

Now, by default, any Windows program that transmits a MIDI message on channel 10 will send that message to the Monterrey's MIDI OUT.

For multi-port (ie, multiple bus) MIDI interfaces, the driver for that device should make it look like each bus on that one card is itself a separate card. Therefore, you should have a unique "device name" for each port on that multi-port interface. For example, maybe port 1 will have a device name of "Brand X MIDI Port 1", whereas port 2 will have a device name of "Brand X MIDI Port 2", etc). Hopefully, the driver will also be "multi-client" so that, if desired, you can have different programs simultaneously doing MIDI data transfers to different outputs. (In that case, you'd have to setup "custom configuration" and restrict each program to using different MIDI channels than the other programs).

Note that System Exclusive, as well as System Realtime (ie, MIDI Clock/Start/Stop, MIDI Time Code, etc) messages do not have a "MIDI channel" associated with them. I'm not sure how Windows

handles such events given a custom configuration. It may simply output such events to whatever device you've selected for "Single Instrument" or may automatically duplicate all System Realtime, Common, and Exclusive messages to all MIDI devices.

Just like with the Audio settings, the above MIDI settings don't mean that all programs **must** use the "Single Instrument" for MIDI. Rather, this is the default MIDI IN and OUT (for programs that don't offer ways to change such). On the other hand, a Windows program could query all of the MIDI devices installed on your system (assuming that you had more than one installed), and let you choose to use any and all devices, maybe even simultaneously for many MIDI channels (ie, 16 per device). The program could provide controls with which you adjust which MIDI data goes to which device. But such a program would have to be written to use Windows' MCI API in a fancy manner. In fact, CakeWalk Pro does exactly that, which is how it supports MIDI interfaces with multiple MIDI busses. Your average CD-ROM Encyclopedia, and other titles that aren't geared for fancy MIDI work, generally use either the "Single Instrument" (ie, one MIDI card) or for "Custom configuration", will automatically have the MIDI data routed to various cards as per your channel/device assignments.

How do I remove an audio/MIDI driver?

Never, never, never manually delete driver files by dragging icons to the Recycle Bin nor typing "del" from a command prompt. That's not the way that you uninstall a driver. There's a difference between uninstalling a driver and deleting files on your HD. If you do the latter, then you'll likely still leave behind references (to the now-missing driver files) in Windows' registry and INI files. This can cause all sorts of error messages to pop up. And if you later install other drivers without removing those references, you could end up with a completely confused set of error messages and weird behavior.

The proper way to uninstall a driver is as follows:

1. Open Control Panel's "System" notebook.
2. Flip to the "Device Manager" page. There you'll see a listing of all of the hardware in your computer, separated by category.

   Note: In Windows NT/2000/XP, Device Manager is located by flipping to the "Hardware" page and then clicking the "Device Manager" button.
3. You should see a category for sound and gameport devices with a small "+" sign next to it. Click on the + sign and this should drop down a list of the installed audio cards on your system. (Alternately, some sound cards list their driver components under their own category).
4. Highlight (ie, click on) the name of whichever component you wish to remove.
5. Click the "Properties" button.
6. Click the "Remove" button.

If your audio device isn't listed on System notebook's "Device Manager" page, then you'll have to open the MultiMedia notebook and flip to the "Advanced" page. This is like the Device Manager

listing, but it only shows MultiMedia components broken down into smaller catagories. You'll need to go through each sub-category (ie, "Audio Devices", "MIDI Devices and Instruments", etc) to remove each component that pertains to the device that you wish to remove.

I have an external MIDI module (ie, a Roland JV-80) that doesn't have a General MIDI patch set, nor drum note assignments per GM. Whenever I use the software mixer in my sequencer to select a patch, it always selects the wrong patch because it expects a GM patch set. In Win 3.1, I could use the MIDI Mapper to remap the GM patches and drum note assignments so that they selected the proper patches and drum sounds upon my JV-80. Where's the MIDI Mapper for Win95?

Microsoft didn't have time to finish rewriting the MIDI Mapper for Win95 before the OS's release. Subsequently, MS created the IDFedit.zip utility which you can download from my Web Site now. This utility can be used to create an Instrument Definition File (ie, IDF) in which you determine how the GM patch set and drum note assignments are remapped. Then, you install this IDF as if it were another MIDI device upon your system, and use it for MIDI IN and OUT.

For example, let's assume that the very first GM patch "Acoustic Grand Piano" is actually the second patch upon your JV-80 and you wish to remap GM patch 1 to JV-80 patch 2.

1. Run IDFedit.
2. Pull down the "Edit" menu and select "New instrument". This will create a new instrument with a patch set, drum note assignments, a "name" or "ID" (which defaults to "untitled1"), plus other settings. You should see "untitled1" appear in the list box.
3. Double-click upon "untitled1". This brings up the properties notebook with which you enter your settings.
4. First, you'll want to give this instrument a meaningful name. Flip to the "ID" page of the notebook, and type in your desired Instrument Name. For this example, we'll call it "JV-80 GM Patch Set".
5. Leave "Set as default" checked if you want Windows to apply these settings to the default

MIDI IN and OUT. (You can always change this later via the MultiMedia notebook's MIDI page. In fact, all this checkmark does is automatically select this new instrument as if you had picked it out of the list below "Single Instrument" on the MIDI page).

6. Now flip to the "Patch Map" page. Here you see a listing of all 128 GM patches. Highlight the "Acoustic Grand Piano", and click the "Edit" button. A dialog appears asking you to which patch to remap this GM Patch. Remember we said that the JV-80's Acoustic Grand Piano is #2, so type a 2 here, and press ENTER. You can then highlight other GM patches and remap them to equivalent patch numbers in your JV-80 patch set. You've now remapped GM to your JV-80. Whenever you use the "JV-80 GM Patch Set" as the output for your sequencer program, this remapping automatically occurs. (If you can't find a respective patch on your module for a given GM patch, then set the GM patch to the closest equivalent. For example, if you only have one piano sound whereas the GM set includes several piano variations, then set all of the GM piano patches to point to that one piano patch. ie, It's OK to set several GM patches to the same patch number).
7. You can flip to the "Percussion Map" page and reassign each GM drum note to a different MIDI note number in order to remap the drum note assignments.
8. Furthermore, you can flip to the "Key Map" page and remap all of the MIDI note numbers for the instrumental parts (ie, not the drum part, usually on channel 10). For example, if you add 12 to each note number in the list, all of the notes sent to the JV-80 will be transposed up an octave (except for the notes on the drum part's channel).
9. The "Channels" page allows you to select which channels are excluded from being affected by this IDF. This is useful if you want to apply different remapping (ie, different IDFs simultaneously) to different MIDI channels going through one MIDI card. For example, assume that you have a Proteus listening to MIDI channels 1 to 8, and a JV-80 listening to channels 9 to 16. These two units have non-GM patch sets, and are different from each other. They are both daisy-chained to one MIDI card in your computer. You'll want to create an IDF that remaps the Proteus patch set (and only affects channels 1 to 8), and another IDF that remaps the JV-80 patch set (and only affects channels 9 to 16). Then, you'll apply both IDFs to that one MIDI card. (In fact, you can create these two sets of IDF settings within one IDF file, by creating a "New Instrument" within IDFEdit, so that you actually have two instrument definitions with the one file. The second instrument will default to a name of "untitled2". That way you'll apply both remappings when you install just the one IDF, and this is more efficient).

   You can also choose which MIDI channel the "drum kit" is assigned to (which defaults to channel 10) in case you have a drum box set to a MIDI channel other than 10 and wish to apply the IDF drum note remapping to that MIDI channel.
10. The "Info" page simply lets you add more descriptive comments to your IDF.
11. You can click on the "OK" button when you're done with your settings. Then, pull down the "File" menu, select "Save As", and pick out a filename for your IDF.

Once you've saved your new IDF, you can "install" this new instrument.

1. Go to Control Panel and open the MultiMedia notebook.
2. Flip to the MIDI page.
3. Click on the "Add New Instrument" button. The "MIDI Instrument Installation Wizard" dialog appears.

4. Highlight the MIDI (hardware port) to which you want your IDF to be applied (ie, the device that you want to use for MIDI IN/OUT in conjunction with these IDF settings). If you only have one device, that is already highlighted.
5. Click on "Next".
6. Click on the "Browse" button and locate your new IDF. Select it. Its filename will now appear as one of the Instrument Definitions in the list.
7. Click "Next".
8. Now type in a descriptive name for this instrument. We'll just use "JV-80 GM Patch Set" again.
9. Click "Finish".

"JV-80 GM Patch Set" should now appear in the MultiMedia notebook's list of "Single Instruments". You can select this as if it were a device in your system, and when you do, Windows will apply the remapping to MIDI IN and OUT. This new "device" should also appear in CakeWalk's list of output devices.

You can add as many IDFs as you like, even applying more than one to a given MIDI card. (This may be useful if you have several non-GM multitimbral devices attached to a single MIDI interface. By using "Custom Configuration", and then selecting different IDFs for the various 16 MIDI channels, you can divide up the 16 MIDI channels between those instruments, as well as apply different remappings to each MIDI module. But if dealing with multitimbral instruments, you'll still have to set each instrument to actively ignore MIDI channels reserved for other modules).

If you wish to remove a particular IDF, you can do so. Let's remove our "JV-80 GM Patch Set" IDF.

1. Go to Control Panel and open the MultiMedia notebook.
2. Flip to the "Advanced" page (ie, the last page). This displays something that looks like Device Manager's list, except that it only contains devices pertaining to audio, MIDI, and video.
3. Click on the small + sign beside "MIDI Devices and Instruments". This will drop down a list of all of your installed MIDI devices.
4. There will another + sign next to the MIDI device which you've applied the IDF to. For example, let's assume that our IDF had been applied to the "Roland Audio Producer" device. Clicking on the + sign drops down a list of the IDFs that were applied to this device. You should see "JV-80 GM Patch Set" there.
5. Highlight this IDF.
6. Click on the "Properties" button at the bottom of the page. A new dialog should pop up containing information about this IDF.
7. Click on the "Remove" button. The IDF is no longer applied to the MIDI output, and has been removed from Win95's list of instruments. The actual IDF itself has not been erased. You can reinstall that file later if desired.

NOTE: Some sequencers support defining patch maps for each of your MIDI modules. (For more information, see MIDI connection and computer setup). This is more flexible than using an IDF because you can usually define more than 128 patches with the former (ie, the sequencer supports banks of patches whereas IDFs don't). Plus, the sequencer lets you apply any patch map to any track,

regardless of MIDI channel, so it's a lot easier to setup.

Just don't use an IDF in conjunction with the built-in patch naming features of your sequencer. You don't want to remap your custom patch sets. The exception to this is if your sequencer completely bypasses Win95's MultiMedia MIDI setup. Such a sequencer would use the "low level" MIDI API of Windows. In this case, you can still setup an IDF for the benefit of sequencers and multimedia software that uses the "high level" MCI API of Windows. But then, don't select the MIDI Mapper as the output driver for your software that has its own patch naming features.

I have both a Roland MPU-401 card and a Sound Blaster 16 card in my computer. When installing Windows, the "auto-detect" found my SB16 (and installed its drivers) but didn't find my MPU-401 card (nor install its drivers). Why did this happen, and how do I get my MPU-401 working?

Windows' "AutoDetect hardware" only does a search for one (non-PnP) sound card (since that's all that is really needed for normal use). As soon as it discovers one card, that's the end of the search for an audio card. Needless to say, the first card that Windows searches for is an SB compatible. I'm not surprised that it found your SB16 (and set that up) before finding your MPU-401.

Also, Windows is written to only check for those sound cards that have drivers that are shipped with Windows. If you have a card for which drivers were not included on your Windows CDROM (or floppies), Windows will not detect it (unless it's a PnP card. In that case, Windows will detect, but if there are no drivers, it will label it an "Unknown device" without any driver support).

In conclusion, Windows will not autodetect more than one non-PnP audio/MIDI card. For any second card, you'll have to manually install the drivers. Windows will also not autodetect any card that it doesn't know anything about. (ie, Windows will not autodetect a card for which drivers aren't included with Windows itself). You'll have to manually install the drivers for this case too. This is easy to do, and is explained in "How do I install/setup an audio/MIDI driver?".

When using my 16-bit Windows sequencer software under Windows, I get erratic digital audio

playback. Sometimes, there's static or tempo glitches. Why does this happen, and how can I fix it?

Most of the problems with old (16-bit) software playing digital audio tracks (ie, recording to, or playing from, the hard drive) are due to the differences between Win3.1's and Win95's disk cache handling and virtual memory. 16-bit Windows software is written for Win3.1, and consequently may not be designed to perform well with Win95's different methods. You can minimize adverse effects Win95 may have on 16-bit software by forcing the disk cache and virtual memory (ie, swap file on your HD) to fixed sizes.

To set Win95's disk cache to a fixed size, load the file SYSTEM.INI (found in your Windows directory) into NotePad. Find the line that reads [vcache]. Alter the MinFileCache and MaxFileCache lines below it (or add these 2 lines if they aren't already there). If you have a system with 8 MEG of RAM, set both to 1024. If you have a system with 16 MEG of RAM, set both to 2048. For example, here's what it would look like for 16 MEG:

[vcache]
MinFileCache=2048
MaxFileCache=2048

Resave the file.

This limits the RAM that Win95 uses for its disk cache. Normally, Win95 allocates all free RAM for that, which could interfere with memory allocations that your sequencer is trying to make during playback, and cause slowdowns in performance.

To set Win95's virtual memory (ie, swap file) to a fixed size, open Control Panel's System notebook and flip to the Performance page. Click on the "Virtual memory" button. Select "Let me specify my own virtual memory settings". For the Minimum setting, enter 0. For the Maximum setting, enter 20 if you have 8 MEG of RAM, or 28 if you have 16 MEG.

This prevents the swap file from growing and shrinking on your hard drive. It's also recommended that you defrag your drive afterwards. If you have more than one drive, try putting the swap file on a drive other than the one where you'll be recording digital audio.

After the above adjustments, reboot your system.

I can't get any sound from my MS-DOS game

programs when I play them under Win95 (ie, in a DOS prompt window or screen). I have to reboot the computer in "MS-DOS Mode" (ie, without Win95 loading at all) in order to get any sound. Why?

The problem is that this DOS prompt window or screen is really running under Win95 (ie, you're running an MS-DOS shell that uses Win95's "Current Configuration", rather than booting into a "New Configuration" without Win95 loaded and in control), and you have incomplete Win95 drivers for your sound card.

Here's the deal. When you boot into this DOS shell, Win95 is still loaded and in control (ie, you're running a Win95 DOS VDM). All of your Win95 drivers are managing access to your hardware, including your sound card. In MS-speak, you're using Win95's "Current Configuration". Your system isn't really rebooting into DOS so much as it's running DOS on top of Win95. (And that's why you can enter and exit from that DOS prompt, back to Win95, so quickly). Whether you select an MS-DOS Prompt from the Start Menu's Programs group, or Shutdown to MS-DOS Mode, Win95 is still loaded and in control (if you're using the "Current Configuration"). In fact, the Win95 drivers for your sound card are always in control of the card, unbeknowst to your MS-DOS program even. It's sort of like how a DOS "Terminate and Stay Resident" (TSR) utility can surreptiously wedge itself inbetween a program and some hardware.

There are 2 things that your Win95 drivers do. First, they manage interaction between Windows programs and your hardware. Secondly, they manage interaction between MS-DOS programs and your hardware... when Win95 is still in control. Your MS-DOS programs don't even know that Win95 (and its drivers) are still in control. Everytime that your MS-DOS program tries to execute an instruction that reads or writes to hardware, the Intel CPU (sneaky guy that it is) immediately "traps" that access and gives control over to Win95 and its drivers, who then manage the actual access. (In the process, Win95 also gets a chance to try to implement some crash protection, and perform multi-tasking).

Whereas your sound card's Win95 drivers may well support Windows apps access to hardware, it appears that those drivers don't properly support access by an MS-DOS program running under Win95. So, you can't run your MS-DOS stuff under Win95 (unless you get new Win95 drivers that support MS-DOS programs). What you're going to have to do is completely flush Win95 and its drivers out of memory, and reload MS-DOS so that it really is in control. That's exactly what happens when you boot into "MS-DOS Mode" without loading Win95. You can also choose to have your MS-DOS programs run under a "New Configuration" which causes Win95 to kill itself, and MS-DOS 7.0 (included with Win95) then loads.

You can setup that MS-DOS Prompt (or the Shutdown "Restart in MS-DOS Mode") to boot into a new configuration of MS-DOS only, instead of using Win95's current configuration (and leaving

Win95 still in control). The only problem is that, now that your DOS programs don't have Win95's drivers managing the hardware, for example Win95's drivers for your CDROM and mouse, you're going to need to copy some DOS drivers for your mouse and CD-ROM to your HD, and create CONFIG.SYS and AUTOEXEC.BAT files that load these drivers.

Here's how to setup your computer so that it boots into MS-DOS only:

1. Make sure that you copy MS-DOS drivers to your HD for any devices that need such, like your CD-ROM and mouse.
2. Open "Start -> Programs -> Windows Explorer".
3. Go to the "Windows" directory, and find the "Exit to Dos" icon. This holds the settings used when you boot into MS-DOS mode via "Start -> Programs -> MS-DOS Prompt" or Shutdown with "Restart in MS-DOS Mode".
4. Click once on it to open its pop-up menu, and select "Properties".
5. Go to the "Program" page.
6. Click on the "Advanced" button.
7. Select "Specify a new MS-DOS configuration" (instead of "Use current MS-DOS configuration"). Now the two boxes below will let you enter text. In the top box, you need to type out a CONFIG.SYS file exactly like it would have appeared if you only had MS-DOS on your system, and you wanted to load your DOS drivers. Of course, you need to specify everything. You need to load your mouse driver for MS-DOS. You need to load your CD-ROM driver for MS-DOS (probably using MSCDEX if you've got an IDE CD-ROM). You may need to copy these MS-DOS drivers from the disks shipped with your mouse and CD-ROM, to someplace on your HD. In short, you need to recreate CONFIG.SYS like it would be for MS-DOS, and copy any MS-DOS drivers for your mouse, CDROM, and sound card to your HD. For example, here's an example of the lines that I typed into the CONFIG.SYS box. I'm loading my mouse driver MOUSE.EXE, and my CD-ROM driver d011v110.sys. I'm not using any driver for my Vibra16 sound card (but the SB applets need an environment variable set).

   ```
   REM Stuff that you probably need
   DOS=HIGH,UMB
   Device=C:\WINDOWS\Himem.sys
   FILES=20
   REM Set this to the last drive on your system
   LastDriveHigh=E
   REM Here's my CD-ROM driver. Your driver may have a different name.
   REM It may require you to use "Device" instead of "DeviceHigh".
   REM Consult your CDROM manual.
   DeviceHigh=C:\d011v110.sys /d:mscd000 /n:1
   ```

   Now, in the lower box, you need to type the lines for an AUTOEXEC.BAT file. Here's what I use:

   ```
   REM A place where programs can toss garbage files. It must exist
   ```

```
SET TMP=C:\WINDOWS\TEMP
SET TEMP=C:\WINDOWS\TEMP
REM Just for the shell's prompt
SET PROMPT=&p$g
REM Where Win95 system files reside
SET winbootdir=C:\WINDOWS
REM Where MS-DOS can find system files
SET PATH=C:\WINDOWS;C:\WINDOWS\COMMAND
REM This is for the Sound Blaster Vibra16 PnP applets
SET BLASTER=A220 I5 D1 H5 P330 T6
REM Here's my mouse driver. Yours may have a different name.
C:\MOUSE.EXE
REM My CDROM is an IDE unit. Its driver uses MSCDEX to manage
REM the driver under MS-DOS.
MSCDEX.EXE /d:mscd000 /m:18
```

8. Now select "OK" on the dialogs to close them down and you're in business.

So what's happening here? What's the difference between "Specify a new MS-DOS configuration" and "Use current MS-DOS configuration"? As I explained, with the latter, Win95 doesn't reboot the computer. Win95 is still loaded and in control, using its Win95 drivers to manage your MS-DOS programs use of hardware. But, when you use a "new MS-DOS configuration", Win95 is completely flushed out of memory, and runs those CONFIG.SYS and AUTOEXEC.BAT scripts above. As you'll notice above, I've using my old MS-DOS drivers. So this is how you can boot into an environment that is the same as MS-DOS running your old drivers or TSRs, and yet, when you go back to Win95, you're back to using your new Win95 drivers (and the old real-mode drivers or TSRs are flushed from memory again -- which is good because, you don't want that stuff loaded with Win95 if you can avoid it). This is how you can have your MS-DOS games run without the interference of incomplete Win95 drivers that don't support sound for those MS-DOS programs, and yet still use those Win95 drivers with Windows programs which are supported.

The disadvantage is that, in order to completely flush Win95 and leave MS-DOS totally in control, Win95 has to reboot the computer. So, using "Specify a new MS-DOS configuration" will take much longer to switch to than "Use current MS-DOS configuration". But, it will allow you to use your old DOS TSRs with MS-DOS software, and your new Win95 drivers under Win95 (without resorting to always loading your old drivers/TSRs under Win95 and risking any weirdness between the two simultaneously loaded "old versus new" drivers). Plus, if you have Win95 drivers that don't properly support MS-DOS programs running under Win95, then you need to boot into MS-DOS mode anyway.

Incidentally, if you prefer to have only certain programs reboot into this "MS-DOS only" mode, then instead of performing the above procedure on the "Exit to Dos" shortcut, do this on the shortcut for the particular MS-DOS program(s) you desire. (ie, Click on the program's shortcut to bring up "Properties", and take it from there).

When I opened up Control's Panel's System notebook, and flipped to the Device Manager page (ie, to see a list of the hardware in my system), I noticed a small, yellow exclamation mark next to my sound card's name. What is this?

That exclamation mark you see next to a device name in Windows' System Notebook "Device Manager" page indicates a conflict (usually IRQ) or that something is wrong with the device's response. (ie, For example, I use SCSI devices rather than IDE, so I disabled my motherboard's IDE ports. My Plug and Play BIOS told Windows that I had IDE ports, but Windows didn't get any response from them, hence that exclamation mark next to my "PCI IDE controller".

In fact, it's common to see exclamation marks related to Plug And Play ISA cards. Often the devices appear to have several "phantom" units, each one dedicated to whatever IRQs the device supports, and Windows thinks that these are real yet gets no response from them. Hence you may see several instances of a particular Plug and Play ISA card listed, with all but one having an exclamation mark. It's OK to leave these phantom devices, and if Windows ever tells you it detected new hardware the next time you boot, tell Windows to ignore each one of those phantom devices as it pops up a dialog for each, asking you to choose a driver to install). Furthermore, for such phantom devices, you should highlight the name of each one in Device Manager, click on the Properties button, and remove the checkmark for "Original Configuration (current)". Leave only the real device (ie, the one whose IRQ and base is set to what your card really is using) enabled for "Original Configuration".

Furthermore, Device Manager can be used to identify possible hardware conflicts in your system. If you click upon the "Print" button, a dialog will pop up which allows you to print out a listing of what devices are using various IRQ, base I/O addresses, and DMA channels. (The dialog also offers an option, "print to file" to save this text listing in a file on your harddrive. The file will end in a .PRN extension. This is handy if you need to email your settings to tech support). If you look through this list and find a resource being used by more than one hardware device, that would indicate a likely hardware conflict. You should change the DIP switches or jumpers upon one of the devices (or if both devices are PnP, then try to use a software configuration utility to alter one of the device's resources. Better yet, make sure that you're using a PnP BIOS, and it's the latest version).

When I use Windows NT 4.0 (or Windows 2000 or Windows XP) Control Panel's "Add New Hardware", I can manually locate my sound card's INF file, and NT/2000/XP seems to install the

card's Windows 95 driver. But my card does not appear to be operating at all under Windows NT/2000/XP. Why?

Windows NT/2000/XP uses an entirely different driver model than Windows 95, 98, ME, or Windows 3.1. Windows NT/2000/XP has a 32-bit driver model whereas Windows 3.1 and Windows 95/98/ME still have some 16-bit components in their sound card drivers. You cannot use Windows 3.1 or Windows 95/98/ME drivers under Windows NT/2000/XP. You need a Windows NT/2000/XP driver for your sound card. Although Windows NT/2000/XP recognizes the INF file for a Win3.1 or Win95/98/ME driver (since WinNT/2000/XP drivers also use the INF file for installation purposes), it may appear that WinNT/2000/XP has installed the 16-bit driver, but it can't use such a driver. Typically, you'll get some sort of error message that the driver is not readable when you reboot your system

Recently, Microsoft has created a new driver model (Windows Driver Model, or WDM) which works with both the Windows ME and 2000 and XP operating systems. So a WDM driver written for Windows ME will should work under 2000/XP (and vice versa). Of course, you would need a new driver for your sound card if you wanted one of these WDM drivers. Otherwise, Win2000/XP continues to support current Windows NT drivers (so if you can't specifically find a Windows XP driver, then at least try to find a Windows 2000 or Windows NT driver), and Win98 continues to support Win3.1 and Win95 drivers, but neither will be able to use the others' current drivers.

Windows does not yet have a MIDI Manager. What this means is that it doesn't really arbitrate access to MIDI Input and Output intelligently between applications. Essentially, it leaves it up to each device driver to arbitrate access to its MIDI Input and Output (and without any standardized means for the driver to provide information about its status to the application). Needless to say, there are some caveats to this method (and general guidelines you should follow) as described below.

Single client drivers -- Device is busy errors

Many drivers are single client. Such a driver allows only one call to open its MIDI In to succeed (regardless of whether that Input is opened via midiInOpen(), or a high level MCI call). The driver also allows only one call to open its MIDI Out to succeed (regardless of whether that Output is opened via midiOutOpen(), a high level MCI call, or even midiStreamOpen()).

For example, if your application has successfully opened that driver for output (via a midiOutOpen call perhaps), then it owns that driver's MIDI Out. (ie, Your application has exclusive access to that MIDI Output). If some other application tries to subsequently open that Output (perhaps via midiOutOpen), then that application's call to midiOutOpen() will fail (and usually return an error number that is translated to a message of "Device is busy" by midiOutGetErrorText).

Only when your application finally closes that MIDI Output (via midiOutClose perhaps) will the "lock" upon that particular Output be released, and then another application can successfully open that Output.

Of course, the above example also pertains to MIDI Inputs (ie, midiInOpen, etc).

What is the result of this? Well, it means that, despite your application running under a multitasking system (ie, Win32), if your application opens a MIDI Input or Output and "hogs" it by keeping it open the entire time that your application is running, then the enduser will not be able to simultaneously run other MIDI software that wants to use that MIDI Input/Output. He'll just get "Device is busy" error messages from that other application every time he tries to do something with that application's MIDI Input/Output. You'll effectively be preventing him from multitasking another MIDI application in conjunction with that particular MIDI Input/Output. He'll have to close down your application every time he wants to use that other application with that MIDI Input/Output. And if the other application hogs that Input/Output too, then the reverse is also true. (ie, Every time he wants to run your application, he'll have to close down that other application).

This is very annoying to an enduser. It turns his multitasking operating system into something that looks like a singletasking operating system -- and one that doesn't manage resources.

So what is one way of getting around this? Well, a simple solution is not to hog the Input/Output. Don't open the MIDI Input/Output until you're actually ready to do some MIDI input/output (ie, midiOutShortMsg, midiOutLongMsg, doing some input/output via MCI, etc). Do that MIDI input/output as

quickly as possible, and then immediately close that MIDI Input/Output. The net result is that you'll only be preventing multitasking for a short time while you're actually doing some input/output. If the other applications are doing the same, then everyone will be happy, as the enduser can switch between those applications running concurrently, and as long as he doesn't try to make both applications do MIDI input/output simultaneously (which can be a bad thing as you'll learn below), his applications will be sharing the MIDI Input/Output.

For example, let's say that you're writing a patch editor for a sound module. You need to send it a System Exclusive message to request it perform a data dump. Then you need to input a bunch of System Exclusive messages until you receive the entire data dump. Here's the approach you could take:

1. Give the enduser a means to start the operation of receiving a dump. For example, maybe have a button labeled "Receive Dump" upon which he clicks to begin the operation. Do **not** open MIDI Input or Output until he clicks upon that button.
2. When he clicks upon the button (to start the data dump), open MIDI Out with a call to midiOutOpen(). Also open MIDI In with a call to midiInOpen() (and setup some means of collecting the incoming bytes that Windows will pass to your application).
3. Send your "dump request" System Exclusive message to MIDI Out.
4. Collect the incoming MIDI data that Windows passes to you.
5. When you see that you have a complete dump from the sound module, call midiOutClose() and midiInClose().

The above solution is all well and good for some applications, but it's an impractical solution for some other types of applications. Let's take the example of a "MIDI Input Viewer" program. It "displays" each incoming MIDI message in realtime (ie, immediately when the message comes into the MIDI In and is passed to the application by Windows).

Obviously, that application can't know ahead of time when MIDI messages will be forthcoming and how much data will be coming in (unlike the patch editor, which knows that a dump will be following shortly after that "dump request", and knows how many bytes to expect).

A slightly different approach could be to give the enduser a button to start recording (at which time you call midiInOpen) and a button to stop recording (at which time you call midiInClose). In this way, he could stop recording before he switched to another application, and therefore still have his applications share that MIDI Input. But there is an even better solution which makes sharing the Input/Outputs transparent to the enduser, as you'll learn later.

But as (bad) luck would have it, many programmers wrote MIDI software that doesn't even allow this minimal level of sharing. They wrote software that hogs the MIDI In or Out for the entire time that the program is running. Why? Well, sometimes it could be because the software allows access to numerous MIDI Inputs and Outputs simultaneously, and it takes too much time and trouble to open/close them all before/after each operation. If the software is a MIDI sequencer, such timing delays prior to recording/playing may be crucial. Therefore, the software simply leaves all inputs/outputs constantly open so that the software will be "ready to go" whenever the user starts it recording/playing. But this is still a poor excuse for why the software doesn't close those inputs/outputs when the enduser switches to some other application. This is easy to implement under Win32, as you'll learn later. After all, if the enduser is switching to another application, he probably doesn't really care whether the first application is ready to do input/output at that

moment (and if he does, there's a simple solution to handle that case too, which I'll outline later).

Multiclient drivers -- The perils of multitasking without management

Because too many programmers wrote MIDI applications that hog the inputs/outputs, some manufacturers of sound cards and MIDI interfaces started shipping multiclient drivers with their products.

A multiclient (sometimes called multi-instance) driver is one which allows more than one application to open a MIDI input or output. (ie, More than one call to midiOutOpen or midiInOpen will succeed). In essense, the driver lets two or more applications simultaneously open a MIDI input/output, unbeknowst to each other, and the driver itself tries to manage (ie, mix) their MIDI output, and also provides the applications separate copies of each incoming MIDI message.

Lots of endusers scramble for multiclient drivers so that they don't have to shut down MIDI programs in order to run other MIDI programs.

But there is one problem here. The two applications don't know that they're both outputting to the same MIDI port perhaps simultaneously. And the driver may not be intelligent enough to know when that's a good or bad thing. Let's take the example of that patch editor above.

Let's say that when the patch editor sends a data dump back to the sound module, instead of sending one large, complete system exclusive message in a single call to midiOutLongMsg(), it breaks one System Exclusive message into numerous "blocks of data bytes" sent via several calls to midiOutLongMsg(). Well, hopefully the driver is smart enough to not allow another application to do some MIDI Output inbetween the other application's midiOutLongMsg() calls until the driver detects the end of a System Exclusive message. Otherwise, the other application will screw up the patch editor's dump.

And Windows provides no standardized way for an application to tell a driver to give it exclusive access to a port, so an application is at the mercy of a multiclient driver's behavior in handling multiple output requests.

So too, if both programs are simultaneously outputting MIDI note messages, they may be turning off each other's Note On messages (with Note Off messages) or stacking Note Ons for the same note number and resulting in "stuck notes". Or they could be doing other conflicting things which the driver doesn't resolve.

And all of this multiclient stuff, particularly in regards to System Exclusive message input and output, imposes unnecessay overhead upon the driver. A single client driver can be **much** more stream-lined and efficient.

In fact, multiclient drivers decrease efficiency and yield poorer performance where a single client driver would suffice. (And those "loopback drivers" are even worse. These really play havoc with the efficiency of MIDI input/output). A MIDI Manager that arbitrated access to multiple ports, in conjunction with application support, would not only be more efficient (because all of its drivers could then be single client, instead of every driver duplicating the functionality and overhead of a poor "MIDI Manager" inside of it), it could also be more flexible (for example, allowing applications to choose when they want to share an Input/Output and when they don't -- control which they do not have over a multi-client driver).

Needless to say, a multiclient driver is no substitute for a MIDI Manager that is capable of allowing applications some control over sharing an Input/Output. Mostly, musicians use multiclient drivers for the sole purpose of avoiding having to shut down programs that hog the MIDI Input/Output. Clearly, what is needed is a better way for an application to access a MIDI Input/Output without hogging it, but also having to avoid the speed overhead of opening an Input/Output before every operation and then immediately closing it afterward (which as I pointed out above, is not suitable for some types of MIDI applications). If the enduser had software that could do that, instead of hogging an Input/Output, then he wouldn't need multiclient drivers at all. In fact, his software could share Inputs/Outputs with the multitude of single client drivers that are out there now.

Such a method is what we'll investigate now.

A more elegant, transparent way of sharing a MIDI input/output

If your application has a window of your own creation (and virtually all non-console Windows applications do), then a simple way of sharing a MIDI port with other applications, transparently to the enduser, is to process the WM_ACTIVATEAPP message. This message tells you whether your application is becoming active (ie, the user has selected any window that you created, and thereby given your program the focus) or inactive (ie, the user has selected a window created by another application, and therefore, if your application isn't already doing something in the background, it becomes inactive). In processing this message, you should close any MIDI ports you have open if the wParam argument is 0 (ie, you're going inactive) or reopen them if wParam is 1 (ie, you're active).

In this way, your application transparently releases the MIDI devices when the user switches away from it to another program. And your application also transparently reopens the MIDI devices when the user switches back to it. Because this processing is done in the WM_ACTIVATEAPP, it happens as soon as your application becomes active. This includes when your application initially starts, and so, the WM_ACTIVATEAPP code also serves to initially open your MIDI devices. Quick and easy.

What's so great about this? Well, you **can** hog the MIDI Input/Output to your hearts content... while your program is active (and the other MIDI programs are inactive). And when your program is inactive, you don't care about hogging the MIDI Input/Output. Now, you don't have to open/close the MIDI devices before/after each operation, and yet, you'll be sharing the MIDI ports with other applications -- even doing so transparently to the enduser, in a

way that works with single client drivers too. That's what is so great about it.

Below is some example code to illustrate this. It contains a minimal WinMain() function (ie, program entry point for a Windows C program) which opens a window of my own creation. (For simplicity, I use a dialog template in a resource file, and call CreateDialog, but you could use CreateWindowEx). Its message handler is mainWndProc(). That's where I open and close MIDI devices by processing the WM_ACTIVATEAPP message. mainWndProc() references two globals, MidiInHandle and MidiOutHandle, which are initially set to 0 at program startup. Whenever these two handles are 0, I know that I don't have my MIDI In and Out devices open. By checking these globals for 0, this makes the code bulletproof. I sort of made "wrapper functions" for midiOutOpen(), midiInOpen(), midiOutClose(), and midiInClose() that operate upon those global handles, just to make it easier.

```
DWORD MidiInHandle = 0;
DWORD MidiOutHandle = 0;

/* ****************************** WinMain()
*******************************
 * Program Entry point
 ***************************************************************************
*/

int WINAPI WinMain(HINSTANCE hinstExe, HINSTANCE hinstPrev, LPSTR
lpszCmdLine, int nCmdShow)
{
   MSG   msg;
   HWND  mainWindow;

   /* Create Main window */
   if (!(mainWindow = CreateDialog(hinstExe,
MAKEINTRESOURCE(IDD_MAINWINDOW), 0, mainWndProc)))
   {
      return(-1);
   }
```

```
    /* Show the window with default size */
    ShowWindow(mainWindow, SW_SHOWDEFAULT);
    UpdateWindow(mainWindow);

    /* Get the next msg (until WM_QUIT) */
    while (GetMessage(&msg, 0, 0, 0))
    {
       /* Send msg to window procedure */
       DispatchMessage((CONST MSG *)&msg);
    }

    /* Close any Midi Input device */
    closeMidiIn();

    /* Close any Midi Output device */
    closeMidiOut();

    /* Exit */
    return(0);
}

/* ******************************** mainWndProc()
*********************************
 * Main Window message handler called by Windows
 ************************************************************************************
*/

long APIENTRY mainWndProc(HWND hwnd, UINT uMsg, UINT wParam, LONG
```

```
lParam)
{
   switch(uMsg)
   {
      /* ============== App is gaining or losing activation
============= */
      case WM_ACTIVATEAPP:
      {
         /* Is app losing activation? */
         if (!wParam)
         {
            /* Close the MIDI devices (so other apps can use them)
*/
            closeMidiIn();
            closeMidiOut();
         }

         /* App gaining activation so reopen
         MIDI devices if not already open */
         else
         {
            /* Open MIDI In */
            openMidiIn();

            /* Open MIDI Out */
            openMidiOut();
         }

         /* Allow windows to continue handling WM_ACTIVATEAPP */
```

```
            break;
         }
      }

      // Let windows handle it
      return(0);
   }

   /* ****************************** closeMidiIn()
   ****************************
    * Close MIDI In Device if it's open.
    ************************************************************************
   */

   DWORD closeMidiIn(void)
   {
      DWORD   err;

      /* Is the device open? */
      if ((err = (DWORD)MidiInHandle))
      {
         /* Unqueue any buffers we added. If you don't
         input System Exclusive, you won't need this */
         midiInReset(MidiInHandle);

         /* Close device */
         if (!(err = midiInClose(MidiInHandle)))
         {
            /* Clear handle so that it's safe to call closeMidiIn()
```

```
anytime */
            MidiInHandle = 0;
        }
    }

    /* Return the error */
    return(err);
}

/* ************************** openMidiIn()
*****************************
 * Opens MIDI In Device #0. Stores handle in MidiInHandle. Starts
 * recording. (midiInputEvt is my callback to process input).
 * Returns 0 if success. Otherwise, an error number.
 * Use midiInGetErrorText to retrieve an error message.
 ***************************************************************************
*/

DWORD openMidiIn(void)
{
    DWORD   err;

    /* Is it not yet open? */
    if (!MidiInHandle)
    {
        /* Open MIDI Input and set Windows to call my
        midiInputEvt() callback function. You may prefer
        to have something other than CALLBACK_FUNCTION. Also,
        I open device 0. You may want to give the user a choice */
```

```
      if (!(err = midiInOpen(&MidiInHandle, 0,
(DWORD)midiInputEvt, 0, CALLBACK_FUNCTION)))
      {
         /* Start recording Midi and return if SUCCESS */
         if (!(err = midiInStart(MidiInHandle)))
         {
            return(0);
         }

      /* ============== ERROR ============== */

      /* Close MIDI In and zero handle */
      closeMidiIn();

      /* Return the error */
      return(err);
   }

   return(0);
}

/* ***************************** closeMidiOut()
*****************************
 * Close MIDI Out Device if it's open.
 ***************************************************************************
*/

DWORD closeMidiOut(void)
{
```

```
    DWORD   err;

    /* Is the device open? */
    if ((err = (DWORD)MidiOutHandle))
    {
        /* If you have any system exclusive buffers that
        you sent via midiOutLongMsg(), and which are still being
output,
        you may need to wait for their MIDIERR_STILLPLAYING flags to
be
        cleared before you close the device. Some drivers won't
close with
        pending output, and will give an error. */

        /* Close device */
        if (!(err = midiOutClose(MidiOutHandle)))
        {
            /* Clear handle so that it's safe to call closeMidiOut()
anytime */
            MidiOutHandle = 0;
        }
    }

    /* Return the error */
    return(err);
}

/* ************************** openMidiOut()
****************************
```

```
   * Opens MIDI Out Device #0. Stores handle in MidiOutHandle.
   * Returns 0 if success. Otherwise, an error number.
   * Use midiOutGetErrorText to retrieve an error message.
   ****************************************************************
  */

  DWORD openMidiOut(void)
  {
     DWORD   err;

     /* Is it not yet open? */
     if (!MidiOutHandle)
     {
        /* Open MIDI Output. I open device 0. You
        may want to give the user a choice */
        if (!(err = midiOutOpen(&MidiOutHandle, 0, 0, 0,
  CALLBACK_NULL)))
        {
           return(0);
        }

        /* ============== ERROR ============== */

        /* Return the error */
        return(err);
     }

     return(0);
  }
```

Note: For the sake of simplicity, I omit code above to notify the user if an error occurred in opening a MIDI device. That's usually something that he would want to know.

Of course, you would likely add menus and controls to your window, to allow the user to perform some MIDI operations. And you would add WM_COMMAND message processing to your window procedure to do that user interaction. One caveat is that, before you make any call to do MIDI output, you should first check that MidiOutHandle is not 0. After all, it may be that some other application has not been as nice as you, and didn't relinquish its death grip on the MIDI Output you're trying to use. In that case, the MIDI Output may not reopen. (Same thing with MIDI Input, but since Windows calls **you** for MIDI input, rather than you calling a Windows function, all that happens when MIDI Input doesn't reopen is that your MIDI input routines never get called).

There is one thing that I've noticed about some drivers. They allow an application to quickly return from a call to midiOutClose() or midiInClose(), but the driver then goes on to do some "cleanup" in the background. The driver may not yet be ready to allow someone else to do a midiInOpen() or midiInClose() during that time, and may return a "Device is busy" error. The net result is that, if another application is doing the same thing as above (ie, opening and closing devices in its WM_ACTIVATEAPP processing), there may not be enough time between when that application closes the device and when you try to open it. After all, you both are processing your WM_ACTIVATEAPP messages around the same time (on each of your timeslices).

I have found that a good work around is not to do the device opens while processing the WM_ACTIVATEAPP message. Rather, have that message post another, user-defined message to your window handler (ie, a message with an ID of WM_USER or greater). And then when your window procedure processes the WM_USER message is where you do the opens. This works because, when your application is getting the focus, there are usually a slew of messages sent and processed by your window procedure, and these get done well before you get around to that WM_USER message. Typically, a driver's background tasks get higher priority, so the driver will likely get done well before you get around to reopening the device.

Another concern I have observed under Windows 2000 (but not other versions of Win32) is when you go to close down your application. (ie, Close down all windows). You should close any MIDI handles you have before you call DestroyWindow() upon your last window. Otherwise, if you wait to close those handles after the call to DestroyWindow(), then the WM_ACTIVATEAPP message may be sent to another app which will try to open those same MIDI handles. (ie, During your call to DestroyWindow() upon your last open window, Windows 2000 will fire off the WM_ACTIVATEAPP message to some other app).

Here then is a slightly modified window procedure with these fixes:

```
/* ****************************** mainWndProc()
******************************
```

```
 * Main Window message handler called by Windows
 ************************************************************************
*/

LONG APIENTRY mainWndProc(HWND hwnd, UINT uMsg, UINT wParam, LONG
lParam)
{
   switch(uMsg)
   {
      /* ============== App is gaining or losing activation
============= */
      case WM_ACTIVATEAPP:
      {
         /* Is app losing activation? */
         if (!wParam)
         {
             /* Close the MIDI devices (so other apps can use
them) */
             closeMidiIn();
             closeMidiOut();
         }

         /* App gaining activation */
         else
         {
             /* Post a message to myself to reopen MIDI devices
             later, if they're not already open */
             PostMessage(hwnd, WM_USER, (WPARAM)0, (LPARAM)0);
```

```
            }

            /* Allow windows to continue handling WM_ACTIVATEAPP */
            break;
         }

         /* ================== MIDI Port allocation
 ==================== */
         /* I send a WM_USER message to myself during WM_ACTIVATEAPP
            to open the MIDI In and Out devices. */
         case WM_USER:
         {
            /* Open MIDI In */
            openMidiIn();

            /* Open MIDI Out */
            openMidiOut();

            /* Handled this message */
            return(1);
         }

         /* ============== App is about to close its last window
 ============= */
         case WM_CLOSE:
         {
            /* Close the MIDI devices (so other apps can use them) */
            closeMidiIn();
            closeMidiOut();
```

```
            /* NOW close your last window */
            DestroyWindow(hwnd);

            /* Handled this message */
            return(1);
         }
      }

      // Let windows handle it
      return(0);
}
```

Ok, you say, "But what about those times when I want to continue doing something with MIDI in the background while another program gets the focus?" For example, maybe you want to continue playing a MIDI file in the background, and don't want to stop just because the user switches to a paint program.

In that case, you should give the user the option of whether he wants your program to continue doing MIDI in the background. One good approach is to have some menu item he can check or uncheck which signals whether he wants you to "relinquish" the MIDI devices when your application is deactivated. Here I have a global, NoRelinquishFlag. When the user tells me to "hold onto the device", I set this to one. If the user wants me to relinquish the MIDI devices when he moves to another app, then I clear it. (I won't bother showing you how to set and clear variables). Below, I show you the simple modification you need to make to implement intelligent MIDI sharing that should encompass your software's requirements.

```
unsigned char NoRelinquishFlag = 0;

/* ****************************** mainWndProc()
*********************************
 * Main Window message handler called by Windows
 ************************************************************************
*/
```

```
LONG APIENTRY mainWndProc(HWND hwnd, UINT uMsg, UINT wParam, LONG
lParam)
{
   switch(uMsg)
   {
      /* ============== App is gaining or losing activation
============= */
      case WM_ACTIVATEAPP:
      {
         /* Is app losing activation? */
         if (!wParam)
         {
             /* Does user want the feature where this app
relinquishes its
             access to MIDI In and Out when user switches to
another app? */
             if (!NoRelinquishFlag)
             {
                /* Close the MIDI devices (so other apps can use
them) */
                closeMidiIn();
                closeMidiOut();
             }
         }

         /* App gaining activation */
         else
         {
```

```
                /* Post a message to myself to reopen MIDI devices
                later, if they're not already open */
                PostMessage(hwnd, WM_USER, (WPARAM)0, (LPARAM)0);
            }

            /* Allow windows to continue handling WM_ACTIVATEAPP */
            break;
        }

        /* ================== MIDI Port allocation
==================== */
        /* I send a WM_USER message to myself during WM_ACTIVATEAPP
            to open the MIDI In and Out devices. */
        case WM_USER:
        {
            /* Open MIDI In */
            openMidiIn();

            /* Open MIDI Out */
            openMidiOut();

            /* Handled this message */
            return(1);
        }
    }

    // Let windows handle it
    return(0);
}
```

Crashes under Windows 2000/NT/XP, and lost handles

When a program crashes, and it has no exception handling of its own, the Win32 default exception handling is invoked. This default handling attempts to free up resources for the program. For example, if you have some file open, the handle to it is closed. Under Windows 2000/NT/XP, the default exception handling unfortunately does not close any MIDI handles you have open. The net result is that, after a program crashes, no program will be able to open that MIDI port until you reboot the system.

But there is a way to solve this problem. What the default exception handling does do is to unload any Dynamic Link Libraries (DLL) that you have open. The operating system calls a special function (usually named DllMain()) in the DLL, passing a value of DLL_PROCESS_DETACH. The DLL is expected to do any cleanup then. So, if you place your MIDI handle variables inside of a DLL that you write, and then put code in your DllMain() function to close these handles if they are not zero, then your program will effectively close it MIDI handles, even when it crashes under Windows 2000/NT/XP.

So, how do you get access to those MIDI handles if they're inside of a DLL? Well, if you statically link with the DLL, and you've put the names of those variables in your Module Definition (.DEF) file, then Win32 automatically resolves those links when your program is loaded. You simply access those variables just like any other global variable in your program. But an even better idea is to put those above "wrapper functions" I wrote into a DLL, export their names in your DEF file, and call them just like any other functions in your program. Then, your DllMain()'s DLL_PROCESS_DETACH can simply call closeMidiOut() and closeMidiIn() to do the needed cleanup. You can download my MIDI Skeleton C example which implements this sharing. Run multiple copies of this program with a single client driver and you'll note that they all share a MIDI port. In fact, this example shows skeleton code for writing a MIDI C application, and may be of some use as a starting point for your own projects. Included are the Project Workspace files for Visual C++ 4.0. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames. It also includes the source code to a DLL with the wrapper functions in it. The C app uses this DLL to deal with the situation of freeing handles when crashing under Windows 2000/NT/XP.

Conclusion

By implementing the above, you'll be able to write a program that shares the MIDI ports with other such well-behaved MIDI programs (including multiple instances of your own program). It will work with single client drivers, and it will be transparent to the enduser. It will free MIDI handles under Windows 2000/NT/XP if your program crashes.

In conclusion, if you don't care about the overhead of opening and closing a device before and after every operation, and you do quick, MIDI operations, then you can just do the technique described in my first example. You won't hog a port that way.

But if you must hog a port, then do it the intelligent, user-friendly way as shown above.

Do Windows-using musicians a favor -- If you run across a MIDI program that doesn't share the MIDI port when you go to activate other MIDI programs, send the author of that first program a copy of this web page.

Using the Low level MIDI API, you need to first call midiOutOpen() or midiInOpen() to open some MIDI device for output or input respectively.

In order to write out MIDI data to a particular device, you need to first call midiOutOpen() once, passing it the Device ID of that desired device. Then, you can subsequently call a function such as midiOutShortMsg() which (immediately) outputs MIDI data to that device.

In order to read incoming MIDI data from a particular device, you need to first call midiInOpen() once, passing it the Device ID of that desired device. Then, Windows will subsequently pass your program each incoming MIDI message from that device.

After you're done inputting or outputting to a device (and have no further use for it), you must close that device.

Think of a MIDI device like a file. You open it, you read or write to it, and then you close it.

Opening the default MIDI device for output

How does your program choose a MIDI device for output? There are several different approaches you can take, depending upon how fancy and flexible you want your program to be.

Recall that Windows maintains separate lists of the devices which are capable of inputting MIDI data, and the devices capable of outputting MIDI data. For output, there is also the MIDI Mapper. It's not really a device in the MIDI Output list. (Ie, If you enumerate the devices in that list, you won't come across the MIDI Mapper). But it is there nonetheless. So what MIDI Output is the MIDI Mapper attached to? Well, that depends upon the settings that the user has made in Control Panel's Multimedia MIDI page. This Control Panel utility lets him route default MCI MIDI Output to a single MIDI Output (ie, one of the real MIDI Outputs in Windows' list). Or, he can use the "Custom configuration" setup (not available in Windows 2000/NT/XP) to split up the 16 MIDI channels among several of the real MIDI Outputs, for example, he could set all MIDI events on channel 1 to go to the built-in wavetable module on his Creative Labs sound card, and all MIDI events on channel 2 to go to the built-in wavetable on his Turtle Beach sound card. So when using the MIDI Mapper, although your program outputs to only one "device", it actually supports having the various MIDI channels going to different devices (which the user may desire for more polyphony or because some cards are better suited for certain sounds, etc). Plus, the "Add new Instrument" feature (not available in Windows earlier than Windows 95) allows the user to apply Instrument Definition Files thus remapping your program's MIDI output even more, for example, to make non-General MIDI instruments conform to General MIDI.

When you use the MIDI Mapper for output, think of the Multimedia utility's "MIDI" page as becoming the "MIDI Setup" dialog for your own application. Whichever way the user set that page up is where your calls to midiOutShortMsg() and midiOutLongMsg() get routed. So, by opening the MIDI Mapper, you use the "default MIDI Out" setup.

The MIDI Mapper has a defined Device ID of -1, so to open MIDI Mapper for MIDI Output:

```
unsigned long result;
HMIDIOUT      outHandle;

/* Open the MIDI Mapper */
result = midiOutOpen(&outHandle, (UINT)-1, 0, 0, CALLBACK_WINDOW);
if (result)
{
   printf("There was an error opening MIDI Mapper!\r\n");
```

```
}
```

One drawback with MIDI Mapper is that it does impose an extra layer of software processing upon your MIDI output. If the user never enables the "Custom configuration", then all MIDI data ends up going to one device anyway, so you gain nothing here (and lose a little efficiency).

Opening the default MIDI device for input

OK, that works for MIDI output, but what about MIDI input? There is no MIDI Mapper for input. You have to open one of the real devices. Remember that the first device in each list has a Device ID of 0. If he has at least one device capable of MIDI input, then you at least have a device with ID #0. So, you can simply use a Device ID of 0 with midiInOpen() as so:

```
unsigned long result;
HMIDIIN      inHandle;

/* Open the MIDI In device #0. Note: myWindow is a handle to some open window */
result = midiInOpen(&inHandle, 0, (DWORD)myWindow, 0, CALLBACK_WINDOW);
if (result)
{
   printf("There was an error opening the default MIDI In device!\r\n");
}
```

Of course, if the user has no device installed capable of inputting MIDI data, the above call returns an error, so always check that return value.

Note that the device(s) opened for output via MIDI Mapper, and the device opened for input above, may or may not be components of the same card. In other words, whichever MIDI IN jack is the default MIDI Input, and whichever MIDI OUT jack is the default MIDI Output, could be on two entirely different cards. But that is irrelevant to your purposes).

The MIDI Input device with an ID of 0 is whichever MIDI device happened to first get into the list of MIDI Input devices upon bootup. The user really has no control over setting this.

The most flexible way to choose a MIDI device for input or output

The most flexible way would be to present the user with all of the names in the list of MIDI Output devices and let him choose which one he wants (or if your program supports multiple MIDI output devices, you may wish to let him pick out several names from the list, and assign each sequencer "track" to one of those Device IDs. This is how professional sequencers implement support for multiple cards/outputs).

Whereas Windows maintains separate lists of MIDI Input and Output devices, so too, Windows has separate functions for querying the devices in each list.

Windows has a function that you can call to determine how many device names are in the list of devices that support outputting or playing MIDI data. This function is called midiOutGetNumDevs(). This returns the number of devices in the list. Remember that the Device IDs start with 0 and increment. So if Windows says that there are 3 devices in the list, then you know that their Device IDs are 0, 1, and 2 respectively. You then use these Device IDs with other

Windows functions. For example, there is a function you can call to get information about one of the devices in the list, for example its name, and what sort of other features it has. You pass the Device ID of the device which you want to get information about (as well as a pointer to a special structure called a MIDIOUTCAPS into which Windows puts the info about the device), The name of the function to get information about a particular MIDI Output device is midiOutGetDevCaps().

Here then is an example of going through the list of MIDI Output devices, and printing the name of each one:

```
MIDIOUTCAPS     moc;
unsigned long   iNumDevs, i;

/* Get the number of MIDI Out devices in this computer */
iNumDevs = midiOutGetNumDevs();

/* Go through all of those devices, displaying their names */
for (i = 0; i < iNumDevs; i++)
{
    /* Get info about the next device */
    if (!midiOutGetDevCaps(i, &moc, sizeof(MIDIOUTCAPS)))
    {
        /* Display its Device ID and name */
        printf("Device ID #%u: %s\r\n", i, moc.szPname);
    }
}
```

Likewise with MIDI Input devices, Windows has a function that you can call to determine how many device names are in the list of devices that support inputting or creating MIDI data. This function is called midiInGetNumDevs(). This returns the number of devices in the list. Again, the Device IDs start with 0 and increment. There is a function you can call to get information about one of the devices in the list, for example its name, and what sort of other features it has. You pass the Device ID of the device which you want to get information about (as well as a pointer to a special structure called a MIDIINCAPS into which Windows puts the info about the device), The name of the function to get information about a particular MIDI Input device is midiInGetDevCaps().

Here then is an example of going through the list of MIDI Input devices, and printing the name of each one:

```
MIDIINCAPS     mic;
unsigned long    iNumDevs, i;

/* Get the number of MIDI In devices in this computer */
iNumDevs = midiInGetNumDevs();

/* Go through all of those devices, displaying their names */
for (i = 0; i < iNumDevs; i++)
{
    /* Get info about the next device */
    if (!midiInGetDevCaps(i, &mic, sizeof(MIDIINCAPS)))
    {
        /* Display its Device ID and name */
        printf("Device ID #%u: %s\r\n", i, mic.szPname);
    }
}
```

You can download my ListMidiDevs C example to show how to print the names of all the installed MIDI Input and Output devices, as well as other info about each device. Included are the Project Workspace files for Visual C++ 4.0,

but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

Outputting MIDI data (except System Exclusive)

How does an application tell Windows to output some MIDI bytes? That depends upon whether you're outputting System Exclusive Messages, or some other kind of MIDI message. All MIDI messages, except for System Exclusive, always have 3 or less bytes. So, Windows has a function through which you can pass such a MIDI message in its entirety for output. This function is called **midiOutShortMsg()**. What you do is pack up the 3 or less bytes of that MIDI message as an unsigned long value, and pass it to midiOutShortMsg(). The bytes of this MIDI message then get output as soon as possible (ie, hopefully immediately).

With midiOutShortMsg(), you need to pack up these 3 bytes into one unsigned long which is passed as one arg. The LSB of the low word is the MIDI status (for example, 0x90 for MIDI channel 1). The MSB of the low word is the first MIDI data byte, if any. (For Note events, this would be the MIDI note number). The LSB of the high word is the second MIDI data byte, if any. (For Note events, this would be the note velocity). The MSB of the high word is not used.

Note: Always include a status byte. The device driver for the card will implement running status when it outputs the MIDI message.

Let's take an example of playing a 3 note chord -- a C chord (ie, C, E, and G notes).

Each musical pitch of a chord is expressed as a MIDI note number (middle C is note number 60, so D# above middle C is #61, etc). We'll create a MIDI message for each one of those 3 note numbers. A MIDI message takes the form of 3 bytes; the Status byte, the note number, and velocity (usually implements note volume). The Status byte for turning a note on is 0x9**X** where X is the MIDI channel number desired (0 to F for MIDI channels 1 to 16 -- we'll use a default of 0 but you may want to allow the user to change this). So for MIDI channel 1, the status is always 0x90. For velocity, we'll use a default of 0x40.

```
HMIDIOUT    handle;

/* Open default MIDI Out device */
if (!midiOutOpen(&handle, (UINT)-1, 0, 0, CALLBACK_NULL) )
{
    /* Output the C note (ie, sound the note) */
    midiOutShortMsg(handle, 0x00403C90);

    /* Output the E note */
    midiOutShortMsg(handle, 0x00404090);

    /* Output the G note */
    midiOutShortMsg(handle, 0x00404390);

    /* Here you should insert a delay so that you can hear the notes sounding */
    Sleep(1000);

    /* Now let's turn off those 3 notes */
    midiOutShortMsg(handle, 0x00003C90);
    midiOutShortMsg(handle, 0x00004090);
```

```
    midiOutShortMsg(handle, 0x00004390);

    /* Close the MIDI device */
    midiOutClose(handle);
}
```

Note: If your application is doing some sort of sequencing (ie, playback of a musical piece), you'll have to maintain a timer in order to figure out when it's time to output the next MIDI message via midiOutShortMsg(). (Note that 32-bit Windows MultiMedia Timer callbacks under Win95 may suffer severe timing fluctuations. Since Win95's multimedia system is still 16-bit, you need to put your timer callback (and any functions it calls) into a 16-bit DLL in order for it to exhibit solid performance under Win95. WinNT doesn't exhibit this aberrant behavior with 32-bit code).

Warning: There are some badly written drivers out there, especially for Windows NT/2000/XP. midiOutShortMsg() doesn't actually output the MIDI message. Instead, the driver puts the message in some internal queue for another thread (running inside of the driver) to output later, and then midiOutShortMsg() returns immediately. There's nothing wrong with that per se, but in these badly written drivers, when you call midiOutClose(), the driver discards any MIDI messages in its queue (as opposed to properly flushing them to MIDI Out). So if you call midiOutClose() too soon after calling midiOutShortMsg() (or perhaps even midiOutLongMsg()), then your MIDI output may end up being discarded before that other thread inside of the driver gets a chance to output the message. If the above example code results in stuck notes (or no sound at all) unless you put another Sleep() before the call to midiOutClose(), then you're dealing with such a badly designed driver. You could try to write some workaround that, whenever to call midiOutShort(), set a flag variable and start a one-shot timer. At the end of the timer, clear the flag. Before you ever call midiOutClose(), make sure that this flag is clear. If not, postpone the close. But a better solution is to contact the author of your card's driver, and tell him to fix his badly designed code.

You can download my Twinkle C example to show how to use midiOutShortMsg to play MIDI notes on the default MIDI Out device. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it.

Outputting System Exclusive MIDI messages

Since System Exclusive messages can be any length, Windows has a different means for outputting them. You use the function midiOutLongMsg(), passing it a buffer filled with the MIDI message. (In fact, you can use midiOutLongMsg to pass a buffer filled with several non-System Exclusive messages. This is very handy if you need to output several MIDI messages that should occur simultaneously. Using midiOutShortMsg() for each individual MIDI message as we did in the example above may cause too much of a delay inbetween each MIDI message). You actually pass a special structure called a MIDIHDR which has a field where you store the pointer to your buffer containing the MIDI data.

But there are a few caveats:

1. If running Win3.1, the data buffer (and perhaps the MIDIHDR structure, although MS examples show otherwise) must be allocated with GlobalAlloc() using the GMEM_MOVEABLE and GMEM_SHARE flags, and locked with GlobalLock. Under Win95 and WinNT, this no longer appears to be a requirement.

2. Before you pass the buffer to midiOutLongMsg(), you must first "prepare" it by calling midiOutPrepareHeader().

3. The MIDI Output device's driver determines whether the data is sent synchronously or asynchronously. So, with some devices, your app won't return from the call to midiOutLongMsg() until all of the data is output, whereas with other devices, you may return immediately and the driver will continue outputting the data in the background.

4. After you're done with the buffer, you must "unprepare" it by calling midiOutUnprepareHeader().

Here's an example of outputting a System exclusive message under Win3.1:

```
HMIDIOUT    handle;
MIDIHDR     midiHdr;
HANDLE      hBuffer;
UINT        err;
char            sysEx[] = {0xF0, 0x7F, 0x7F, 0x04, 0x01, 0x7F, 0x7F, 0xF7};

/* Open default MIDI Out device */
if (!midiOutOpen(&handle, (UINT)-1, 0, 0, CALLBACK_NULL))
{
    /* Allocate a buffer for the System Exclusive data */
    hBuffer = GlobalAlloc(GHND, sizeof(sysEx));
    if (hBuffer)
    {
        /* Lock that buffer and store pointer in MIDIHDR */
        midiHdr.lpData = (LPBYTE)GlobalLock(hBuffer);
        if (midiHdr.lpData)
        {
            /* Store its size in the MIDIHDR */
            midiHdr.dwBufferLength = sizeof(sysEx);

            /* Flags must be set to 0 */
            midiHdr.dwFlags = 0;

            /* Prepare the buffer and MIDIHDR */
            err = midiOutPrepareHeader(handle,  &midiHdr, sizeof(MIDIHDR));
            if (!err)
            {
                /* Copy the SysEx message to the buffer */
                memcpy(midiHdr.lpData, &sysEx[0], sizeof(sysEx));

                /* Output the SysEx message */
                err = midiOutLongMsg(handle, &midiHdr, sizeof(MIDIHDR));
                if (err)
                {
                    char   errMsg[120];

                    midiOutGetErrorText(err, &errMsg[0], 120);
                    printf("Error: %s\r\n", &errMsg[0]);
                }

                /* Unprepare the buffer and MIDIHDR */
                while (MIDIERR_STILLPLAYING == midiOutUnprepareHeader(handle,
&midiHdr, sizeof(MIDIHDR)))
                {
                    /* Should put a delay in here rather than a busy-wait */
                }
            }

            /* Unlock the buffer */
            GlobalUnlock(hBuffer);
        }
```

```
        /* Free the buffer */
        GlobalFree(hBuffer);
    }

    /* Close the MIDI device */
    midiOutClose(handle);
}
```

Win95 and WinNT are easier. Here's an example to output a System Exclusive message under Win95/NT:

```
HMIDIOUT    handle;
MIDIHDR     midiHdr;
UINT        err;
char            sysEx[] = {0xF0, 0x7F, 0x7F, 0x04, 0x01, 0x7F, 0x7F, 0xF7};

/* Open default MIDI Out device */
if (!midiOutOpen(&handle, (UINT)-1, 0, 0, CALLBACK_NULL))
{
    /* Store pointer in MIDIHDR */
    midiHdr.lpData = (LPBYTE)&sysEx[0];

    /* Store its size in the MIDIHDR */
    midiHdr.dwBufferLength = sizeof(sysEx);

    /* Flags must be set to 0 */
    midiHdr.dwFlags = 0;

    /* Prepare the buffer and MIDIHDR */
    err = midiOutPrepareHeader(handle,  &midiHdr, sizeof(MIDIHDR));
    if (!err)
    {
        /* Output the SysEx message */
        err = midiOutLongMsg(handle, &midiHdr, sizeof(MIDIHDR));
        if (err)
        {
            char   errMsg[120];

            midiOutGetErrorText(err, &errMsg[0], 120);
            printf("Error: %s\r\n", &errMsg[0]);
        }

        /* Unprepare the buffer and MIDIHDR */
        while (MIDIERR_STILLPLAYING == midiOutUnprepareHeader(handle, &midiHdr,
sizeof(MIDIHDR)))
        {
            /* Should put a delay in here rather than a busy-wait */
        }
    }

    /* Close the MIDI device */
    midiOutClose(handle);
}
```

You can download my MidiVol C example to show how to use midiOutLongMsg to output a MIDI System Exclusive

message on the default MIDI Out device. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it.

Inputting MIDI data

In the examples of outputting MIDI data, you'll notice that in the call to midiOutOpen(), I specified CALLBACK_NULL. What this flag means is that (other than the return codes from functions such as midiOutShortMsg or midiOutLongMsg), I require no feedback from Windows on the progress of the output of that MIDI data. (Remember that these output functions may return to the app before the data is finished being sent, if the driver has some means of doing that output in the background). I could have asked Windows to provide me with feedback (as we'll do in our MIDI input examples), but it wasn't necessary in those output examples, and it's simpler not to do it (although you should do it if you've called midiOutLongMsg -- in order to know when it's time to call midiOutUnprepareHeader. Otherwise, you'd have to do polling on the MHDR_DONE bit of the dwFlags field of the MIDIHDR structure. This bit will be set when the MIDI driver is finished with the MIDIHDR).

With MIDI input, you must provide a means of Windows giving you some sort of feedback. Why? Because you can't keep continuously polling the MIDI In port of a MIDI Input device waiting for incoming MIDI messages. That's an outdated MS-DOS programming technique. Rather, Windows programs are supposed to relinquish control back to Windows when the program has nothing to do except wait for something to happen, (Although it's possible, albeit not good programming practice, to do polling of the MHDR_DONE bit when inputting System Exclusive messages, for input of other types of MIDI messages, Windows requires that you provide a different means for Windows to pass you MIDI data).

Instead, Windows will interact with your program whenever each MIDI message is input (ie, all of the bytes in it, for non-System Exclusive messages) or some input buffer you've supplied is filled (for System Exclusive messages). How does Windows interact with your program? You have several options as follows:

1. CALLBACK_EVENT -- You allocate some Event with CreateEvent(), and Windows uses this to signal your app. (ie, Your app can wait on that Event signal, for example with WaitForSingleObject). You pass the handle of the Event as the 3rd arg to midiInOpen().

2. CALLBACK_THREAD -- Windows causes some suspended thread within your app to run. (ie, Your app's thread can suspend itself via SuspendThread). You pass the Thread ID of the desired thread to be run as the 3rd arg to midiInOpen().

3. CALLBACK_WINDOW -- Windows sends a message to some open window in your app. The parameters for the message will contain additional information about what caused Windows to send that message. You pass the desired window's handle as the 3rd arg to midiInOpen().

4. CALLBACK_FUNCTION -- Windows directly calls some function in your app. It passes args that contain additional information about what caused Windows to call your function. You pass a pointer to the desired function as the 3rd arg to midiInOpen(). The 4th arg to midiInOpen() can be anything you desire, and this will be passed to your callback function each time that Windows calls your callback.

The latter two methods allow you to better determine what exactly caused Windows to notify you, because they supply additional information to you. In fact, for regular MIDI messages (ie, everything except System Exclusive messages -- I'll simply refer to these as "regular messages"), the latter two methods are the only methods you can use to actually get Windows to pass you the incoming MIDI data. (For System Exclusive, you could use the CALLBACK_EVENT or CALLBACK_THREAD, if you're not really interested in being notified of errors). For this reason, I'll only detail the

latter two methods.

So when does Windows notify you? Here are the times when Windows notifies you:

1. When you open a MIDI In device via midiInOpen().
2. When you close a MIDI In Device via midiInClose().
3. When Windows has finished inputting one regular MIDI message in its entirety.
4. When Windows has filled a MIDIHDR's buffer with a portion, or all, of a System Exclusive message.
5. When there has been an error inputting a regular or System Exclusive MIDI message.
6. When your handling of a particular MIDI message (that you've been passed) is so slow that the MIDI driver (and possibly the MIDI Interface) has had to throw away incoming MIDI data while it was waiting for you to finish processing a previous message.

In conclusion, you can have Windows call a function you have written, passing the MIDI data that has been input, or you can have Windows pass a message to one of your program's windows, with the MIDI data that has been input as part of that message.

To have Windows call a function you have written, when you open the device, you specify the flag CALLBACK_FUNCTION. The third arg is a pointer to your function. (The fourth arg can be any value that you want Windows to pass to your function each time that function is called).

```
result = midiInOpen(&inHandle, 0, (DWORD)myFunc, 0, CALLBACK_FUNCTION);
```

To have Windows pass a message to one of your windows, when you open the device, you specify the flag CALLBACK_WINDOW. The third arg is a handle to your window. (The fourth arg is not used).

```
result = midiInOpen(&inHandle, 0, (DWORD)myWindow, 0, CALLBACK_WINDOW);
```

One caveat with this second method is that Windows doesn't timestamp each MIDI message. So if you need timestamps, you would have to timestamp each MIDI message yourself using some software timer (ie, perhaps the one that you're using to time the output of MIDI messages, for example, a Windows MultiMedia timer implemented using functions such as timeGetTime). But such message passing is not very efficient. By the time that your window procedure finally got around to pulling that MIDI data out of a message and obtaining a time stamp for it, a long come could have transpired since it actually arrived at the computer's MIDI IN. Trying to get an accurate time stamp using this method is very difficult, especially if other things are happening in the system such as mouse and window movement. It is recommended that you use CALLBACK_WINDOW only when you don't need time stamps. Otherwise, use the CALLBACK_FUNCTION method.

If using the CALLBACK_FUNCTION method, then you need to write a function that has the following declaration (although you can name the function anything you like):

```
void CALLBACK midiCallback(HMIDIIN handle, UINT uMsg, DWORD dwInstance, DWORD
dwParam1, DWORD dwParam2);
```

As mentioned, you pass a pointer to this function as the 3rd arg to midiInOpen(). The 4th arg to midiInOpen() can be anything you desire, and this will be passed to your callback function (as the dwInstance arg) each time that Windows

calls your callback. The handle arg is what was returned from midiInOpen().

The other args may be interpreted differently depending upon why Windows has called your callback. Here are those reasons:

1. You open a MIDI In Device via midiInOpen(). In this case, the uMsg arg to your callback will be MIM_OPEN.

2. You close a MIDI In Device via midiInClose(). In this case, the uMsg arg to your callback will be MIM_CLOSE.

3. One, regular (ie, everything except System Exclusive messages) MIDI message has been completely input. In this case, the uMsg arg to your callback will be MIM_DATA. The dwParam1 arg is the bytes of the MIDI Message packed into an unsigned long in the same format that is used by midiOutShort(). The dwParam2 arg is a time stamp that the device driver created when it recorded the MIDI message.

4. Windows has either completely filled a MIDIHDR's memory buffer with part of a System Exclusive message (in which case you had better continue queuing the MIDIHDR again in order to grab the remainder of the System Exclusive), or the MIDIHDR's memory buffer contains the remainder of a System Exclusive message (or the whole message if it happened to fit into the memory buffer intact). In this case, the uMsg arg to your callback will be MIM_LONGDATA. The dwParam1 arg is a pointer to the MIDIHDR whose memory buffer contains the System Exclusive data. The dwParam2 arg is a time stamp that the device driver created when it recorded the MIDI message.

5. Your callback is not processing data fast enough such that the MIDI driver (and possibly the MIDI In port itself) has had to throw away some incoming, regular MIDI messages. In this case, the uMsg arg to your callback will be MIM_MOREDATA. The dwParam1 arg is the bytes of the MIDI Message that was not handled (by an MIM_DATA call) packed into an unsigned long in the same format that is used by midiOutShort(). The dwParam2 arg is a time stamp that the device driver created when it recorded the MIDI message. In handling a series of these events, you should store the MIDI data in a global buffer, until such time as you receive another MIM_DATA (which indicates that you can now do the more time-consuming processing that you obviously were doing in handling MIM_DATA). In other words, when Windows calls your callback with MIM_MOREDATA, this is it's way of saying "You're handling your previous MIM_DATA messages too slowly. (And in fact, I may have preemptively interrupted a previous MIM_DATA handling of your callback). This is your last chance to quickly do something with this one message that I'm passing you now. Otherwise, you're so far behind in handling the MIDI input that data is about to be permanently lost".

   NOTE: Windows sends an MIM_MOREDATA event only if you specify the MIDI_IO_STATUS flag to midiInOpen().

6. An invalid, regular MIDI message was received. In this case, the uMsg arg to your callback will be MIM_ERROR. The dwParam1 arg is the bytes of the MIDI Message that was not handled (by an MIM_DATA call) packed into an unsigned long in the same format that is used by midiOutShort(). The dwParam2 arg is a time stamp that the device driver created when it recorded the MIDI message.

7. An invalid, System Exclusive message was received. In this case, the uMsg arg to your callback will be MIM_LONGERROR. The dwParam1 arg is a pointer to the MIDIHDR whose memory buffer contains the System Exclusive data. The dwParam2 arg is a time stamp that the device driver created when it recorded the MIDI message.

MIDI time stamps are defined as the time the first byte of the message was received and are specified in milliseconds. The midiInStart() function resets the time stamps for a device to 0.

You can download my Midi In Callback C example to show how to input MIDI messages. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it.

Using the Low level Digital Audio API, you need to first call waveOutOpen() or waveInOpen() to open some Digital Audio device for output (use its Digital to Analog Converter to play audio) or input (use its Analog to Digital Converter to record audio) respectively.

In order to write out Digital Audio data to a particular device's DAC, you need to first call waveOutOpen() once, passing it the Device ID of that desired device. Then, you can subsequently call a function to send blocks of Digital Audio data to that device's DAC. One of the other parameters you pass is a pointer to a WAVEFORMATEX structure. You fill in the fields of this structure (prior to calling waveOutOpen) to tell the device such things as the sample rate and bit resolution of the digital audio data you intend to play, as well as whether it is Mono (1 channel) or stereo (2 channels).

In order to read incoming Digital Audio data from a particular device's ADC, you need to first call waveInOpen() once, passing it the Device ID of that desired device. Then, Windows will subsequently pass your program blocks of incoming Digital Audio from that device's ADC. One of the other parameters you pass is a pointer to a WAVEFORMATEX structure. You fill in the fields of this structure (prior to calling waveInOpen) to tell the device such things what sample rate and bit resolution to use when recording the digital audio data, as well as whether to record in Mono (1 channel) or stereo (2 channels).

After you're done recording or playing Digital Audio on a device (and have no further use for it), you must close that device.

Think of a Digital Audio device like a file. You open it, you read or write to it, and then you close it.

Easy way to choose a Digital Audio device for input or output

How does your program choose a Digital Audio device for input or output? There are several different approaches you can take, depending upon how fancy and flexible you want your program to be.

Recall that Windows maintains separate lists of the devices which are capable of recording Digital Audio data, and the devices capable of playing Digital Audio data. Pass the value WAVE_MAPPER as the Device ID to open the "preferred" Digital Audio Input device and Digital Audio Output device respectively. So, if you simply want to open the preferred Digital Audio Output device, then use a Device ID of WAVE_MAPPER with waveOutOpen() as so:

```
unsigned long result;
HWAVEOUT      outHandle;
WAVEFORMATEX  waveFormat;

/* Initialize the WAVEFORMATEX for 16-bit, 44KHz, stereo */
waveFormat.wFormatTag = WAVE_FORMAT_PCM;
waveFormat.nChannels = 2;
waveFormat.nSamplesPerSec = 44100;
waveFormat.wBitsPerSample = 16;
waveFormat.nBlockAlign = waveFormat.nChannels * (waveFormat.wBitsPerSample/8);
waveFormat.nAvgBytesPerSec = waveFormat.nSamplesPerSec * waveFormat.nBlockAlign;
waveFormat.cbSize = 0;

/* Open the preferred Digital Audio Out device. Note: myWindow is a handle to some
open window */
result = waveOutOpen(&outHandle, WAVE_MAPPER, &waveFormat, (DWORD)myWindow, 0,
CALLBACK_WINDOW);
if (result)
{
   printf("There was an error opening the preferred Digital Audio Out device!\r\n");
}
```

Note: If the preferred device does not support your desired choice of sample rate and channels, then Windows will instead open some other device that does (assuming that there is such other device available).

Of course, if the user has no device installed capable of outputting or playing Digital Audio data, the above call returns an error, so always check that return value.

Likewise, use a Device ID of WAVE_MAPPER with waveInOpen() to open the preferred Digital Audio Input device. (Note that these two preferred devices may or may not be components of the same card. But that is irrelevant to your purposes. The only caveat is that if they are components upon the same card, the card's driver needs to be full duplex in order to simultaneously open both the Digital Audio input and output. In this way, your program can play back previously recorded waveforms while recording new waveforms. Without a full duplex driver, you have to open for recording, record a waveform, close the device using waveInClose(), and open for playback. Some sound card designs do not allow the card to simultaneously record and play digital audio, so they have only half duplex drivers).

```
unsigned long result;
HWAVEIN       inHandle;
WAVEFORMATEX  waveFormat;

/* Initialize the WAVEFORMATEX for 16-bit, 44KHz, stereo */
waveFormat.wFormatTag = WAVE_FORMAT_PCM;
waveFormat.nChannels = 2;
waveFormat.nSamplesPerSec = 44100;
waveFormat.wBitsPerSample = 16;
waveFormat.nBlockAlign = waveFormat.nChannels * (waveFormat.wBitsPerSample/8);
waveFormat.nAvgBytesPerSec = waveFormat.nSamplesPerSec * waveFormat.nBlockAlign;
waveFormat.cbSize = 0;

/* Open the preferred Digital Audio In device */
result = waveInOpen(&inHandle, WAVE_MAPPER, &waveFormat, (DWORD)myWindow, 0,
CALLBACK_WINDOW);
if (result)
{
   printf("There was an error opening the preferred Digital Audio In device!\r\n");
}
```

So what actually is the preferred Digital Audio Output device? Well, that's whatever device that the user choose from the dropdown list of Digital Audio Output devices under "Playback" of Control Panel's *Multimedia* utility (ie, on the "Audio" page). The list on this page is Windows actually displaying all of the names that were added to its list of devices capable of outputting or playing Digital Audio data.

The preferred Digital Audio Input device is whatever device that the user chooses from the dropdown list of Digital Audio Input devices under "Recording" of Control Panel's *Multimedia* utility (ie, on the "Audio" page). The list on this page is Windows actually displaying all of the names that were added to its list of devices capable of inputting or recording Digital Audio data.

The most flexible way to choose a Digital Audio device

The most flexible way would be to present the user with all of the names in the list of Digital Audio Output devices and let him choose which ones he wants (or if your program supports multiple Digital Audio output devices, you may wish to let him pick out several names from the list, and assign each digital audio "track" to one of those Device IDs. This is how professional sequencers implement support for multiple cards/outputs, in addition to perhaps implementing virtual tracks).

Whereas Windows maintains separate lists of Digital Audio Input and Output devices, so too, Windows has separate functions for querying the devices in each list.

Windows has a function that you can call to determine how many device names are in the list of devices that support outputting or playing Digital Audio data. This function is called waveOutGetNumDevs(). This returns the number of devices

in the list. Remember that the Device IDs start with 0 and increment. So if Windows says that there are 3 devices in the list, then you know that their Device IDs are 0, 1, and 2 respectively. You then use these Device IDs with other Windows functions. For example, there is a function you can call to get information about one of the devices in the list, for example its name, and what sort of other features it has such as what sample rates it supports. You pass the Device ID of the device which you want to get information about (as well as a pointer to a special structure called a WAVEOUTCAPS into which Windows puts the info about the device), The name of the function to get information about a particular Digital Audio Output device is waveOutGetDevCaps().

Here then is an example of going through the list of Digital Audio Output devices, and printing the name of each one:

```
WAVEOUTCAPS     woc;
unsigned long   iNumDevs, i;

/* Get the number of Digital Audio Out devices in this computer */
iNumDevs = waveOutGetNumDevs();

/* Go through all of those devices, displaying their names */
for (i = 0; i < iNumDevs; i++)
{
    /* Get info about the next device */
    if (!waveOutGetDevCaps(i, &woc, sizeof(WAVEOUTCAPS)))
    {
        /* Display its Device ID and name */
        printf("Device ID #%u: %s\r\n", i, woc.szPname);
    }
}
```

Likewise with Digital Audio Input devices, Windows has a function that you can call to determine how many device names are in the list of devices that support inputting or recording Digital Audio data. This function is called waveInGetNumDevs(). This returns the number of devices in the list. Again, the Device IDs start with 0 and increment. There is a function you can call to get information about one of the devices in the list, for example its name, and what sort of other features it has such as what sample rates it supports. You pass the Device ID of the device which you want to get information about (as well as a pointer to a special structure called a WAVEINCAPS into which Windows puts the info about the device), The name of the function to get information about a particular Digital Audio Input device is waveInGetDevCaps().

Here then is an example of going through the list of Digital Audio Input devices, and printing the name of each one:

```
WAVEINCAPS     wic;
unsigned long  iNumDevs, i;

/* Get the number of Digital Audio In devices in this computer */
iNumDevs = waveInGetNumDevs();

/* Go through all of those devices, displaying their names */
for (i = 0; i < iNumDevs; i++)
{
    /* Get info about the next device */
    if (!waveInGetDevCaps(i, &wic, sizeof(WAVEINCAPS)))
    {
        /* Display its Device ID and name */
        printf("Device ID #%u: %s\r\n", i, wic.szPname);
    }
}
```

Recording Digital Audio

The device's driver manages the actual recording of data. You can start and stop this process with waveInStart() and waveInStop(). While a driver records digital audio, it stores data into a small fixed-size buffer (for example 16K). When that buffer is full, the driver "signals" your program that the buffer is full and needs to be processed by your program (for example, your program may save that 16K of data to a disk file if your program is doing hard disk recording). The driver then goes on to store another "block" (ie, 16K section) of data into a second, similiarly-sized buffer. It's assumed that your program is simultaneously processing that first buffer of data, while the driver is recording into the second buffer. It's also assumed that your program finishes processing that first buffer before the second buffer is full. When the driver fills that second buffer, it again signals your program that now the second buffer needs to be processed. While your program is processing the second buffer, the driver is storing more audio data into the now-empty, first buffer. Etc. This all happens nonstop, so the process of recording digital audio is that two (or more if desired) buffers are constantly being filled by the driver (alternating between the 2 buffers), while your program is constantly processing each buffer immediately upon being signaled that the buffer is full. So, you end up dealing with a series of "blocks of data".

In fact, your program supplies each buffer to the driver, using waveInAddBuffer() (and waveInPrepareHeader() to initialize it). You supply the first 2 buffers to the driver using waveInAddBuffer() before recording. Every time that you're signaled that a buffer is filled, you need to use waveInAddBuffer() to indicate what buffer the driver will use after it finishes filling whatever buffer it is currently filling. (For double-buffering, that will be the same buffer that you're currently processing).

You can download my WaveIn C example to show how to record a (raw) digital audio file using double-buffering. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

Playing Digital Audio

Playback is also done via "blocks of data". Here, your application reads a block of data from the WAVE file on disk (for example, you may read the next 16K of the file into a 16K buffer). (You must use waveOutPrepareHeader() to initialize the buffer before reading into it). You pass this block to the driver for playback via waveOutWrite(). While the driver is playing this block, you're reading in another block of data into a second buffer. When the driver is finished playing the first block, it signals your program that it needs another block, and your driver passes that second buffer via waveOutWrite(). Your program will now read in the next block of data into the first buffer while the driver is playing the second buffer. Etc. Again, this is all non-stop until the WAVE is fully played (at which point you can call waveOutReset() to stop the driver's playback process).

So how does the driver "signal" your program? You've got a few choices. You can choose to have the driver send messages to your program's Window, for example, the MM_WOM_DONE message is sent each time the driver finishes playing a given buffer. Parameters with that message include the address of the given buffer (actually the address of the WAVEHDR structure which encompasses the buffer) and the device's handle (ie, the handle supplied to you when you opened the device). Or, you can have the driver automatically call a particular function in your program (ie, a "callback") passing such parameters. There are a couple of other choices such as having the driver use event signals or start a particular thread in your program.

You tell the driver how you want to be signaled by setting certain flags in one of the arguments to waveInOpen() or waveOutOpen().

You can download my WavePly1 C example to show how to play a WAVE file using a callback and double-buffering.

Setting volume and other parameters

There are other APIs that you'll likely want to use, such as waveOutSetVolume to set the volume for playback. Or, you may prefer to use the Mixer API (if the card's driver supports it) so that you can mute unneeded inputs/outputs, and adjust other parameters of a particular input/output line, or to determine what types of lines are available for recording (ie, analog microphone input, digital input, etc).

If you don't want the hassle of timing out the playback of MIDI data, nor loading/parsing MIDI files, you could use Windows High level MCI functions. These functions can use the MCI Sequencer Device to play an entire MIDI file on its own, in the background (ie, while your app does other things). But, you lose control over MIDI playback other than being able to stop, start, pause, rewind, and fast-forward the playback. There's Windows MCI functions for setting up, and controlling the MCI Sequencer Device, for example using mciSendCommand to send commands to the MCI Sequencer Device such that it opens a MIDI file on disk, reads it in, and plays it back.

Before proceeding, you should now read MCI Devices. This article gives necessary background information about the MCI API.

Opening the Sequencer Device

To open the Sequencer Device, you need to issue an "open" command, using either using mciSendString() or mciSendCommand() (depending upon whether you want to specify your open command by passing formatted strings, or binary values/structures, respectively), and specify that you want the Sequencer device as the type of device opened. Of course, you also need to supply the name of a MIDI file that you want the Sequencer to open and perform operations upon.

If you're using mciSendString(), you literally include "type sequencer" as part of the command string to indicate that you're opening the Sequencer device.

You can also specify an *alias*. This is just a string name that you use to identify the open device. Think of it as a string version of a device handle. After all, the string interface doesn't return binary values, so it can't return a handle. Instead it allows you to pick out a name, which you'll use with other commands you issue, in lieu of having a handle.

Here then is an example of opening the Sequencer device using the Command String interface. The MIDI file that we ask it to open is named C:\WINDOWS\SONG.MID. The alias we give this instance of the Sequencer is A_Song. (ie, Whenever we subsequently use A_Song as the device name with other commands, we'll be performing operations on the C:\WINDOWS\SONG.MID file).

```
TCHAR   buf[128];
DWORD   err;

/* Open a Sequencer device associated with the C:\WINDOWS\SONG.MID file */
if ((err = mciSendString("open C:\\WINDOWS\\SONG.MID type sequencer alias A_Song", 0,
0, 0)))
{
    /* Error */
    printf("Sequencer device did not open!\r\n");
    if (mciGetErrorString(err, &buf[0], sizeof(buf))) printf("%s\r\n", &buf[0]);
}
```

If you're using mciSendCommand(), then you need to initialize and pass a MCI_OPEN_PARMS structure. You set the lpstrDeviceType field of this structure to MCI_DEVTYPE_SEQUENCER. You also must pass mciSendCommand() the MCI_OPEN_TYPE and MCI_OPEN_TYPE_ID flags to indicate that you've set the lpstrDeviceType field to a predefined constant (MCI_DEVTYPE_SEQUENCER).

If playing back a MIDI file, then you also need to specify the name of the MIDI file to open by setting the lpstrElementName field to point to the name of the desired file. You also must pass mciSendCommand() the MCI_OPEN_ELEMENT flag to indicate that you've set the lpstrElementName field. The Sequencer device does not support recording MIDI.

The second arg will be MCI_OPEN to indicate an open command.

Here then is an example of opening the Sequencer device using the Command Message interface.

```
DWORD          err;
```

```
MCI_OPEN_PARMS midiParams;
TCHAR          buffer[128];

/* Open a Sequencer device associated with the C:\WINDOWS\SONG.MID file */
midiParams.lpstrDeviceType = (LPCSTR)MCI_DEVTYPE_SEQUENCER;
midiParams.lpstrElementName = "C:\\WINDOWS\\SONG.MID";
if ((err = mciSendCommand(0, MCI_OPEN,
MCI_WAIT|MCI_OPEN_ELEMENT|MCI_OPEN_TYPE|MCI_OPEN_TYPE_ID,
(DWORD)(LPVOID)&midiParams)))
{
    /* Error */
    printf("ERROR: Sequencer device did not open!\r\n");
    if (mciGetErrorString(err, &buffer[0], sizeof(buffer))) printf("%s\r\n",
&buffer[0]);
}
else
{
    /* The device opened successfully. midiParams.wDeviceID now contains the device
ID */
}
```

Playing the Sequencer Device

To have the Sequencer device play its open MIDI file, you issue a play command to it.

If you're using mciSendCommand(), then you need to initialize and pass a MCI_PLAY_PARMS structure. (Actually, unless you use the MCI_NOTIFY, MCI_FROM, and/or MCI_TO flags, there is no initialization required). If you want to play only part of the MIDI file, then you can set the dwFrom and/or dwTo fields to the start and end positions respectively, and pass the MCI_FROM and/or MCI_TO flags respectively. The dwFrom and dwTo fields are normally expressed in terms of PPQN clocks. But you can utilize other means of expressing this offset, for example in terms of milliseconds, by first issuing an MCI_SET command (with the MCI_SET_TIME_FORMAT flag, and your desired choice of how to express such offsets).

The second arg will be MCI_PLAY to indicate a play command.

The first arg will be the device ID that you obtained when you opened the device.

Here then is an example of playing a MIDI file on the Sequencer device using the Command Message interface.

```
DWORD          err;
MCI_PLAY_PARMS playParams;
TCHAR          buffer[128];

/* Play a Sequencer device. Assume that midiParams.wDeviceID was set according to our
open example above */
if ((err = mciSendCommand(midiParams.wDeviceID, MCI_PLAY, MCI_WAIT,
(DWORD)(LPVOID)&playParams)))
{
    /* Error */
    printf("ERROR: Midi did not play!\r\n");
    if (mciGetErrorString(err, &buffer[0], sizeof(buffer))) printf("%s\r\n",
&buffer[0]);
}
else
{
    /* The midi song has played from beginning to end */
}
```

You can download my Message Midi Play C example to show how to play a MIDI file using the Command Message interface. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

Closing the Sequencer Device

To close the Sequencer device, you issue a close command to it.

If you're using mciSendCommand(), then you need to initialize and pass a MCI_GENERIC_PARMS structure. (Note that this structure is a subset of all of the other MCI structures that you pass to mciSendCommand(). In other words, you can substitute any of the other MCI structures for this one). Unless you use the MCI_NOTIFY flag, there is actually no initialization required.

The second arg will be MCI_CLOSE to indicate a close command.

The first arg will be the device ID that you obtained when you opened the device.

Here then is an example of closing the Sequencer device using the Command Message interface.

```
/* Close the Sequencer device. Assume that midiParams.wDeviceID was set according to
our open example above */
mciSendCommand(midiParams.wDeviceID, MCI_CLOSE, MCI_WAIT,
(DWORD)(LPVOID)&midiParams);
```

With the advent of computer audio cards that can record and playback digital audio, computer software has appeared which turns a computer into a device that can not only record and playback that digital audio, but also present that data in a way that makes it easy for musicians to view and edit it. The data can be graphically displayed upon a computer monitor, and can be manipulated with the mouse, for example. Furthermore, computer software can also perform its own manipulations of that data, yielding effects such as delay, transposition, chorus, compression, etc, sometimes even in real-time (ie, while the audio data is playing back). The wide range of computer audio products also means that a computer digital audio system can be tailored to many budgets. And being that computers are typically more easily upgradable than dedicated digital audio units (for example, adding a second hard drive to accomodate more digital audio tracks), and can do other things besides digital audio work, they are often ultimately more versatile and cost effective than dedicated digital audio units. In short, with good audio hardware and software, computers make very good digital audio workstations.

Examples of software that supports both digital audio recording, as well as MIDI, are CakeWalk Pro Audio, PG Music's Power Tracks, Steinberg's Cubase, etc. Examples of programs that specialize in digital audio recording (and may have a more powerful and easier feature set for digital audio editing than the sequencers) are Cool Edit, Sound Forge, SAW, Samplitude, etc.

Digital Audio Recording

A typical computer system works with digital audio in the following way. First, to record digital audio, you need a card with an Analog to Digital Converter (ADC) circuitry on it. This ADC is attached to the Line In (and Mic In) jack of your audio card, and converts the incoming analog audio to a digital signal that computer software can store on your hard drive, visually display on the computer's monitor, mathematically manipulate in order to add effects or process the sound, etc. (When I say "incoming analog audio", I'm referring to whatever you're pumping into the Line In or Mic In of your sound card, for example, the output from a mixing console, or the audio output of an electronic instrument, or the sound of some acoustic instrument or voice being feed through a microphone plugged into the sound card's Mic In, etc). While the incoming analog audio is being recorded, the ADC is creating many, many digital values in its conversion to a digital audio representation of what is being recorded. Think of it as analogous to a cassette recorder. While you're recording some analog audio to a cassette tape, the tape is constantly passing over the record head. So the longer the passage of music you record, the more cassette tape you use to record that analog audio signal onto the tape. So too with the conversion to digital audio. The longer the passage of music you record (ie, digitize), the more digital values are created, and these values must be stored for later playback.

Where are these digital values stored? Well, as your sound card creates each value, that data is passed to the card's software driver which then passes that data to the software program managing the recording process. Such software might be CakeWalk recording a digital audio track, or SAW, or Samplitude, or Windows Sound Recorder, or any other program capable of initiating and managing the recording of digital audio. Whereas the program may temporarily accumulate those digital audio values in the computer's RAM, those values will eventually have to be stored upon some fixed

medium for permanence. That medium is your computer's hard drive. (For this reason, sometimes people refer to the process of recording digital audio to a hard drive as "Hard Disk Recording". Henceforth, I will abbreviate "hard drive" as HD). Usually, how this works is that the program accumulates a "block" of data in RAM (while the sound card is digitizing the incoming analog audio), for example 4,000 data values, and then writes these 4,000 values into a file on your HD. (It's a lot more efficient to write 4,000 values at once to your hard drive, than it is to write those 4,000 values one after the other separately). All of these 4,000 values go into one file. Usually, the format for how the data is arranged within this file follows the WAVE file format. (But there are other formats that may also be used by programs to store digital audio values. For example, AIFF is often used on the Macintosh. AU format is used on Sun computers. MP3 is a popular format nowadays because it compresses the data's size. Etc. Any of these formats could be used on any computer, but WAVE is considered the standard on a Windows-based PC). Now, if the recording process is still going on, the software program will collect 4,000 more values in RAM, and then write them out to the same WAVE file on the HD (without erasing the previously stored 4,000 values -- ie, the values accumulate within the WAVE file, so that it now has 8,000 values in it). This process continues until the musician halts the recording process. (ie, If you let recording continue long enough, it will eventually fill up your HD with digital audio values in one, big WAVE file). At that point, the WAVE file is complete, and contains all of the digital audio values representing the analog audio recorded.

So, digital audio recorded by most PC programs is predominantly stored in a WAVE file on your HD, and that file is created while the digital audio is being created/recorded.

Digital Audio Playback

In order to subsequently playback this digital audio (ie, WAVE file), you need a card with a Digital To Analog Converter (DAC) circuitry on it. Needless to say, most sound cards have both an ADC and a DAC so that the card can both record and play digital audio. This DAC is attached to the Line Out jack of your audio card, and converts the digital audio values back into the original analog audio (that was initially recorded during the recording process). This analog audio can then be routed to a mixer, or speakers, or headphones so that you can hear the recreation of what was originally recorded. You need software to manage the playback of digital audio, and not surprisingly, the same program that was used to manage the recording process usually can also manage the playback. For example, CakeWalk can playback the digital audio track that it recorded. The playback process is almost an exact reverse of the recording process. The program reads a block of digital audio data from the WAVE file on the HD. For example, CalkWalk may read the first 4,000 data values. (It's more efficient to read 4,000 values off of the HD at once, than it is to read those 4,000 values one after the other separately). Then, the program passes each one of these values to the card's driver which feeds it to the card's DAC. The program then reads the next 4,000 values from the WAVE file, and plays those back as described. In other words, the sound card is recreating the original analog audio while the program is reading the digital audio values off of the HD and passing them back to the card. This continues until the program has played all of the values in the WAVE file (or until the musician interrupts the playback).

Data processing during playback

Some programs can optionally perform some mathematical manipulation of the digital audio values immediately before the data is passed to the card's driver (ie, during playback). Such manipulations may be to add effects such as reverb, chorus, delay, etc. Programs that do such realtime processing are SAW and Samplitude.

Some programs also can process the digital audio to do some valuable things such as "time-stretching" and "pitch shift" (although these are often too computationally complex for today's computer to do during playback. So, this processing usually has to be applied before playback, and may be "destructive" in that it permanently alters the digital audio data).

Time-stretching: When you play a one-shot (ie, non-looped) waveform, it lasts only so long (ie, in terms of time). For example, maybe you've got digital audio tracks of a piece of music whose duration is 2 minutes. Sometimes, people need to adjust the length of time over which the waveform plays. For example, maybe the producer of a movie says "I want this piece of music to last exactly 2 minutes and 3 seconds in order to fit with this filmed scene I have which is this long". Well, if you had a MIDI track, you'd just slow the tempo a little in order to make the music last that extra 3 seconds. The music would sound exactly the same, but at a slightly slower tempo. OK, so how do you "slow down" the digital audio tracks? Well, you could reduce the playback rate a little. But, if you've ever used a sampler, you'll notice that when you play a waveform at a rate different than it was recorded, this changes the character of the waveform itself. You don't just hear a different tempo, you hear different pitch, vibrato, tremulo, tone, etc. So, time-stretching was devised to take a waveform, analyze it, and change its length without (hopefully) changing its characteristics. The net result is that you get the same effect that you had by changing the tempo of the MIDI track; ie, merely a change in tempo/duration rather than a change in the characteristics of the waveform. (Nevertheless, there is always some potential for a change in timbre when time-stretching algorithms are applied to a waveform).

Pitch shift: This changes the note's pitch without altering the playback rate (which would alter other characteristics of the waveform). So, if you sampled the middle C note of a piano at 44KHz, but when you play it back at 44KHz, you really want to hear a D note, you could apply pitch shift to it to create a waveform whose "root" (ie, the pitch you get when you playback at the same sample rate as when recorded) is D. If you take a musical performance with "rhythms" in it, and apply pitch shift, you should get a different pitch, but retain the same rhythms (ie, tempo).

Virtual Tracks

Although most sound cards have only 2 discrete digital audio channels (ie, stereo digital audio capabilities), many programs such as CakeWalk support recording and playing many more tracks of digital audio. And yet, CakeWalk seemingly plays more than 2 digital audio tracks using such a card.

How is this done? It is accomplished through the use of "virtual tracks". What this means is that the program allows you to record as many digital audio tracks as you like, mono and/or stereo. The only limitation is your HD space, plus how many tracks can be read off of your HD at the required rate of playback. (You can record only 2 mono tracks, or 1 stereo track at one time, due to the card's limit of only 2 channels. But, you can do as many iterations of the recording process as you wish to yield many more than 2 tracks. It's on playback that the concept of virtual tracks works). For example, you could record 4 mono tracks, plus 3 stereo tracks (if you had a fast enough HD -- doubtful on slow IDE stuff). You can set the individual panning for each of the mono tracks. Then, upon playback (ie, when CakeWalk reads the data for all of those digital audio tracks from their WAVE files), CakeWalk itself mathematically mixes all of the digital audio tracks into one stereo digital audio mix, and outputs this mix to the sound card's stereo DAC. In other words, CakeWalk itself functions as a sort of "digital mixer", mixing many mono and stereo tracks together (during playback) into one stereo digital audio track. So, using any sound card with a stereo digital audio DAC, you can actually record as many tracks as you like. CakeWalk virtualizes the card so that it appears to have many digital audio tracks, mono and/or stereo. Most pro programs that work with digital audio support the concept of "virtual tracks".

There are a few drawbacks to this scheme though. First, the more tracks that you record, the more that you have to back off on the individual volume of each track. Why? Because when all of the tracks are mathematically summed together by CakeWalk, if at any point the sum exceeds a 16-bit value, you'll get clipping. The more tracks that you sum, the more likely you are to get clipping -- unless you back off on the individual track volumes more with each added track. It doesn't matter if your card has 18-bit (ie, Tahiti) or 20-bit (ie, Pinnacle) DACs. CakeWalk itself performs the sum into a 16-bit mix, so there is an inherent 16-bit limitation to CakeWalk's output (or any other program that is limited to 16-bit digital audio -- some digital audio programs offer greater than 16-bit resolution for their internal mixing, such as 24-bit or 32-bit. These programs don't require you to back off on the individual wave volumes so much. If you're using a lot of virtual tracks, use a program with at least 24-bit resolution for its internal mixing). So the more tracks you record, the less dynamic range you get for each individual track as you turn down the individual volume to avoid clipping the mix. I doubt that you'd want to mix more than 4 mono virtual tracks to one card. (On the plus side, if you get more digital audio cards, you can then split up virtual tracks among them since CakeWalk can use more than 1 card simultaneously. But remember that most ISA cards require at least 1 DMA channel for playback, and 1 for full-duplex recording, and a PC has only 3 16-bit DMA channels, so you're pretty much limited to 2 ISA sound cards -- unless you get a Tahiti, Fiji, or Pinnacle which use a proprietary, non-DMA method of output, or a PCI card as those do not use motherboard DMA). Of course, you could get one of those cards with 8 digital audio channels like a DAL V8, Antex SoundCard, EMagic Audiowerk8, or DigiDesign Session 8, which usually have their own accompanying program to support better than 16-bit DACs and perhaps better throughput.

Secondly, you may lose some of the individual hardware control of the card when CakeWalk takes over the task of digitally mixing the output. For example, CakeWalk has to operate a Roland RAP-10 in its stereo mode, so you lose the RAP's individual control over each track's reverb/delay and chorus (effective when the RAP-10 channels are used as 2 Mono channels). In other words, most programs take a generic approach to virtual tracks which may not support some of the more esoteric functions of certain cards. So, it's important that if you get a card that has something more than just a simple, stereo 16-bit DAC, you make sure that you have software support for that additional functionality lest

you want it to be wasted by a program that treats your card as if it were a simple, stereo 16-bit DAC.

Hard Drive requirements

With digital audio tracks recorded to a HD, usually the limiting factor in how many virtual tracks can be simultaneously played, and how well the audio sounds, is the speed at which data can be read/written to your HD. A slow HD (more than 12ms access time) is usually the limiting factor in how many virtual tracks can be used. Also, you need a HD that has no problems with thermal recalibration (ie, lengthy adjustments the HD may make during reading/writing that can cause a delay in accessing the drive). These delays may cause a program to fail to feed a continuous stream of digital audio values to/from the sound card, and you'll then hear "glitches" in the audio. Newer HD designs have tended to minimize this problem. Select a good HD for digital audio. It may be even more critical than differences in sound card designs. If you go with an EIDE HD, make sure that you get one that supports bus mastering (in Mode 4) or at least DMA/33, and you use it with a motherboard that has a PCI EIDE controller and drivers that support EIDE bus mastering. (Win95's Service Pack 2 added such support to Win95). Otherwise, SCSI has an advantage in that most SCSI controllers support bus mastering, scatter-gather lists, and other features that make for efficient HD I/O, and which are supported by most operating systems' drivers.

Furthermore, since digital audio requires a constant stream of values throughout the recording process (unlike with MIDI), you need a large HD if you want to record long digital audio tracks. You'll use up 5 MEG of HD space for every minute of a 16-bit mono digital audio track recorded at 44.1Khz (and 10 MEG for a stereo track). In other words, recording a CD-quality digital stereo track that is 5 minutes long will use up 50 MEG of your HD.

Audio card requirements

Of course, this is not to say that the quality of your sound card isn't important too. If you've got a card with a cheap DAC or ADC, you're going to get noisy digital audio. (ie, The result will typically sound like a "graininess" to the audio or even "tape hiss". On the other hand, problems with the HD usually manifest in horrid distortion or weird noises such as pops and clicks). To get nicely recorded digital audio, and as many simultaneous virtual tracks as possible, you'll want both a fast, large HD with efficient controller I/O, and a card with a good DAC and ADC.

Unless you've got a card that has digital I/O so that you can run a digital connection right to a DAT deck, for example, you'll also want a card that has a clean (ie, low Thd distortion) and quiet (ie, low signal-to-noise ratio) audio output stage.

For more information about digital audio cards themselves, see Digital Audio Cards.

If you don't want the hassle of loading/parsing WAVE files, nor handling buffers of digital audio data, then you can use the Windows High level Digital Audio API. These functions can use the MCI Wave Audio Device to play or record an entire WAVE file on its own, in the background (ie, while your app does other things). But, you lose control over Digital Audio playback/recording other than being able to stop, start, pause, rewind, and fast-forward the playback/recording. There's Windows MCI functions for setting up, and controlling the MCI Wave Audio Device, for example using mciSendCommand() to send commands to the the MCI Wave Audio Device such that it opens a WAVE file on disk, reads it in, and plays it back.

Very simple digital audio playback: PlaySound()

Windows has a function called PlaySound(). Calling this function causes Windows to play an entire, digital audio waveform that has been loaded into memory upon the default Digital Audio Output device. (The default device is whatever device the user has picked out for Audio playback upon the Audio page of Control Panel's MultiMedia utility). It's the easiest method for playing a waveform, but offers the least amount of control, for example, you can only start and stop the playback. (ie, You can't set the playback to start at a certain location in the waveform). Of course, as its name implies, it is only concerned with playing, not recording, digital audio.

PlaySound() can open some waveform file (on disk), and load the waveform data into memory prior to playback. PlaySound() completely manages opening, loading, and closing the file, Of course, one of the args you pass to PlaySound() lets it know which WAVE file to play.

Furthermore, it has the option of doing this in the background. (ie, PlaySound can return immediately while Windows goes off and does all of the work of loading and playing that waveform file in the background). Of course, one of the args you pass to PlaySound() lets it know whether you want this option (or other options to be discussed later).

But there are a few caveats to this approach

1. The waveform must be stored in WAVE File Format. This is the standard format for digital audio on an Intel based PC. PlaySound does not deal with raw waveform data (ie, stored without the appropriate WAVE File Format headers). It also cannot play AIFF or MPEG audio files.

2. The entire waveform must be able to fit into available memory. If you need to play a waveform larger than available memory (by loading it in smaller "blocks"), then you'll either have to use the other High level options (ie, the Wave Audio device) described later, or use the Low level Digital Audio API.

3. Because the entire waveform has to be completely loaded before playback even starts, with large files, this can result in a significant delay before the audio actually starts. An exception to this is using the SND_MEMORY flag described later.

4. The PlaySound function can't really be used simultaneously by multiple threads in the same process. It provides no means to arbitrate your threads' calls to PlaySound(), so unless you do that yourself, results may be unpredictable.

Playing WAVE files on disk

In order to tell PlaySound() to load and play a particular WAVE file on disk, you pass a pointer to the filename, and also set the SND_FILENAME flag. PlaySound() will completely load and play the WAVE file, and then return when that playback is finished.

Here I play the file C:\WINDOWS\CHORD.WAV and wait for it to finish:

```
if (!PlaySound("C:\\WINDOWS\\CHORD.WAV", 0, SND_FILENAME))
{
    printf("The sound didn't play!");
}
```

NOTE: It is possible to omit the SND_FILENAME flag. In this case, Windows first searches the WIN.INI file for any [Sound] key names (described later) that match the name you've passed, and then if none match, it searches for a matching filename on disk. But to avoid possible name collisions with WIN.INI [Sound] keys, I recommend using the SND_FILENAME flag.

Background playback

If you want PlaySound() to return immediately (so that your program can go on to do other things) while Windows loads/plays the file in the background, then also specify the SND_ASYNC flag.

Here I play the file C:\WINDOWS\CHORD.WAV in the background:

```
if (!PlaySound("C:\\WINDOWS\\CHORD.WAV", 0, SND_ASYNC | SND_FILENAME))
{
    printf("The sound didn't play!");
}

/* Now I can do other things while that waveform is playing */
```

Stopping playback

Of course, Windows stops the playback automatically when it gets to end of the waveform (unless you loop it as described later).

But what if you want to stop playback prematurely? (ie, You're a premature audiojaculator). First of all, you must use the SND_ASYNC flag (ie, specify an asyncronously playing waveform). After all, if you don't, then PlaySound() doesn't return until the waveform is completely finished playing.

As noted, when you specify the SND_ASYNC flag, then the call to PlaySound() returns while Windows loads and plays the waveform in the background. Your program can then go on to do other things. One of the things that it can do is make another call to PlaySound() to play another WAVE file. So what happens to that first waveform that has been playing in the background? Windows stops playing it and starts playing the new WAVE that you've specified. In conclusion, one way to prematurely stop a waveform from playing is to simply call PlaySound() again to play another waveform. An exception to this is if your second call to PlaySound() specifies the SND_NOSTOP flag as described later.

Alternately, if you want to prematurely stop a waveform's playback without starting another waveform playing, then simply call PlaySound(), passing a 0 as the first arg. Here, I stop any currently playing waveform:

```
PlaySound(0, 0, 0);
```

Preventing premature playback stop

So what if you don't wish to prematurely stop the playback of some asyncronously playing waveform when you make another call to PlaySound()? Simply specify the SND_NOSTOP flag. Does this mean that the two waveforms then overlap in playback? (ie, Does the second waveform start playing along with the first, thus mixing the two waveforms' playback?) You wish! Unfortunately, it just means that if some asyncronously playing waveform is still playing when you call PlaySound() with the SND_NOSTOP flag, then PlaySound() returns immediately with a 0 indicating an error. It doesn't interrupt the previous playing waveform in order to play the new waveform. (For easy, real-time mixing of digital audio, you should look at the DirectSound API). So, SND_NOSTOP simply allows you to prevent one call to PlaySound() from inadvertently cutting off the playback of a previous call to PlaySound().

Waveform looping

You can also have PlaySound() continuously loop the playback (ie, as soon as Windows gets to the end of the waveform, it immediately starts playing again from the start of the waveform). You specify the SND_LOOP flag. In this case, you must also specify the SND_ASYNC flag so that PlaySound() returns immediately. The looping then continues (in the background) until

you call PlaySound() again, either to play another waveform, or to stop the currently playing waveform.

Here is an example of playing a looped waveform in the background:

```
BOOL result;

/* Tell Windows to load and continuously loop the waveform */
result = PlaySound("C:\\WINDOWS\\CHORD.WAV", 0, SND_LOOP | SND_ASYNC | SND_FILENAME);

/* Go do something else here while the waveform is looping */

/* Now finally stop the looped playback (without playing another waveform) */
if (result) PlaySound(0, 0, 0);
```

What happens in the case of an error

If the WAVE file does not exist, or doesn't fit into the available memory, PlaySound() plays the default "system sound". What is that? That is whatever WAVE file the user has set to play as the "Default Sound" via the Control Panel's *Sound* utility. (ie, The user can set various WAVE files to play for various "operations" he performs. For example, he can set a WAVE file of a door opening to play every time that he opens a Desktop folder. These are referred to as "system sounds", and are set via the Sound utility). Of course, if the user hasn't set a WAVE file for the Default Sound, then PlaySound plays nothing. It simply returns 0 indicating an error.

If you don't want PlaySound() to play the Default Sound when it can't find or load your desired WAVE file, then specify the SND_NODEFAULT flag. In this case, if PlaySound() can't find/load your desired file, it plays nothing, and simply returns a 0 to indicate an error.

WIN.INI [Sound] keys

PlaySound() can also play sounds referred to by a keyname under the [Sound] section of the WIN.INI file. To play one of these "sound events", pass a pointer to the name of the key that identifies the sound and don't specify the SND_FILENAME flag. For example, let's assume that the [Sound] section of the WIN.INI file looks like this:

```
[Sound]
MouseClick = C:\WINDOWS\CHORD.WAV
```

To play the sound associated with the "MouseClick" key (ie, the WAVE file named "C:\WINDOWS\CHORD.WAV") and to wait for the sound to complete before returning, you do:

```
PlaySound("MouseClick", 0, SND_NODEFAULT);
```

The names of some [Sound] keys that you'll find on all Win32 platforms are:

```
SystemAsterisk
SystemExclamation
SystemExit
SystemHand
SystemQuestion
SystemStart
```

There may be other [Sound] keys. In fact, your program can add more keys to WIN.INI's [Sound] section using WriteProfileString(). For example, maybe your installer will add a key name of JoeCompanyWavePlayerStart with its value set to "C:\START.WAV". Then, when your program starts up, it can call PlaySound() passing the string "JoeCompanyWavePlayerStart". An advantage of this is that the user can easily reassign different WAVE files to the various keys you've added, by editing the WIN.INI file. (ie, It's a text file).

But on Win32, you can add keys to the registry instead of the WIN.INI file, and group them under your own heading. Instead of having to edit the WIN.INI file, the user can use Control Panel's Sound utility to edit your keys. (ie, All of your keys will be listed under your heading, and you can give a more meaningful label to each key as well). The net result is that when they are displayed in Control Panel's Sound utility, the user sees all of your keys grouped under a heading of your choice with descriptive names. It makes it a lot easier for him to edit those WAVE associations. You can download my Playsound Registry C example to show how to add such registry keys and then play the waves associated with them. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

Playing embedded WAVE resources

PlaySound() can even be used to load and play a WAVE resource embedded in your program's executable. To do so, specify the SND_RESOURCE flag. In this case, the first arg is a pointer to a string that specifies the ID of your embedded resource,

Of course, when you create your executable, you should make sure that you embed the desired WAVE file in it as a resource.

For example, assume that you have the following WAVE resource in your resource file (referencing the C:\WINDOWS\CHORD.WAV file)

```
IDR_WAVE1   WAVE    DISCARDABLE    "c:\\windows\\chord.wav"
```

And furthermore, let's say that the IDR_WAVE1 symbol is defined to be the value 129.

Here is how you play that WAVE resource:

```
PlaySound("#129", 0, SND_RESOURCE | SND_NODEFAULT);
```

Download my Playsound Resource C example to show how to play a WAVE resource. Included are the Project Workspace files for Visual C++ 4.0. This example is a plain C program that any Windows compiler should be able to compile.

If the WAVE isn't embedded in your executable, for example, it's in a resource embedded in a Dynamic Link Library, then pass a handle to that object (ie, for example, a handle to the DLL) as the second arg to PlaySound(). One good use of waveforms embedded in DLLs, is that you can have different "libraries" of WAVE files, and easily switch between them just by referencing different DLLs. For example, you can have a DLL that contains 8-bit versions of your waveforms versus a DLL that contains 16-bit versions of the same. You don't have to use different filenames for the waveforms -- just different filenames for the DLLs (ie, LoadLibrary() different DLLs depending upon which versions of waves to use).

Playing WAVEs already in memory

PlaySound() can play a WAVE file that is already loaded into memory. To do so, specify the SND_MEMORY flag. In this case, the first arg is a pointer to the memory buffer containing the waveform.

Doing this eliminates most all of the delay prior to playback that you may get using other PlaySound() methods.

One caveat is that the memory buffer you pass to PlaySound() must contain the image of a complete WAVE File Format file. It can't just be raw waveform data.

Download my Playsound Memory C example to show how to play a WAVE file in memory. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it.

If you need a more sophisticated way of playing/recording digital audio data than PlaySound() offers (but don't want to resort

to the Low level Digital Audio API), then you need to utilize the MCI Wave Audio Device. This is a part of Windows that manages a lot of the playback/recording process. But because it offers you a bit more direct control over the playback/recording, needless to say, it requires a bit more programming than using the very simple approach of PlaySound().

The Wave Audio device supports playing a WAVE file, without requiring that it fit into available RAM. And for large WAVE files, it can start playback quicker than PlaySound() because the Wave Audio device loads/plays the waveform in "smaller blocks" (ie, spools the waveform data into RAM) rather than loading it entirely into RAM prior to playback.

Before proceeding, you should now read MCI Devices. This article gives necessary background information about the MCI Wave Audio Device.

Opening the Wave Audio Device

To open the Wave Audio Device, you need to issue an "open" command, using either using mciSendString() or mciSendCommand() (depending upon whether you want to specify your open command by passing formatted strings, or binary values/structures, respectively), and specify that you want the Wave Audio device as the type of device opened. Of course, you also need to supply the name of a WAVE file that you want the Wave Audio to open and perform operations upon.

If you're using mciSendString(), you literally include "type waveaudio" as part of the command string to indicate that you're opening the Wave Audio device.

You can also specify an *alias*. This is just a string name that you use to identify the open device, Think of it as a string version of a device handle. After all, the string interface doesn't return binary values, so it can't return a handle. Instead it allows you to pick out a name, which you'll use with other commands you issue, in lieu of having a handle.

Here then is an example of opening the Wave Audio device using the Command String interface. The WAVE file that we ask it to open is named C:\WINDOWS\CHORD.WAV. The alias we give this instance of the Wave Device is A_Chord. (ie, Whenever we subsequently use A_Chord as the device name with other commands, we'll be performing operations on the C:\WINDOWS\CHORD.WAV file).

```
TCHAR   buf[128];
DWORD   err;

if ((err = mciSendString("open C:\\WINDOWS\\CHORD.WAV type waveaudio alias A_Chord",
0, 0, 0)))
{
    /* Error */
    printf("Wave Audio device did not open!\r\n");
    if (mciGetErrorString(err, &buf[0], sizeof(buf))) printf("%s\r\n", &buf[0]);
}
```

If you're using mciSendCommand(), then you need to initialize and pass a MCI_WAVE_OPEN_PARMS structure. You set the lpstrDeviceType field of this structure to MCI_DEVTYPE_WAVEFORM_AUDIO. You also must pass mciSendCommand() the MCI_OPEN_TYPE and MCI_OPEN_TYPE_ID flags to indicate that you've set the lpstrDeviceType field to a predefined constant (MCI_DEVTYPE_WAVEFORM_AUDIO).

If playing back a WAVE file, then you also need to specify the name of the WAVE file to open by setting the lpstrElementName field to point to the name of the desired file. You also must pass mciSendCommand() the MCI_OPEN_ELEMENT flag to indicate that you've set the lpstrElementName field. If recording a WAVE, you have the option to specify the name now, or do so later with a Save operation.

The second arg will be MCI_OPEN to indicate an open command.

Here then is an example of opening the Wave Audio device using the Command Message interface.

```
DWORD               err;
MCI_WAVE_OPEN_PARMS waveParams;
TCHAR               buffer[128];

/* Open a Wave Audio device associated with the C:\WINDOWS\CHORD.WAV file */
waveParams.lpstrDeviceType = (LPCSTR)MCI_DEVTYPE_WAVEFORM_AUDIO;
waveParams.lpstrElementName = "C:\\WINDOWS\\CHORD.WAV";
if ((err = mciSendCommand(0, MCI_OPEN,
MCI_WAIT|MCI_OPEN_ELEMENT|MCI_OPEN_TYPE|MCI_OPEN_TYPE_ID,
(DWORD)(LPVOID)&waveParams)))
{
    /* Error */
    printf("ERROR: Wave Audio device did not open!\r\n");
    if (mciGetErrorString(err, &buffer[0], sizeof(buffer))) printf("%s\r\n",
&buffer[0]);
}
else
{
    /* The device opened successfully. waveParams.wDeviceID now contains the device
ID */
}
```

Playing the Wave Audio Device

To have the Wave Audio device plays its open WAVE file, you issue a play command to it.

If you're using mciSendCommand(), then you need to initialize and pass a MCI_PLAY_PARMS structure. (Actually, unless you use the MCI_NOTIFY, MCI_FROM, and/or MCI_TO flags, there is no initialization required). If you want to play only part of the waveform, then you can set the dwFrom and/or dwTo fields to the start and end positions respectively, and pass the MCI_FROM and/or MCI_TO flags respectively. The dwFrom and dwTo fields are normally expressed in terms of sample offset (ie, a byte offset). But you can utilize other means of expressing this offset, for example in terms of milliseconds, by first issuing an MCI_SET command (with the MCI_SET_TIME_FORMAT flag, and your desired choice of how to express such offsets).

The second arg will be MCI_PLAY to indicate a play command.

The first arg will be the device ID that you obtained when you opened the device.

Here then is an example of playing a WAVE on the Wave Audio device using the Command Message interface.

```
DWORD          err;
MCI_PLAY_PARMS playParams;
TCHAR          buffer[128];

/* Play a Wave Audio device. Assume that waveParams.wDeviceID was set according to
our open example above */
if ((err = mciSendCommand(waveParams.wDeviceID, MCI_PLAY, MCI_WAIT,
(DWORD)(LPVOID)&playParams)))
{
    /* Error */
    printf("ERROR: Wave did not play!\r\n");
    if (mciGetErrorString(err, &buffer[0], sizeof(buffer))) printf("%s\r\n",
&buffer[0]);
}
else
{
```

```
    /* The wave has played from beginning to end */
}
```

You can download my MCI Wave Play C example to show how to play a WAVE file using the Command Message and String interfaces. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

Closing the Wave Audio Device

To close the Wave Audio device, you issue a close command to it.

If you're using mciSendCommand(), then you need to initialize and pass a MCI_GENERIC_PARMS structure. (Note that this structure is a subset of all of the other MCI structures that you pass to mciSendCommand(). In other words, you can substitute any of the other MCI structures for this one). Unless you use the MCI_NOTIFY flag, there is actually no initialization required.

The second arg will be MCI_CLOSE to indicate a close command.

The first arg will be the device ID that you obtained when you opened the device.

Here then is an example of closing the Wave Audio device using the Command Message interface.

```
/* Close the Wave Audio device. Assume that waveParams.wDeviceID was set according to
our open example above */
mciSendCommand(waveParams.wDeviceID, MCI_CLOSE, MCI_WAIT,
(DWORD)(LPVOID)&waveParams);
```

Recording with the Wave Audio Device

You can download my MCI Wave Record C example to show how to record a WAVE file using the Command Message and String interfaces. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it.

To use the MIDI Stream API, you need to first call midiStreamOpen() once to open some MIDI stream device for output. (This is similiar to the Low level MIDI API's midiOutOpen). You pass the Device ID of that desired stream device. Then, you can subsequently call other Stream functions, such as midiStreamOut() to queue MIDI messages for playback upon that device, and midiStreamRestart() to start the actual playback. Windows and that device will take care of timing out each event and outputting its MIDI bytes.

After you're done outputting to a stream device (and have no further use for it), you must close that device.

Think of a stream device like a file. You open it, you write to it (ie, queue MIDI events to it which Windows "plays"), and then you close it.

Opening the default Stream for playback

This is extremely similiar to the Low level MIDI API's midiOutOpen() (except that a pointer to the device ID is passed rather than the device ID directly passed, and also, there is one extra arg that is currently set to 1). To open the default device, use a Device ID of 0 as so:

```
unsigned long result, deviceID;
HMIDISTRM     outHandle;

/* Open default MIDI Out stream device */
deviceID = 0;
result = midiStreamOpen(&outHandle, &deviceID, 1, 0, 0, CALLBACK_NULL);
if (result)
{
   printf("There was an error opening the default MIDI stream device!\r\n");
}
```

The most flexible way to choose a MIDI Stream device for input or output

Windows Stream Manager is capable of streaming MIDI data to any one of the MIDI Output Devices on a system. So, in order to get a list of MIDI Out Devices that you can open with midiStreamOpen(), you simply do things exactly as you would with the low level's midiOutOpen(). You query all of the MIDI Output devices in the system using midiOutGetDevCaps(), and remember that the first device always has an ID of 0, and subsequent devices have ascending ID numbers.

Individual Windows 95 drivers for any installed cards may be specially written to directly support the Stream API. When you query a particular MIDI Output device (using midiOutGetDevCaps), you can check the dwSupport field of the MIDIOUTCAPS structure. If the driver directly supports the Stream API, the MIDICAPS_STREAM bit of dwSupport will be set. (Download my ListMidiDevs C example to show how to query the MIDI Output devices for such support). Such a driver may offer special features such as allowing the Stream Manager to sync playback to incoming SMPTE or MIDI Time Code. Those features would be determined by what code has been added to the driver, as well as perhaps hardware support by the MIDI card itself.

One caveat is that the MIDI Mapper cannot be opened as the output for a MIDI stream. The Windows MIDI Stream Manager supports only one MIDI output at a time (unless the card's driver directly supports the Stream API and adds some sort of routing feature based upon the MIDIEVENT's dwStreamID field. Currently, the Stream Manager doesn't support this routing).

General overview of stream playback

The process of passing the MIDI events that you want Windows to time out and output is similiar to passing a buffer of data to midiOutLongMsg(). In fact, you use a MIDIHDR structure, as well as calls to midiOutPrepareHeader() and midiOutUnprepareHeader(). But, you also use one or more MIDIEVENT structures, and the actual time out and output of the MIDI messages doesn't begin until you call midiStreamRestart(). (ie, Until you actually call midiStreamRestart, Windows simply queues the MIDI messages in the order you pass them).

Here's how you pass a MIDI message to be played back:

1. Place one MIDI message into a structure called a MIDIEVENT.

2. Place a pointer to this MIDIEVENT into the lpData field of a MIDIHDR structure. Also set the MIDIHDR's dwBufferLength and dwBytesRecorded fields to the size of this MIDIEVENT structure. Set the dwFlags field to 0.

3. Pass the MIDIHDR to midiOutPrepareHeader().

4. Pass the MIDIHDR to midiStreamOut().

The MIDI message is now queued for playback. If you haven't called midiStreamRestart() yet, then Windows does not yet start timing out and outputting this event. So, you can queue several MIDI messages prior to the start of playback by repeating the above steps, using a separate MIDIEVENT and MIDIHDR for each message.

It's also possible to queue several MIDI messages using one MIDIHDR and one call to midiOutPrepareHeader() and midiStreamOut(). You do this by using an array of MIDIEVENT structures (ie, one for each of your MIDI messages). Place each MIDI message into one of the MIDIEVENT structures in that array. Then place a pointer to the entire array in the MIDIHDR's lpData field. Of course, the MIDIHDR's dwBufferLength and dwBytesRecorded fields are set to the size of the entire array of MIDIEVENT structures. It is much more efficient and memory-conserving to combine all MIDI messages that occur upon the same musical beat into one array of MIDIEVENTS, and cue them with one MIDIHDR.

Note that after Windows finishes playing all of the events queued with a particular MIDIHDR, it sets the MHDR_DONE bit of the dwFlags field. This may be very useful later when you're dealing with processing MIDIHDRs as a result of being notified by Windows that a MIDIHDR's events have finished playing. Also note that a MIDIHDR has a dwUser field that you can use for your own purposes.

Now it's certainly feasible to queue up all of the MIDI messages of a sequence prior to calling midiStreamRestart() by simply putting them into one gigantic array of MIDIEVENT structures, and passing the whole array in one call to midiStreamOut(). You can download my Simple Stream C example which illustrates this approach. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. My code takes this approach in order to present the simplest possible example of using the Stream API to play a musical sequence, and therefore give you an easy introduction to how the Stream API works.

But that approach is sort of like using the High level MIDI API. After all, by passing all of the data all at once, you then lose control over too much of the playback process. The whole point of the Stream API is to feed Windows a few events at a time so that the playback doesn't get too far ahead that you lose the sense of real-time control over individual events. If you want to be doing real-time mixing of several "tracks" of MIDI messages into a single stream, giving the user the option to mute one of the tracks at any time for example, then obviously you don't want to queue MIDI messages too far in advance.

But, if you want a smooth playback, you obviously have to queue those events before they need to be played. The best approach to take is to use a sort of double-buffering scheme. In other words, you'll always have one "block" of MIDI messages queued while the current block of MIDI messages is playing. When the current block is finished playing (and the queued block starts playing), then you'll queue another block.

You'll select a brief "time window", for example 1 quarter note. (ie, Assume our "musical beat" is a quarter note). Then, before playback is started with midiStreamRestart(), you'll place all of the MIDI messages whose timing falls within the first quarter note (ie, the first beat) into an array of MIDIEVENTS (and a MIDIHDR) and queue it with midiStreamOut(). You'll also place

all of the MIDI messages whose timing falls within the second quarter note (ie, second beat) into another array of MIDIEVENTS (and another MIDIHDR) and queue it with midiStreamOut(). Then, you'll call midiStreamRestart() to start playback. As soon as the last MIDI message in the first array of MIDIEVENTS is finished playing, you'll place all of the MIDI messages whose timing falls within the third quarter note into that same array of MIDIEVENTS and queue it with midiStreamOut(). Of course, while you're doing this, the second queued array has been playing. After that array's last MIDI message is played, you'll use that array to queue another "block" of MIDI messages whose timing falls within the fourth quarter note. Etc. In this way, you always have data queued for continuous, smooth playback, but the playback is only 1 musical beat ahead of any real-time action the user instructs you to perform upon the data. So, for example, if he tells you to mute one of the tracks, maybe it won't effectively work out that way for one musical beat (some events on that track may already be queued for the next beat), but that's plenty close enough.

So what if there happens to be no MIDI events that fall within the next musical beat? What do you queue? As you'll see in the next section, you can queue a single NOP event, timed to delay for an entire beat.

The MIDIEVENT structure

Now let's take a closer look at the MIDIEVENT structure to see how various MIDI messages are stored in it. The Stream API documents it as so:

```
typedef struct {
    DWORD dwDeltaTime;
    DWORD dwStreamID;
    DWORD dwEvent;
    DWORD dwParms[];
} MIDIEVENT;
```

The dwDeltaTime field is just like the timing in MIDI File Format (except that it's an unsigned long rather than a variable length quantity). It represents the amount of time to delay before outputting this MIDI message. You can specify the time in terms of PPQN clocks, or a SMPTE time. (When using the former, you can send "Tempo Events" to the stream device to change the tempo, or call a Stream function that allows you to change the tempo on the fly. When using the latter, Windows can sync the MIDI playback to streaming video or other SMPTE cues).

Currently, the dwStreamID field isn't used and should be set to 0. (My own tests show that whatever value you stuff into this field is ignored, but unless you're using a driver that directly supports the MIDI Stream API and the driver uses this field, it is safest to set it to 0 in case some future version of Windows utilizes this field).

Although declared as an unsigned long, the 4 bytes of the dwEvent field are actually individual pieces of information. The highest byte contains some flag bits and some bits that form an "event type" value. I'll refer to this as the Event Type byte.

If this MIDIEVENT contains a normal MIDI Voice message such as a Note-On, Program Change, Aftertouch, or any of the other MIDI messages that are 3 or less bytes, then the Event Type byte is set to the value MEVT_SHORTMSG.

In this case, the remaining 3 bytes of this field are the MIDI Status byte, the first MIDI data byte (if any), and the second MIDI data byte (if any). Note that this is packed up exactly the way that a MIDI message is passed to midiOutShortMsg(). Do not use running status. The stream device will implement running status when it outputs the MIDI messages.

Here then is how you would initialize a MIDIEVENT with a Note-On MIDI message for middle C on channel 1 (with velocity of 0x40) and a delta time of 0 (gets output as soon as Windows finishes playing the previously queued event):

```
MIDIEVENT mevt;

mevt.dwDeltaTime = 0;
mevt.dwStreamID = 0;
```

```
mevt.dwEvent = ((unsigned long)MEVT_SHORTMSG<<24) | 0x00403C90;
```

Note that since MEVT_SHORTMSG is really 0x00, we can simply the above by actually removing the setting of the Event Type byte (since its already 0 in our packed MIDI message):

mevt.dwEvent = 0x00403C90;

What about that dwParms field? Well, if you look closely at the declaration, it isn't really there. There's no size for this dwParms array. Say what?!? That's right. There is no dwParms field on the above MIDIEVENT. The structure really has only 3 fields. It should have been declared as something like this (which I'll call MY_MIDI_EVT):

```
typedef struct {
    DWORD dwDeltaTime;
    DWORD dwStreamID;
    DWORD dwEvent;
} MY_MIDI_EVT;
```

So what was that dwParms field doing there? Well, here's where it gets tricky. There are other types of events that you can pass. For example, consider a System Exclusive event. It can have many more than 3 bytes. Where do you put all of those bytes? Well, **now** you use that extra dwParms array appended to the end. Its size is set to however many bytes you need to pass. In essense, you're appending the data to the end of the MIDIEVENT structure. That's right. The size of the MIDIEVENT structure you pass will vary depending upon the type of event it contains.

Let's take an example where we want to pass a System Exclusive event that consists of 7 bytes. Well, the first thing that we need to do is redeclare the MIDIEVENT structure, thereby creating a new structure. I'll call this MY_SYSEX_EVT.

```
typedef struct {
    DWORD         dwDeltaTime;
    DWORD         dwStreamID;
    DWORD         dwEvent;
    unsigned char dwParams[8];
} MY_SYSEX_EVT;
```

See? I declared an array large enough to hold my extra bytes. Why did I use a size of 8 rather than 7? Well, the Stream API requires that the size of all MIDIEVENT structures passed to it be aligned on a doubleword boundary. So, if I'm going to stuff this MIDIEVENT into a buffer containing another MIDIEVENT that comes after it, I want that second MIDIEVENT to be properly aligned. In essense, I've added a pad byte above to make the structure's size a multiple of 4, and thereby ensure that subsequent MIDIEVENT structures in the same array are properly aligned.

OK, now let's initialize it. For such a MIDIEVENT, its Event Type byte must be the value MEVT_LONGMSG. Furthermore, the remaining 3 bytes of the dwEvent field must be a 24-bit count of the number of bytes in our System Exclusive message (ie, 7 in this case).

```
MY_SYSEX_EVT  mevt;
unsigned char sysEx[] = {0xF0, 0x7F, 0x7F, 0x01, 0x02, 0xF7};

mevt.dwDeltaTime = 100; /* Just for the hell of it, delay 100 clocks before sending
it */
mevt.dwStreamID = 0;
mevt.dwEvent = ((unsigned long)MEVT_LONGMSG<<24) | sizeof(sysEx);
memcpy(&mevt.dwParams[0], &sysEx[0], sizeof(sysEx));
```

What other Event Types are there (besides the "short" type for MIDI messages of 3 bytes or less, and the "long" type for System Exclusive)? The most useful is a Tempo event. For such a MIDIEVENT, its Event Type byte must be the value MEVT_TEMPO. Furthermore, the remaining 3 bytes of the dwEvent field must be a Tempo value as a 24-bit value. It is expressed in microseconds per quarter note, just like in the MIDI File Format's MetaTempo event.

Another Event Type is a comment. Like a System Exclusive MIDIEVENT, the characters that form the comment are appended to the end of the MIDIEVENT structure itself. Its Event Type byte must be the value MEVT_COMMENT. Furthermore, the remaining 3 bytes of the dwEvent field must be a 24-bit count of the number of chars in the comment. Again, pad out the structure's size to a multiple of 4 if you're going to put it in an array with other MIDIEVENTs after it. Windows ignores comment events, so you can use them for your own purposes.

Another Event Type is a NOP (no operation). It's a "short" type of event, like Tempo and regular MIDI Voice messages, so no extra fields are appended to the MIDIEVENT structure. Its Event Type byte must be the value MEVT_NOP. Furthermore, the remaining 3 bytes of the dwEvent field can be used for any purpose you wish. Windows ignores NOP events, so you can use them for your own purposes.

The final Event Type is a version event. It has a MIDISTRMBUFFVER structure appended to the end of the MIDIEVENT structure. This extra structure just contains version information about the Stream. Its Event Type byte must be the value MEVT_VERSION. Furthermore, the remaining 3 bytes of the dwEvent field must be a 24-bit count of the size of the MIDISTRMBUFFVER structure.

OK, I'm sure that there is one nagging question in your mind. If MIDIEVENT structures can be variable size, depending upon the Event Type, then how do you declare a static array of them? Well, you can't really (unless you happen to stick to only using the "short" types). What you'll need to do is just copy them into one large char buffer, using a pointer to that buffer, and do some creative casting. For example, here I copy two MIDIEVENTs into one buffer:

```
MY_SYSEX_EVT * xevt;
MY_MIDI_EVT *  mevt;
unsigned char  sysEx[] = {0xF0, 0x7F, 0x7F, 0x01, 0x02, 0xF7};
unsigned char  buffer[20];

/* Format for a Note-On */
mevt = (MY_MIDI_EVT *)&buffer[0];
mevt->dwDeltaTime = 0;
mevt->dwStreamID = 0;
mevt->dwEvent = 0x00403C90;
mevt++;

/* Format for a System Exclusive */
xevt = (MY_SYSEX_EVT *)mevt;
xevt->dwDeltaTime = 100;
xevt->dwStreamID = 0;
xevt->dwEvent = ((unsigned long)MEVT_LONGMSG<<24) | sizeof(sysEx);
memcpy(&xevt->dwParams[0], &sysEx[0], sizeof(sysEx));
```

Of course, I should make sure that buffer is aligned upon a doubleword boundary (which you can do with your compiler's alignment directive).

But another approach to take is to simply declare your array to be an array of unsigned longs. After all, since all MIDIEVENT structures need to be padded out to a multiple of 4 bytes, then the net result is that each MIDIEVENT structure consists 3 or more unsigned longs. So, here's the above array initialized this way:

```
unsigned long  buffer[] = { 0, 0, 0x00403C90, /* The Note-On */
                            0, 0, ((unsigned long)MEVT_LONGMSG<<24) | 7, 0x017F7FF0,
0x00F70201}; /* The SysEx */
```

Note the order of the packed SysEx bytes. Each unsigned long contains the next 4 bytes, and remember that we're using little endian order on each unsigned long.

Starting playback

You start playback by calling midiStreamRestart() as so:

```
unsigned long   err;

err = midiStreamRestart(outHandle);
if (err)
{
    printf("An error starting playback!\n");
}
```

Note that when playback begins, the stream device's current time is set to 0 (if the playback hasn't been paused).

Stop the current playback

You can stop a playback in progress by calling midiStreamStop() as so:

```
unsigned long   err;

err = midiStreamStop(outHandle);
if (err)
{
    printf("An error stopping playback!\n");
}
```

This flushes all of the queued MIDIEVENTs, and you can subsequently midiOutUnprepareHeader() the MIDIHDRs. (The MHDR_DONE bit is set in the dwFlags field of all queued MIDIHDR structures).

It also turns off any notes that are still turned on. (By contrast, midiOutReset() turns off all notes regardless, and is more of a "panic" button type of response to turn off any "stuck notes").

Calling this function when the output is already stopped has no effect, and doesn't return an error.

WARNING! WARNING! WARNING! The Windows Stream Manager appears to be severely broken. Calling midiStreamStop() (or midiStreamPause) does indeed stop playback, but it appears to also close the stream handle that you pass to it. The result is that a subsequent call to another function using that handle results in the Stream Manager returning an error that the handle is no longer valid. The only options you have are:

1. Open a stream device immediately before you intend to queue and play MIDI events. Play those events. Then close the stream device.

2. Once you call midiStreamRestart(), never call midiStreamStop() or midiStreamPause(). Simply maintain your own clock variable and "play flag" variable that your callback can test to see whether it should continue queueing MIDI events for playback and increment its clock, or just do nothing. In other words, you're going to start up the stream once and leave it constantly running in the background, waiting for more MIDIEVENTs to be queued.

Pause/Resume the current playback

You can pause a playback in progress by calling midiStreamPause() as so:

```
unsigned long   err;
```

```
err = midiStreamPause(outHandle);
if (err)
{
    printf("An error pausing playback!\n");
}
```

If you wish to subsequently resume playback from the point at which it was paused, then call midiStreamRestart(). The stream device's time is not reset to 0, and playback resumes. If instead, you wish to stop the device and flush the queued MIDIEVENTs (perhaps in order to reset its time to 0 and start playback from the beginning of a sequence), then instead call midiStreamStop().

Calling this function when the output is already paused has no effect, and doesn't return an error.

Setting/Querying the Tempo and Timebase

Before playing queued MIDIEVENTs, you'll want to set the Timebase for the stream device. This is equivalent to the MIDI File Format's Division. It tells the stream device how to scale the dwDeltaTime field of each MIDIEVENT. (ie, Consider it a SMPTE time in 30 fps, or a time-stamp at 96 PPQN, or a time-stamp at 120 PPQN, etc). You use midiStreamProperty() with the MIDIPROP_SET and MIDIPROP_TIMEDIV flags as the third arg. The second arg is a pointer to a MIDIPROPTIMEDIV structure whose dwTimeDiv field is initialized to your desired Timebase. For PPQN Timebases, just specify the PPQN value, for example 96 PPQN. For SMPTE, the low two bytes are as per the MIDI File Format's Division field when SMPTE is specified.

Here I set the Timebase to 96 PPQN (which is the default if you don't specify a Timebase):

```
unsigned long   err;
MIDIPROPTIMEDIV prop;

prop.cbStruct = sizeof(MIDIPROPTIMEDIV);
prop.dwTimeDiv = 96;
err = midiStreamProperty(outHandle, (LPBYTE)&prop, MIDIPROP_SET|MIDIPROP_TIMEDIV);
if (err)
{
    printf("An error setting the timebase!\n");
}
```

Besides putting Tempo events in the stream in order to set tempo at any given time, you can also call midiStreamProperty() with the MIDIPROP_SET and MIDIPROP_TEMPO flags as the third arg to set tempo. The second arg is a pointer to a MIDIPROPTEMPO structure whose dwTempo field is initialized to your desired Tempo. Note that this is irrelevant when using a SMPTE Timebase.

Here I set the Tempo to 120 BPM (which is the default if you don't specify a Tempo):

```
unsigned long   err;
MIDIPROPTEMPO   prop;

prop.cbStruct = sizeof(MIDIPROPTEMPO);
prop.dwTempo = 0x0007A120;
err = midiStreamProperty(outHandle, (LPBYTE)&prop, MIDIPROP_SET|MIDIPROP_TEMPO);
if (err)
{
    printf("An error setting the tempo!\n");
}
```

Of course, you can query a stream device's current timebase or tempo by using the MIDIPROP_GET flag instead of MIDIPROP_SET. Windows fills in the MIDIPROPTIMEDIV or MIDIPROPTEMPO you pass.

Here I query the current Tempo:

```
unsigned long   err;
MIDIPROPTEMPO   prop;

prop.cbStruct = sizeof(MIDIPROPTEMPO);
err = midiStreamProperty(outHandle, (LPBYTE)&prop, MIDIPROP_GET|MIDIPROP_TEMPO);
if (err)
{
    printf("An error requesting the tempo!\n");
}
else
{
    printf("Tempo = %u\n", prop.dwTempo);
}
```

Notification during playback

I previously talked about how you want to queue up another array of MIDIEVENTs as soon as one array finishes playing. So how do you receive such notification from Windows? When you call midiStreamOpen(), you can tell it how you want Windows to notify you. In our above example, we specified CALLBACK_NULL (ie, we didn't want Windows to notify us). But there are other choices as follows:

1. CALLBACK_EVENT -- You allocate some Event with CreateEvent(), and Windows uses this to signal your app. (ie, Your app can wait on that Event signal, for example with WaitForSingleObject). You pass the handle of the Event as the 4th arg to midiStreamOpen().

2. CALLBACK_THREAD -- Windows causes some suspended thread within your app to run. (ie, Your app's thread can suspend itself via SuspendThread). You pass the Thread ID of the desired thread to be run as the 4th arg to midiStreamOpen().

3. CALLBACK_WINDOW -- Windows sends a message to some open window in your app. The parameters for the message will contain additional information about what caused Windows to send that message. You pass the desired window's handle as the 4th arg to midiStreamOpen().

4. CALLBACK_FUNCTION -- Windows directly calls some function in your app. It passes args that contain additional information about what caused Windows to call your function. You pass a pointer to the desired function as the 4th arg to midiStreamOpen(). The 5th arg to midiStreamOpen() can be anything you desire, and this will be passed to your callback function each time that Windows calls your callback.

The latter two methods allow you to better determine what exactly caused Windows to notify you, because they supply additional information to you.

So when does Windows notify you? Here are the times when Windows notifies you:

1. When you open a Stream device via midiStreamOpen().

2. When you close a Stream Device via midiStreamClose().

3. When midiStreamOut() encounters an event that has its MEVT_F_CALLBACK flag set.

4. When midiStreamOut() finishes playing a MIDIHDR's block of data.

Here's an example of setting the MEVT_F_CALLBACK flag of an event.

```
xevt->dwEvent = ((unsigned long)MEVT_LONGMSG<<24) | sizeof(sysEx) | MEVT_F_CALLBACK;
```

Now, after Windows plays that above event, it notifies you. For example, if you've chosen CALLBACK_FUNCTION method, Windows calls your callback function.

Windows also will notify you when the last event in a MIDIHDR's block of events is played. This is the point at which you will queue the next array of MIDIEVENTs using that same array (and its associated MIDIHDR).

In fact, you could strategically place NOP events with their dwDeltaTime set so that they just happen to fall upon each musical downbeat, and set the MEVT_F_CALLBACK flag of each of those event's dwEvent fields. Then, everytime that your callback is called, you can update a graphical display counting off the beat. (My Stream Callback C example shows that using CALLBACK_FUNCTION method).

Here's an example of using the CALLBACK_EVENT method to play an array of MIDIEVENTs. This is a complete example that shows you all the bare minimum details you need to know to play a stream of MIDIEVENTs.

```
/* The array of MIDIEVENTs to be output. We only have 2 */
unsigned long myNotes[] = {0, 0, 0x007F3C90, /* A note-on */
192, 0, 0x00003C90}; /* A note-off. It's the last event in the array */

HANDLE          event;
HMIDISTRM       outHandle;
MIDIHDR         midiHdr;
MIDIPROPTIMEDIV prop;
unsigned long   err;

/* Allocate an Event signal */
if ((event = CreateEvent(0, FALSE, FALSE, 0)))
{
    /* Open default MIDI Out stream device. Tell it to notify via CALLBACK_EVENT and
use my created Event */
    err = 0;
    if (!(err = midiStreamOpen(&outHandle, &err, 1, (DWORD)event, 0,
CALLBACK_EVENT)))
    {
        /* Windows signals me once when the driver is opened. Clear that now */
        ResetEvent(event);

        /* Set the timebase. Here I use 96 PPQN */
        prop.cbStruct = sizeof(MIDIPROPTIMEDIV);
        prop.dwTimeDiv = 96;
        midiStreamProperty(outHandle, (LPBYTE)&prop, MIDIPROP_SET|MIDIPROP_TIMEDIV);

        /* If you wanted something other than 120 BPM, here you should also set the
tempo */

        /* Store pointer to our stream (ie, array) of messages in MIDIHDR */
        midiHdr.lpData = (LPBYTE)&myNotes[0];
```

```
        /* Store its size in the MIDIHDR */
        midiHdr.dwBufferLength = midiHdr.dwBytesRecorded = sizeof(myNotes);

        /* Flags must be set to 0 */
        midiHdr.dwFlags = 0;

        /* Prepare the buffer and MIDIHDR */
        err = midiOutPrepareHeader(outHandle,  &midiHdr, sizeof(MIDIHDR));
        if (!err)
        {
            /* Queue the Stream of messages. Output doesn't actually start
               until we later call midiStreamRestart().
             */
            err = midiStreamOut(outHandle, &midiHdr, sizeof(MIDIHDR));
            if (!err)
            {
                /* Start outputting the Stream of messages. This will return
immediately
                   as the stream device will time out and output the messages on its
own in
                   the background.
                 */
                err = midiStreamRestart(outHandle);
                if (!err)

                    /* Wait for playback to stop. Windows signals me using that Event
I created */
                    WaitForSingleObject(event, INFINITE);
            }

            /* Unprepare the buffer and MIDIHDR */
            midiOutUnprepareHeader(outHandle, &midiHdr, sizeof(MIDIHDR));
        }

        /* Close the MIDI Stream */
        midiStreamClose(outHandle);
    }

    /* Free the Event */
    CloseHandle(event);
}
```

If using the CALLBACK_FUNCTION method, then you need to write a function that has the following declaration (although you can name the function anything you like):

```
void CALLBACK midiCallback(HMIDIOUT handle, UINT uMsg, DWORD dwInstance, DWORD
dwParam1, DWORD dwParam2);
```

As mentioned, you pass a pointer to this function as the 4th arg to midiStreamOpen(). The 5th arg to midiStreamOpen() can be anything you desire, and this will be passed to your callback function each time that Windows calls your callback. Windows calls your function whenever 1 of 4 possible things happen:

1. When you open a Stream device via midiStreamOpen(). In this case, the uMsg arg to your callback will be MOM_OPEN. The handle arg will be the same as what is returned to midiStreamOpen(). The dwInstance arg is whatever I passed to midiStreamOpen() as its dwInstance arg.

2. When you close a Stream Device via midiStreamClose().In this case, the uMsg arg to your callback will be MOM_CLOSE. The handle arg will be the same as what was passed to midiStreamClose(). The dwInstance arg is the 5th arg you passed to midiStreamOpen() when you initially opened this handle.

3. When midiStreamOut() encounters an event that has its MEVT_F_CALLBACK flag set. In this case, the uMsg arg to your callback will be MOM_POSITIONCB. The handle arg will be the same as what is passed to midiStreamOut(). The dwInstance arg is the 5th arg you passed to midiStreamOpen() when you initially opened this handle. The dwParam1 arg points to the MIDIHDR that you passed to midiStreamOut(). The dwOffset field of the MIDIHDR will indicate the byte offset into the buffer for the event that caused this callback to be called. See my note below.

4. When midiStreamOut() finishes playing a MIDIHDR's block of data. In this case, the uMsg arg to my callback will be MOM_DONE. The handle arg will be the same as what is passed to midiStreamOut(). The dwInstance arg is the 5th arg you passed to midiStreamOpen() when you initially opened this handle. The dwParam1 arg points to the MIDIHDR that you passed to midiStreamOut().

NOTE: The dwParam2 arg is not used. This is reserved for future use.

Here's an example of using the CALLBACK_FUNCTION method to play an array of MIDIEVENTs. This is a complete example that shows you all the bare minimum details you need to know to play a stream of MIDIEVENTs.

```
/* The array of MIDIEVENTs to be output. We only have 2 */
unsigned long myNotes[] = {0, 0, 0x007F3C90, /* A note-on */
192, 0, 0x00003C90}; /* A note-off. It's the last event in the array */

HANDLE    Event;

void CALLBACK midiCallback(HMIDIOUT handle, UINT uMsg, DWORD dwInstance, DWORD
dwParam1, DWORD dwParam2)
{
    LPMIDIHDR   lpMIDIHeader;
    MIDIEVENT * lpMIDIEvent;

    /* Determine why Windows called me */
    switch (uMsg)
    {
        /* Got some event with its MEVT_F_CALLBACK flag set */
        case MOM_POSITIONCB:

            /* Assign address of MIDIHDR to a LPMIDIHDR variable. Makes it easier to
access the
               field that contains the pointer to our block of MIDI events */
            lpMIDIHeader = (LPMIDIHDR)dwParam1;

            /* Get address of the MIDI event that caused this call */
            lpMIDIEvent = (MIDIEVENT *)&(lpMIDIHeader->lpData[lpMIDIHeader-
>dwOffset]);

            /* Normally, if you had several different types of events with the
               MEVT_F_CALLBACK flag set, you'd likely now do a switch on the highest
               byte of the dwEvent field, assuming that you need to do different
               things for different types of events.
            */

            break;

        /* The last event in the MIDIHDR has played */
        case MOM_DONE:
```

```
            /* Wake up main() */
            SetEvent(Event);

            break;


        /* Process these messages if you desire */
        case MOM_OPEN:
        case MOM_CLOSE:

            break;
    }
}

int main(int argc, char **argv)
{
    HMIDISTRM       outHandle;
    MIDIHDR         midiHdr;
    MIDIPROPTIMEDIV prop;
    unsigned long   err;

    /* Allocate an Event signal */
    if ((event = CreateEvent(0, FALSE, FALSE, 0)))
    {
        /* Open default MIDI Out stream device. Tell it to notify via CALLBACK_EVENT
and use my created Event */
        err = 0;
        if (!(err = midiStreamOpen(&outHandle, &err, 1, (DWORD)midiCallback, 0,
CALLBACK_FUNCTION)))
        {
            /* Set the timebase. Here I use 96 PPQN */
            prop.cbStruct = sizeof(MIDIPROPTIMEDIV);
            prop.dwTimeDiv = 96;
            midiStreamProperty(outHandle, (LPBYTE)&prop,
MIDIPROP_SET|MIDIPROP_TIMEDIV);

            /* If you wanted something other than 120 BPM, here you should also set
the tempo */

            /* Store pointer to our stream (ie, array) of messages in MIDIHDR */
            midiHdr.lpData = (LPBYTE)&myNotes[0];

            /* Store its size in the MIDIHDR */
            midiHdr.dwBufferLength = midiHdr.dwBytesRecorded = sizeof(myNotes);

            /* Flags must be set to 0 */
            midiHdr.dwFlags = 0;

            /* Prepare the buffer and MIDIHDR */
            err = midiOutPrepareHeader(outHandle,  &midiHdr, sizeof(MIDIHDR));
            if (!err)
            {
                /* Queue the Stream of messages. Output doesn't actually start
                   until we later call midiStreamRestart().
                 */
                err = midiStreamOut(outHandle, &midiHdr, sizeof(MIDIHDR));
                if (!err)
                {
                    /* Start outputting the Stream of messages. This will return
```

```
immediately
                        as the stream device will time out and output the messages on
its own in
                        the background.
                     */
                    err = midiStreamRestart(outHandle);
                    if (!err)

                    /* Wait for playback to stop. Windows calls my callback, which
will set this signal */
                        WaitForSingleObject(event, INFINITE);
                }

                /* Unprepare the buffer and MIDIHDR */
                midiOutUnprepareHeader(outHandle, &midiHdr, sizeof(MIDIHDR));
            }

            /* Close the MIDI Stream */
            midiStreamClose(outHandle);
        }

        /* Free the Event */
        CloseHandle(event);
    }

        return(0);
}
```

NOTE: If you happen to have two or more events that occur upon the same beat (or close enough such that Windows doesn't even have time to call your callback once before both events time out), and both their MEVT_F_CALLBACK flags are set, then Windows only calls your callback once for all of those events. The dwOffset of the MIDIHDR would reference the last such event. Therefore, don't bother setting the MEVT_F_CALLBACK flag of more than one event that occurs upon a given time.

Furthermore, while Windows calls your callback, it continues timing out the next event in the background. If it happens that the next event happens to have its MEVT_F_CALLBACK flag also set, and it times out while your callback is still processing that previous MEVT_F_CALLBACK event, then Windows will call your callback again. In other words, if you're not sure that you've placed enough time inbetween each of your MEVT_F_CALLBACK flagged events in order to do what you need to get done in your callback, then you had better make sure your callback handles reentrancy properly. (ie, Don't use global variables whose value changes, or use some sort of mechanism to arbitrate access to those variables such as a Mutex). For practical purposes, you should avoid setting the MEVT_F_CALLBACK flag of an event whose time makes it occur less than 25 milliseconds after the previous event with its MEVT_F_CALLBACK flag set. Microsoft recommends the following approach to handling a situation where MEVT_F_CALLBACK events are timing out quicker than your callback can finish its work: "The callback should use the MIDIHDR's dwOffset to decide what to do. If it is expecting this to reference a particular location (ie, event) but it instead references a location somewhere further along in the array of MIDIEVENTs, it means that your callback is not keeping up or that the MEVT_F_CALLBACK flagged events are scheduled too closely together for the capabilities of the computer. The best thing to do in this case is to just throw the callback away because there will be another one in the queue that will be delivered just as soon as processing of the current callback is completed."

Querying the current playback time

You can retrieve the current playback time by calling midiStreamPosition(). It fills in an MMTIME structure.

Before calling midiStreamPosition(), you set the wType field of the MMTIME structure to indicate the time format you desire

returned. After calling midiStreamPosition(), you should check the wType field. Some stream devices may not support returning certain time formats, for example, a SMPTE format. In that case, midiStreamPosition() will set the wType field to the closest supported format, and return that time.

The allowable time formats (ie, for the wType field) are:

```
TIME_BYTES      Current byte offset from beginning of the playback.
TIME_MIDI       MIDI time (ie, MIDI Song Position Pointer).
TIME_MS         Time in milliseconds.
TIME_SAMPLES    Number of waveform-audio samples (for syncing to WAVE playback).
TIME_SMPTE      SMPTE time.
TIME_TICKS      PPQN clocks.
```

Here I query the current Time. (Note that the Time is set to 0 when playback starts, except for a paused playback that is resumed).

```
unsigned long   err;
MMTIME          time;

/* Request millisecond time returned */
time.wType = TIME_MS;
err = midiStreamPosition(outHandle, &time, sizeof(MMTIME));
if (err)
{
    printf("An error requesting the time!\n");
}
else switch (time.wType)
{
    case TIME_BYTES:
        printf("%u bytes have been played so far.\n", time.cb);
        break;

    case TIME_MIDI:
        printf("Midi Song Position Pointer = %u\n", time.midi.songptrpos);
        break;

    case TIME_MS:
        printf("Millisecond Time = %u\n", time.ms);
        break;

    case TIME_SAMPLES:
        printf("%u digital audio samples have been played so far.\n", time.sample);
        break;

    case TIME_SMPTE:
        printf("SMPTE Time (%u fps) = %u:%u:%u:%u\n", time.smpte.fps,
time.smpte.hour, time.smpte.min, time.smpte.sec, time.smpte.frame);
        break;

    case TIME_TICKS:
        printf("PPQN Clocks = %u\n", time.ticks);
}
```

Outputting MIDI data immediately while a Stream is playing

Note that you can call midiOutShortMsg() or midiOutLongMsg() while the Stream (ie, queued MIDI events) is playing. In such a case, the MIDI data you send via these two functions will be output as soon as possible. (But don't utilize running status

with midiOutShortMsg. Always specify the full MIDI message with Status).

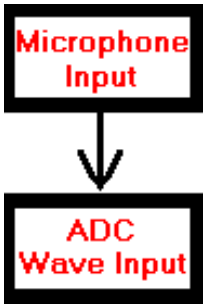
Stream recording

Currently, the Stream API does not support recording. If you need to do MIDI recording, you'll have to use the Low level MIDI API and a MultiMedia Timer instead of the Stream API.

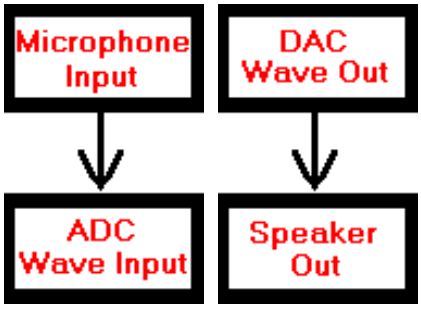
This information is not yet completed.

In order to understand how the Mixer API works, it's important to first understand the hardware layout of a typical audio card. It's necessary to be able to visualize an audio card as having distinct, but interconnected, components upon it.

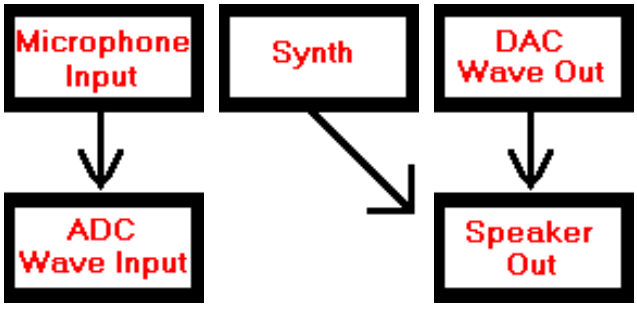
Let's consider a typical, basic audio card. First of all, if the audio card is capable of recording digital audio, then it typically has a microphone input jack (with some sort of pre-amp), and it also has an analog-to-digital converter (ADC) to convert that analog microphone signal to a digital stream. Therefore, it has two components -- the Microphone input component, and the ADC component. The Microphone input is piped into the ADC. So, we can represent the layout with the following block diagram showing two components, with the signal flow between them (ie, the arrow)



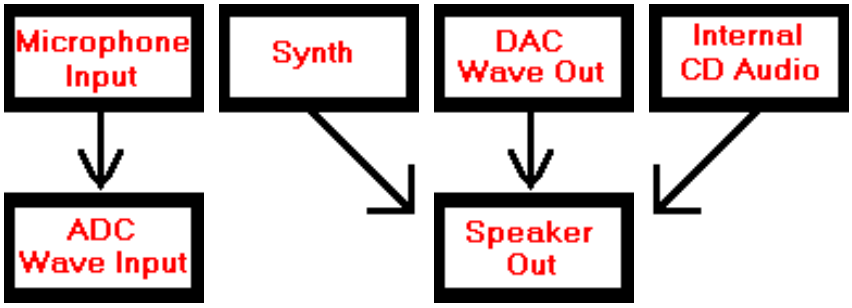
A typical audio card is also capable of playing back digital audio, so it has a DAC to convert the digital stream back to an analog signal, and also it has a speaker output jack (ie, with some sort of analog amplifier). Therefore, it has two more components -- the DAC component, and the Speaker component. The DAC output is piped to the speakers.



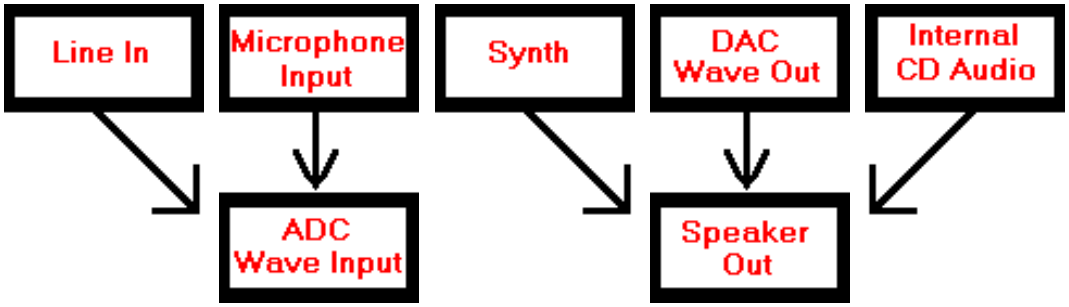
A typical audio card may have some other components. For example, it may have a built-in sound module (ie, synth) capable of playing MIDI data. The audio output of this component would typically be piped to the speaker output just like the DAC. So, our block diagram now looks like so:



So too, a typical audio card has a connector to (internally) attach the audio out of the computer's CDROM drive (so that an audio CD played in that CDROM drive will sound through the computer's speakers). This component is also piped to the speaker output, just like the Synth and DAC. Now, our block diagram looks like this:



Finally, let's assume that this audio card has a Line In component so that the audio signal from an external tape deck or musical instrument or external hardware mixer can be attached to this jack and digitized. This component is piped to the ADC component just like the Microphone Input component. Here is our finished block diagram which contains 7 components (and 5 signal flows -- the arrows that interconnect them):



Typically, each one of these components has its own, individual parameters. For example, the Synth will usually have its own volume (gain) level. The Internal CD Audio will have its own volume level. And the DAC (ie, digital audio, or WAVE playback) will have its own volume level. In this way, if the user is playing an audio CD, playing a MIDI file, and playing back a WAVE file, simultaneously, he can balance out the volumes of all 3 components as they are being output to the speaker jack. So too, the speaker component will typically have its own volume -- a Master volume that affects the overall mix of the 3 components piped to the speaker out.

Likewise, the Line In and Microphone Input typically have separate volume levels so they can be balanced when recording simultaneously from both jacks. And the ADC may have some sort of Master Volume which affects the overall recording level of the 2 components piped to it.

A given component may have other parameters that are controllable. For example, each of the above components may have its own Mute switch so that the component's sound can be quickly turned on/off.

The Mixer device

A given audio card has one Mixer device associated with it. All of the various components on that card are controlled through that card's one Mixer device. The Windows Mixer API is used to access the card's Mixer device. The Mixer API has functions to get a listing of all the various components on a particular card, and to adjust all of their parameters. This is a new API added to Win95/98 and WinNT (4.X and above), although an add-on to Windows 3.1 makes it available to that older OS.

NOTE: A card's device driver needs extra support to work with the Mixer API. Not all Win95 and WinNT drivers have this support. Win3.1 drivers typically do not.

In any computer, there can be more than one installed audio card. You already discovered that Windows maintains lists of all of the WAVE and MIDI input/output devices in a system. Since each installed audio card has its own Mixer device too (as long as its driver supports such), Windows also maintains a list of Mixer devices installed in a system. So for example, if you have two audio cards installed in a given system, then there should be two Mixer devices installed as well (assuming that the drivers for both audio cards support the mixer API).

Just like with the WAVE and MIDI input/output devices, Windows assigns a numeric ID to each Mixer device. So, the Mixer

device with an ID of 0 is the first (default) mixer in the system. If there is a second audio card, then there should be a second Mixer device with an ID of 1.

Just like with other devices, in order to use a Mixer device, you must first open it (with mixerOpen()), and then you may call other Mixer APIs to control the card's line inputs/outputs. When finished, you must close the device (with mixerClose())

Opening a Mixer device

How does your program choose a mixer device to manipulate? There are several different approaches you can take, depending upon how fancy and flexible you want your program to be.

If you simply want to open the preferred Mixer device, then use a Device ID of 0 with mixerOpen() as so:

```
unsigned long err;
HMIXER        mixerHandle;

/* Open the mixer associated with the default
   Audio/MIDI card in the computer */
err = mixerOpen(&mixerHandle, 0, 0, 0, 0);
if (err)
{
    printf("ERROR: Can't open Mixer Device! -- %08X\n", err);
}
else
{
    /* The mixer device is now open, and its
      handle is in the variable 'mixerHandle'.
      You may now use it with other Mixer API
      functions */
}
```

Of course, if the user has no Mixer device installed, the above call returns an error, so always check that return value. (The expected error numbers from the Mixer API's are listed in MMSYSTEM.H. Unfortunately, unlike with the Wave and Midi low level API's, there is no API function to translate these error numbers into strings to present to the user).

So what actually is the preferred Mixer device? Well, that's whatever Mixer device happened to have been installed first in a system. If there is only one audio card in the system, then it's a good bet that you have the mixer device you want. But, what if you're trying to use the Wave Output on a second audio card? You definitely don't want to be using the Mixer device for the first card to control the second card's Wave Out volume. (The first card's Mixer doesn't control the second card's Wave Output).

So, how do you open the Mixer for the desired card? Fortunately, mixerOpen() allows you to pass the device ID or the open handle of some other device associated with desired card. In that case, mixerOpen() will ensure that it returns the Mixer device associated with that card's other device. So for example, here's how you would open the default WAVE OUT device (ie, the WAVE OUT on the default card), and then get the handle to that card's Mixer device:

```
unsigned long err;
HMIXER        mixerHandle;
WAVEFORMATEX  waveFormat;
HWAVEOUT      hWaveOut;

/* Open the default WAVE Out Device, specifying my callback.
   Assume that waveFormat has already been initialized as desired */
err = waveOutOpen(&hWaveOut, WAVE_MAPPER, &waveFormat, (DWORD)WaveOutProc, 0,
CALLBACK_FUNCTION);
if (err)
```

```
{
    printf("ERROR: Can't open WAVE Out Device! -- %08X\n", err);
}
else
{
    /* Open the mixer associated with the WAVE OUT device
       opened above. Note that I pass the handle obtained
       via waveOutOpen() above */
    err = mixerOpen(&mixerHandle, hWaveOut, 0, 0, MIXER_OBJECTF_HWAVEOUT);
    if (err)
    {
        printf("ERROR: Can't open Mixer Device! -- %08X\n", err);
    }
}
```

The key above is in passing not only the handle that you obtain with waveOutOpen() (or waveInOpen(), or midiOutOpen(), or midiInOpen()) but also the last parameter of mixerOpen() must be MIXER_OBJECTF_HWAVEOUT (or MIXER_OBJECTF_HWAVEIN, or MIXER_OBJECTF_HMIDIOUT, or MIXER_OBJECTF_HMIDIIN) to indicate what kind of device handle is being passed to mixerOpen()

Alternately, if you know the device ID number of the desired WAVE OUT device (but haven't yet opened its handle via waveOutOpen()), you can pass that device ID to mixerOpen() and instead specify MIXER_OBJECTF_WAVEOUT. mixerOpen() will find the Mixer that corresponds to that WAVE Out device with the specified ID.

If desired, you can then get the above Mixer's ID (number) from its handle, using mixerGetID() as so:

```
unsigned long mixerID;

err = mixerGetID(mixerHandle, &mixerID, MIXER_OBJECTF_HMIXER);
if (err)
{
    printf("ERROR: Can't get Mixer Device ID! -- %08X\n", err);
}
else
{
    printf("Mixer Device ID = %d\n", mixerID);
}
```

Listing all of the Mixer devices

If you're writing an application where you need to list all of the Mixer devices in the system, Windows has a function that you can call to determine how many Mixer devices are in the list. This function is called mixerGetNumDevs(). This returns the number of Mixer devices in the list. Remember that the Device IDs start with 0 and increment. So if Windows says that there are 3 devices in the list, then you know that their Device IDs are 0, 1, and 2 respectively. You then use these Device IDs with other Windows functions. For example, there is a function you can call to get information about one of the devices in the list, such as its name, and what sort of other features it has such as how many components it contains, and what type of component each one is. You pass the Device ID of the Mixer device which you want to get information about (as well as a pointer to a special structure called a MIXERCAPS into which Windows puts the info about the device), The name of the function to get information about a particular Mixer device is mixerGetDevCaps().

Here then is an example of going through the list of Mixer devices, and printing the name of each one:

```
MIXERCAPS     mixcaps;
unsigned long iNumDevs, i;

/* Get the number of Mixer devices in this computer */
```

```
iNumDevs = mixerGetNumDevs();

/* Go through all of those devices, displaying their IDs/names */
for (i = 0; i < iNumDevs; i++)
{
    /* Get info about the next device */
    if (!mixerGetDevCaps(i, &mixcaps, sizeof(MIXERCAPS)))
    {
        /* Display its ID number and name */
        printf("Device ID #%u: %s\r\n", i, mixcaps.szPname);
    }
}
```

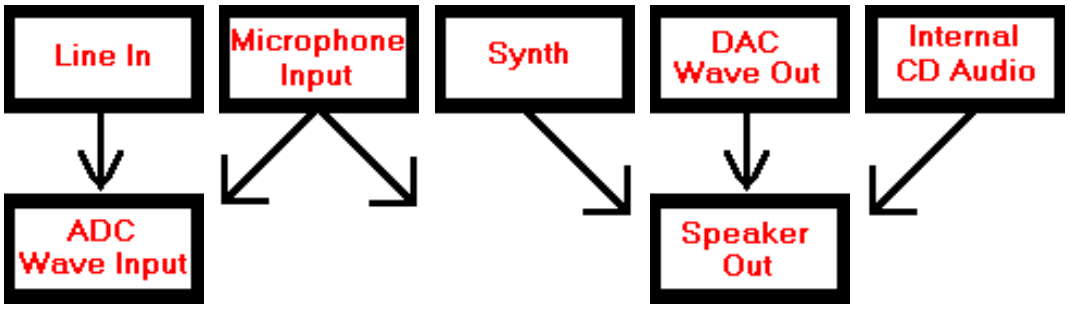
About Lines and Controls

Up to this point, I have been using the term "component" to refer to a hardware section that has its own individual, adjustable parameters. The Mixer API actually concerns itself with signal flows. (ie, In our block diagram, the signal flows are the 5 arrows which connect the components). The Microsoft documention refers to each signal flow (ie, each arrow in our block diagram) as a "source line". So, a Mixer device controls "source lines" -- not components per se. It is each "source line" which has its own individual, adjustable parameters -- not a component per se. Our example audio card has 5 source lines as far as a Mixer is concerned. From here on, whenever you see the word "source line", think of the signal flow between components

I have also been using the term "parameters" to refer to settings that are adjustable on a "line", for example a volume level or a mute switch or a bass boast or a pan setting, etc. The Microsoft documentation refers to these parameters as "controls" (which can be a confusing term since a programmer typically thinks of a control as a graphical window -- part of his graphical interface. But in the case of Mixers, we mean "audio controls").

So for example, it's not really the "Microphone Input" component itself which concerns the Mixer API. Rather, it's the signal flow between the "Microphone Input" and "ADC" components. The "Microphone Input" volume control adjusts the volume of this signal flowing between the Microphone Input and ADC components.

For further illustration of the difference between components and source lines, let's add another feature to our example sound card. Sometimes, you'd like the Microphone Input to not only pipe through the ADC (so that you can record it), but also to pipe through the speakers (so that you can monitor the signal that you're actually recording, and hear what it sounds like through the speakers). So, let's adjust our block diagram to show the signal from the "Microphone Input" as going to both the ADC and the speakers.



.

Notice that we now have 6 "source lines" (arrows). And there are two source lines from the "Microphone Input", even though it is only one component on our card. Each source line has its own settings. For example, there is a volume control for the "Microphone Input" source line going into the ADC (so that you can set the recording level for your WAVE recording software). There is another, separate volume control for the "Microphone Input" source line going to the speakers (so that you can set the monitor level of the microphone, separate from the recording level). These source lines may each have their own mute switches (so that you can, for example, have the "Microphone Input" signal going to only the ADC, but not the Speakers). They may have other controls as well, and each line's controls will be separate from the other line's controls.

There is another important concept to discuss in regard to lines. There are such things as "destination lines", and we differentiate between source lines and destination lines. What are those? Well, a destination line has other lines -- source lines -- flowing into it. In our block diagram, the "Speaker Out" is a destination line, since it has signals from the "Internal CD Audio", "Synth", "DAC Wave Out" and "Microphone Input" flowing into it. And the latter 4 are source lines. They are source lines to the "Speaker Out" destination line since their signals flow into the "Speaker Out" line. Source lines always have to flow into some destination line. (ie, A source line cannot exist without being attached to one destination line).

So, whereas source lines are the arrows in our block diagram, destination lines are actual components in our block diagram.

So too, the "ADC Wave Input" is a destination line in our example card. It has 2 source lines flowing into it -- the "Microphone Input" and "Line In". Those two lines are not source lines to the "Speaker Out" since they do not flow into it. Similiarly, the 4 source lines flowing into the "Speaker Out" are not source lines to the "ADC Wave Input" destination line, since they don't flow into the latter.

Like source lines, destination lines can have controls. And each destination line's controls are separate from the other lines' controls. For example, the "Speaker Out" may have a volume control. This would function as a master volume for all 4 source lines going to the Speaker Out destination line. (And each of those 4 source lines has its own individual volume level control).

NOTE: Although a card may have a stereo component (for example, typically the "Speaker Out" upon most cards is stereo), this is considered one line. It has 2 channels, but it nevertheless is one line on the Mixer device. Indeed, a component could have more than 2 channels, and still be considered only one line. In conclusion, lines are not the same thing as channels. In some ways, it may be helpful to think of a line like a MIDI cord. One MIDI cord connects two components, but there can be several channels going over that one cord. The same thing is true of a line.

In conclusion, our example card has 6 source lines and 2 destination lines. The 4 source lines labeled "Internal CD Audio", "Synth", "DAC Wave Out" and "Microphone Input" are attached to the "Speaker Out" destination line. The 2 source lines labeled "Microphone Input" and "Line In" are attached to the "ADC Wave Input" destination line.

Line IDs and Types

Every line in a mixer must have a unique ID number. Every line also has a type. This is just a numeric value that describes what type of line it is. These are defined in the MMSYSTEM.H include file. For example, a line may have a type of MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER to indicate that it is a signal from a built-in sound module.

The allowable types for source lines are as so:

MIXERLINE_COMPONENTTYPE_SRC_DIGITAL	A digital source, for example, a SPDIF input jack.
MIXERLINE_COMPONENTTYPE_SRC_LINE	A line input source. Typically used for a line input jack, if there is a separate microphone input (ie, MIXERLINE_COMPONENTTYPE_SRC_MICROPHONE).
MIXERLINE_COMPONENTTYPE_SRC_MICROPHONE	Microphone input (but also used for a combination of Mic/Line input if there isn't a separate line input source).
MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER	Musical synth. Typically used for a card that contains a synth capable of playing MIDI. This would be the audio out of that built-in synth.
MIXERLINE_COMPONENTTYPE_SRC_COMPACTDISC	The audio feed from an internal CDROM drive (connected to the sound card).
MIXERLINE_COMPONENTTYPE_SRC_TELEPHONE	Typically used for a telephone line's incoming audio to be piped through the computer's speakers, or the telephone line in jack for a built-in modem.

MIXERLINE_COMPONENTTYPE_SRC_PCSPEAKER	Typically, to allow sound, that normally goes to the computer's built-in speaker, to instead be routed through the card's speaker output. The motherboard's system speaker connector would be internally connected to some connector on the sound card for this purpose.
MIXERLINE_COMPONENTTYPE_SRC_WAVEOUT	Wave playback (ie, this is the card's DAC).
MIXERLINE_COMPONENTTYPE_SRC_AUXILIARY	An aux jack meant to be routed to the Speaker Out, or to the ADC (for WAVE recording). Typically, this is used to connect external, analog equipment (such as tape decks, the audio outputs of musical instruments, etc) for digitalizing or playback through the sound card.
MIXERLINE_COMPONENTTYPE_SRC_ANALOG	May be used similiarly to MIXERLINE_COMPONENTTYPE_SRC_AUXILIARY (although I have seen some mixers use this like MIXERLINE_COMPONENTTYPE_SRC_PCSPEAKER). In general, this would be some analog connector on the sound card which is only accessible internally, to be used to internally connect some analog component inside of the computer case so that it plays through the speaker out.
MIXERLINE_COMPONENTTYPE_SRC_UNDEFINED	Undefined type of source. If none of the others above are applicable.

The allowable types for destination lines are as so:

MIXERLINE_COMPONENTTYPE_DST_DIGITAL	A digital destination, for example, a SPDIF output jack.
MIXERLINE_COMPONENTTYPE_DST_LINE	A line output destination. Typically used for a line output jack, if there is a separate speaker output (ie, MIXERLINE_COMPONENTTYPE_DST_SPEAKERS).
MIXERLINE_COMPONENTTYPE_DST_MONITOR	Typically a "Monitor Out" jack to be used for a speaker system separate from the main speaker out. Or, it could be some built-in monitor speaker on the sound card itself, such as a speaker for a built-in modem.
MIXERLINE_COMPONENTTYPE_DST_SPEAKERS	The audio output to a pair of speakers (ie, the "Speaker Out" jack).
MIXERLINE_COMPONENTTYPE_DST_HEADPHONES	Typically, a headphone output jack.
MIXERLINE_COMPONENTTYPE_DST_TELEPHONE	Typically used to daisy-chain a telephone to an analog modem's "telephone out" jack.
MIXERLINE_COMPONENTTYPE_DST_WAVEIN	The card's ADC (to digitize analog sources, for example, in recording WAVE files of such).
MIXERLINE_COMPONENTTYPE_DST_VOICEIN	May be some sort of hardware used for voice recognition. Typically, a microphone source line would be attached to this.
MIXERLINE_COMPONENTTYPE_DST_UNDEFINED	Undefined type of destination. If none of the others above are applicable.

Control IDs and Types

Each line can have one or more adjustable "audio controls". (But, it's possible that a line could have no controls at all). For example, the Synth line may have a volume fader and a mute switch. Each control has a type. These are defined in the MMSYSTEM.H include file. For example, the volume fader would have a type of MIXERCONTROL_CONTROLTYPE_VOLUME. The Mute switch would have a type of MIXERCONTROL_CONTROLTYPE_MUTE.

Every control also has a unique ID number. No two controls may have the same ID number, even if they belong to 2 different lines.

The control types are divided up into classes. These classes are roughly based upon what type of value a control adjusts, and therefore what kind of graphical user interface you would normally present to the enduser to let him adjust that control's value. For example, you would normally present a graphical fader to allow the user to adjust a control of the type MIXERCONTROL_CONTROLTYPE_VOLUME. On the other hand, you'd typically use a graphical (checkmark) button to allow him to adjust a control of type MIXERCONTROL_CONTROLTYPE_MUTE (since that control has only two possible values, or states).

The allowable classes for controls are as so:

MIXERCONTROL_CT_CLASS_FADER	A control that is adjusted by a vertical fader, with a linear scale of positive values (ie, 0 is the lowest possible value). A MIXERCONTROLDETAILS_UNSIGNED structure is used to retrieve or set the control's value.
MIXERCONTROL_CT_CLASS_LIST	A control that is adjusted by a listbox containing numerous "values" to be selected. The user will single-select, or perhaps multiple-select if desired, his choice of value(s). A MIXERCONTROLDETAILS_BOOLEAN structure is used to retrieve or set the control's value. A MIXERCONTROLDETAILS_LISTTEXT structure is also used to retrieve the text description of each item of this control.
MIXERCONTROL_CT_CLASS_METER	A control that is adjusted by a graphical meter. A MIXERCONTROLDETAILS_BOOLEAN, MIXERCONTROLDETAILS_SIGNED, or MIXERCONTROLDETAILS_UNSIGNED structure is used to retrieve or set the control's value.
MIXERCONTROL_CT_CLASS_NUMBER	A control that is adjusted by numeric entry. The user enters a signed integer, unsigned integer, or integer decibel value. A MIXERCONTROLDETAILS_SIGNED or MIXERCONTROLDETAILS_UNSIGNED structure is used to retrieve or set the control's value.
MIXERCONTROL_CT_CLASS_SLIDER	A control that is adjusted by a horizontal slider with a linear scale of negative and positive values. (ie, Generally, 0 is the mid or "neutral" point). A MIXERCONTROLDETAILS_SIGNED structure is used to retrieve or set the control's value.
MIXERCONTROL_CT_CLASS_SWITCH	A control that is has only two states (ie, values), and is therefore adjusted via a button. A MIXERCONTROLDETAILS_BOOLEAN structure is used to retrieve or set the control's value.
MIXERCONTROL_CT_CLASS_TIME	A control that allows the user to enter a time value, such as Reverb Decay Time. It is a positive, integer value.
MIXERCONTROL_CT_CLASS_CUSTOM	A custom class of control. If none of the others above are applicable.

Each class has certain types associated with it. For example, the MIXERCONTROL_CT_CLASS_FADER class has the following 5 types associated with it:

MIXERCONTROL_CONTROLTYPE_VOLUME	Volume fader. The range of allowable values is 0 through 65,535.

MIXERCONTROL_CONTROLTYPE_BASS	Bass boost fader. The range of allowable values is 0 through 65,535.
MIXERCONTROL_CONTROLTYPE_TREBLE	Treble boost fader. The range of allowable values is 0 through 65,535.
MIXERCONTROL_CONTROLTYPE_EQUALIZER	A graphic EQ. The range of allowable values for each band is 0 through 65,535. A MIXERCONTROLDETAILS_LISTTEXT structure is used to retrieve the text label for each band of the EQ. Typically, this control will also have its MIXERCONTROL_CONTROLF_MULTIPLE flag set, since the EQ will likely have numerous bands.
MIXERCONTROL_CONTROLTYPE_FADER	A generic fader, to be used when none of the above are applicable. The range of allowable values is 0 through 65,535.

In fact, if you look at MMSYSTEM.H, you'll note that a control with a type of MIXERCONTROL_CONTROLTYPE_VOLUME is defined as MIXERCONTROL_CT_CLASS_FADER | MIXERCONTROL_CT_UNITS_UNSIGNED + 1. The class is actually contained in the top 4 bits of the type. So if you know a control's type, then you determine its class by masking off the top 4 bits. For example, assume that you've queried a control's type, and stored the value returned by the Mixer API in your variable named "type". Here's how you'd figure out it's class:

```
unsigned long   type;

/* Figure out the class based upon the top 4 bits of its type */
switch (MIXERCONTROL_CT_CLASS_MASK & type)
{
    case MIXERCONTROL_CT_CLASS_FADER:
    {
        printf("It's a fader class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_LIST:
    {
        printf("It's a list class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_METER:
    {
        printf("It's a meter class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_NUMBER:
    {
        printf("It's a number class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_SLIDER:
    {
        printf("It's a slider class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_TIME:
    {
        printf("It's a time class.");
        break;
    }
    case MIXERCONTROL_CT_CLASS_CUSTOM:
```

```
    {
        printf("It's a custom class.");
        break;
    }
}
```

The MIXERCONTROL_CT_CLASS_SWITCH class has the following 7 types associated with it:

MIXERCONTROL_CONTROLTYPE_BOOLEAN	A control with a boolean value. The value is an integer that is either 0 (FALSE) or non-zero (TRUE).
MIXERCONTROL_CONTROLTYPE_BUTTON	A control whose value is 1 when the button is pressed (ie, some feature/action is enabled), or 0 if not pressed (ie, no action is taken). For example, this type of control may be used by a talkback button or pedal sustain -- the action/feature is on only while the button is pressed, and otherwise is not applicable.
MIXERCONTROL_CONTROLTYPE_LOUDNESS	A control whose value is 1 to boost bass frequencies, or 0 for normal (ie, no boost). (A MIXERCONTROL_CONTROLTYPE_BASS fader control may be used to set the actual amount of boost, in conjunction with this control turning the boost on/off)
MIXERCONTROL_CONTROLTYPE_MONO	A control whose value is 1 for mono operation (ie, all channels are summed into one), or 0 for normal (ie, stereo or multi-channel).
MIXERCONTROL_CONTROLTYPE_MUTE	A control whose value is 1 to mute some feature, or 0 for normal (ie, no mute).
MIXERCONTROL_CONTROLTYPE_ONOFF	A control whose value is 1 to enable some feature/action, or 0 to disable that feature/action. The difference between this and MIXERCONTROL_CONTROLTYPE_BUTTON is that the latter's 0 value doesn't disable the feature/action per se, but rather, simply represents a "not applicable" state for the feature/action. MIXERCONTROL_CONTROLTYPE_ONOFF would be similiar to a real on/off switch (ie, a checkmark button in Windows parlance) whereas MIXERCONTROL_CONTROLTYPE_BUTTON would be more akin to a "momentary switch" (ie, a pushbutton). The difference between MIXERCONTROL_CONTROLTYPE_ONOFF and MIXERCONTROL_CONTROLTYPE_BOOLEAN is merely the labeling/grahics that would be shown on the user interface. The former is "ON" when its value is 1. The latter is "TRUE" when its value is 1. So typically, the two buttons would be represented by different labels and/or graphics to reflect that difference in semantics.
MIXERCONTROL_CONTROLTYPE_STEREOENH	A control whose value is 1 to enable a stereo enhance feature (ie, increase stereo separation), or 0 for normal (ie, no enhance).

The MIXERCONTROL_CT_CLASS_LIST class has the following 4 types associated with it:

MIXERCONTROL_CONTROLTYPE_SINGLESELECT	Allows one selection out of a choice of many selections. For example, this can be used to select a reverb type out of choice of many types (ie, Hall, Plate, Room, etc).

MIXERCONTROL_CONTROLTYPE_MULTIPLESELECT	Like MIXERCONTROL_CONTROLTYPE_SINGLESELECT, but allows the selection of more than one item simultaneously.
MIXERCONTROL_CONTROLTYPE_MUX	Allows the selection of one audio line out of several choices of audio lines. For example, to allow a source line to be routed to one of several possible destination lines -- this control could list all of the possible choices of destination lines, and allow one to be chosen.
MIXERCONTROL_CONTROLTYPE_MIXER	Like MIXERCONTROL_CONTROLTYPE_MUX, but allows the selection of more than one audio line simultaneously. For example, this control could be for a reverb component which allows several source lines to be simultaneously routed to it, and this control determines which source lines are selected for routing to the reverb.

The MIXERCONTROL_CT_CLASS_METER class has the following 4 types associated with it:

MIXERCONTROL_CONTROLTYPE_BOOLEANMETER	A meter whose integer value is either 0 (ie, FALSE) or non-zero (TRUE). A MIXERCONTROLDETAILS_BOOLEAN struct is used to set/retrieve its value.
MIXERCONTROL_CONTROLTYPE_PEAKMETER	A control with a value that is an integer whose allowable, maximum range is -32,768 (lowest) through 32,767 (highest). In other words, its value is a SHORT. A MIXERCONTROLDETAILS_SIGNED struct is used to set/retrieve its value.
MIXERCONTROL_CONTROLTYPE_SIGNEDMETER	A control with a value that is an integer whose allowable, maximum range is -2,147,483,648 (lowest) through 2,147,483,647 (highest) inclusive. In other words, its value is a LONG. A MIXERCONTROLDETAILS_SIGNED struct is used to set/retrieve its value.
MIXERCONTROL_CONTROLTYPE_UNSIGNEDMETER	Like MIXERCONTROL_CONTROLTYPE_SIGNEDMETER, but its allowable, maximum value range is from 0 (lowest) to 4,294,967,295. In other words, its value is a ULONG. A MIXERCONTROLDETAILS_UNSIGNED struct is used to set/retrieve its value.

Some of the allowable types for the MIXERCONTROL_CT_CLASS_NUMBER class are similiar to the MIXERCONTROL_CT_CLASS_METER class types. But with the MIXERCONTROL_CT_CLASS_NUMBER class, you would typically use an Edit control for the graphical user interface to allow him to enter a value. The MIXERCONTROL_CT_CLASS_METER would typically use some sort of graphical display similiar to an audio meter. MIXERCONTROL_CT_CLASS_NUMBER class has the following 4 types associated with it:

MIXERCONTROL_CONTROLTYPE_SIGNED	A control with a value that is an integer whose allowable, maximum range is -2,147,483,648 (lowest) through 2,147,483,647 (highest) inclusive. In other words, its value is a LONG. A MIXERCONTROLDETAILS_SIGNED struct is used to set/retrieve its value.
MIXERCONTROL_CONTROLTYPE_UNSIGNED	Like MIXERCONTROL_CONTROLTYPE_SIGNEDMETER, but its allowable, maximum value range is from 0 (lowest) to 4,294,967,295. In other words, its value is a ULONG. A MIXERCONTROLDETAILS_UNSIGNED struct is used to set/retrieve its value.

MIXERCONTROL_CONTROLTYPE_PERCENT	A control whose integer value is a percent. A MIXERCONTROLDETAILS_UNSIGNED struct is used to set/retrieve its value.
MIXERCONTROL_CONTROLTYPE_DECIBELS	A control with a value that is an integer whose allowable, maximum range is -32,768 (lowest) through 32,767 (highest). In other words, its value is a SHORT. Each increment is a tenth of a decibel. A MIXERCONTROLDETAILS_SIGNED struct is used to set/retrieve its value.

The MIXERCONTROL_CT_CLASS_SLIDER class has the following 3 types associated with it:

MIXERCONTROL_CONTROLTYPE_SLIDER	A slider with a value that is an integer whose allowable, maximum range is -32,768 (lowest) through 32,767 (highest). In other words, its value is a SHORT.
MIXERCONTROL_CONTROLTYPE_PAN	A slider with a value that is an integer whose allowable, maximum range is -32,768 (far left) through 32,767 (far right). In other words, its value is a SHORT. It represents pan position in the stereo spectrum, where 0 is center position.
MIXERCONTROL_CONTROLTYPE_QSOUNDPAN	A slider with a value that is an integer whose allowable, maximum range is -15 (lowest) through 15 (highest). In other words, its value is a SHORT. It represents Qsound's expanded sound setting.

The MIXERCONTROL_CT_CLASS_TIME class has the following 2 types associated with it:

MIXERCONTROL_CONTROLTYPE_MICROTIME	A control with a value that is an integer whose allowable, maximum range is 0 (lowest) through 4,294,967,295. In other words, its value is a ULONG. Its value represents an amount of time in microseconds.
MIXERCONTROL_CONTROLTYPE_MILLITIME	A control with a value that is an integer whose allowable, maximum range is 0 (lowest) through 4,294,967,295. In other words, its value is a ULONG. Its value represents an amount of time in milliseconds.

The MIXERCONTROL_CT_CLASS_CUSTOM class is a proprietary class. A Mixer that uses such a type of control could expect only an application that is written specifically for that Mixer to understand what types of controls are in this class, and what structures to use to set/retrieve their values. (Perhaps even proprietary structures could be used)

MIXERLINE structures, and enumerating lines

One way to get information about a mixer's lines, if you don't know what particular lines it has (ie, you don't know the types of lines it has, nor know their ID numbers), is to first call mixerGetDevCaps() to fetch information about that Mixer device into a MIXERCAPS structure. Using information in this structure, you can determine how many destination lines are on the card. From there, you can enumerate (ie, fetch information about) each destination line, and the source lines to each destination line. And after you enumerate the lines, you can enumerate the controls for each line.

Let's examine such an approach and study the structures that are associated with the Mixer API.

In order to better understand the Mixer API, we should take a peek inside of the Mixer Device for our example audio card, and examine its internal structures which are used with the Mixer API. We'll assume that this Mixer Device is written in C, and therefore uses C structures.

As noted previously, the Mixer API mixerGetDevCaps() is used to fetch information about a Mixer Device. It fills in a

MIXERCAPS structure. In particular, the ***cDestinations*** field of the MIXERCAPS tells you how many destination lines are on the card. It doesn't tell you how many total lines (ie, destination and source lines) there are. It counts only destination lines. As you'll recall, our example audio card has 2 destination lines. We'll fill in the other fields, such as the Mixer name and product ID with arbitrary values for the purposes of this tutorial. Assume that this mixer is the first installed mixer in the system (ie, ID = 0). Here then is our Mixer Device's MIXERCAPS structure (which is defined in MMSYSTEM.H):

```
MIXERCAPS mixercaps = {
    0,       /* manufacturer id */
    0,       /* product id */
    0x0100,  /* driver version #. Note that the high 8-bits
             are the version, and low 8-bits are revision. So,
             our driver version is 1.0 */
    "Example Sound Card", /* product name */
    0,       /* Support bits. None are currently defined */
    2,       /* # of destination lines */
};
```

Here is an example of passing a MIXERCAPS to mixerGetDevCaps() and letting Windows fill it in with the above values. (Assume that we have already opened the mixer and stored its handle into our variable "mixerHandle").

```
MIXERCAPS     mixcaps;
MMRESULT      err;

/* Get info about the first Mixer Device */
if (!(err = mixerGetDevCaps((UINT)mixerHandle, &mixcaps, sizeof(MIXERCAPS))))
{
    /* Success. We can continue on with our tutorial below */
}
else
{
    /* An error */
    printf("Error #%d calling mixerGetDevCaps()\n", err);
}
```

Information about a line is contained in a MIXERLINE structure (as defined in MMSYSTEM.H). Let's assume that our "Speaker Out" destination line has two controls -- a volume slider (to control the overall mix to the speakers) and a mute switch (to mute the overall mix). Here's the MIXERLINE structure for our "Speaker Out" destination:

```
MIXERLINE mixerline_SpkrOut = {
    sizeof(MIXERLINE),              /* size of MIXERLINE structure */
    0,                              /* zero based index of destination line */
    0,                              /* zero based source index (used only if
                                       this is a source line) */
    0xFFFF0000,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE,         /* state/information about line */
    0,                              /* driver specific information */
    MIXERLINE_COMPONENTTYPE_DST_SPEAKERS,  /* component type */
    2,                              /* # of channels this line supports */
    4,                              /* # of source lines connected (used
                                       only if this is a destination line) */
    2,                              /* # of controls in this line */
    "Spkr Out",                     /* Short name for this line */
    "Speaker Out",                  /* Long name for this line */
    MIXERLINE_TARGETTYPE_WAVEOUT,   /* MIXERLINE_TARGETTYPE_xxxx */
    /* The following info is just some info about the Mixer Device for this
    line, prepended to the above info. It's mostly just for reference. */
    0,       /* device ID of our Mixer */
```

```
    0,       /* manufacturer id */
    0,       /* product id */
    0x0100,  /* driver version # */
    "Example Sound Card", /* product name */
};
```

There are a few things to note here. First, notice that the ***dwComponentType*** for the "Speaker Out" destination line is one of the types for a destination line -- appropriately, MIXERLINE_COMPONENTTYPE_DST_SPEAKERS to indicate that it's a speaker output. I've chosen a value of 0xFFFF0000 for the ***dwLineID***. The Mixer Device programmer may chose any value he wishes for this field, but no other line in this mixer may have the same value for its ***dwLineID*** (as you'll notice later). Also, note that since our speaker output is stereo, the ***cChannels*** field is 2. As you'll recall, there are 4 source lines connected to our "Speaker Out" destination line, so the ***cConnections*** field is 4. The name fields are nul-terminated strings. They can be anything that the Mixer Device programmer chooses, but the shorter name is meant to be used as a label for any tightly spaced graphical controls. When the MIXERLINE_LINEF_ACTIVE bit is set in the ***fdwLine*** field, this simply means that the line has not been disabled (such as what may happen if it were muted).

The ***dwDestination*** field is an index value based from 0. The first destination line in a mixer will have an index value of 0 (as in our example above). The second destination line will have an index of 1. The third destination line will have an index of 2. Etc. This is the same concept as how Windows enumerates Mixer Devices (ie, where the first installed Mixer Device has an ID of 0). The index value is not necessarily the same as a line's ID number (as you can see from the example above). What is the purpose of an index, and why do we need both an index and an ID number? As you may guess, the index number is primarily used when you need to enumerate what lines a Mixer has. Until you enumerate the lines (ie, fetch information about each line), you don't know the ID numbers of any of the lines. So, you need to use indexes with the mixer API to enumerate lines. But once you have retrieved info about a line, and therefore know its ID number, then you can alter its settings more directly. So, index numbers are primarily useful for initially enumerating lines and controls to find their ID numbers and types. And then the ID numbers are primarily useful for subsequently performing direct manipulation of the lines and controls.

Now let's take a look at the MIXERLINE structure for our "ADC Wave Input" destination line. Let's assume that it also has two controls -- a volume slider (to control the overall mix to the ADC) and a mute switch (to mute the overall mix). Here's the MIXERLINE structure for our "ADC Wave Input" destination:

```
MIXERLINE mixerline_WaveIn = {
    sizeof(MIXERLINE),              /* size of MIXERLINE structure */
    1,                              /* zero based index of destination line */
    0,                              /* zero based source index (used only if
                                       this is a source line) */
    0xFFFF0001,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE,         /* state/information about line */
    0,                              /* driver specific information */
    MIXERLINE_COMPONENTTYPE_DST_WAVEIN,  /* component type */
    2,                              /* # of channels this line supports */
    2,                              /* # of source lines connected (used
                                       only if this is a destination line) */
    2,                              /* # of controls in this line */
    "Wave In",                      /* Short name for this line */
    "Wave Input",                   /* Long name for this line */
    MIXERLINE_TARGETTYPE_WAVEIN,    /* MIXERLINE_TARGETTYPE_xxxx */
    /* The following info is just some info about the Mixer Device for this
    line, prepended to the above info. It's mostly just for reference and is only
valid if the
        Target Type field is not MIXERLINE_TARGETTYPE_UNDEFINED. */
    0, 0, 0, 0x0100, "Example Sound Card",
};
```

Notice that the ***dwComponentType*** for the "ADC Wave Input" destination line is MIXERLINE_COMPONENTTYPE_DST_WAVEIN to indicate that it's a wave input. Also, I've chosen a value of 0xFFFF0001 for the ***dwLineID*** -- a different value than the "Speaker Out" destination line. Also, note that since our sound card

is capable of digitizing in stereo, the ***cChannels*** field is 2. As you'll recall, there are 2 source lines connected to our "ADC Wave Input" destination line, so the ***cConnections*** field is 2. Finally, note that the ***dwDestination*** field is 1, since this is the second destination line in the Mixer.

The mixer API mixerGetLineInfo() fills in a MIXERLINE struct for a specified line. This is how you retrieve info about a line. If you don't know a line's ID number (as would be the case when you're first enumerating the lines), then you can reference it by index. You pass mixerGetLineInfo() the value MIXER_GETLINEINFOF_DESTINATION to notify it that you want to reference the line by its index value. For example, here's how you would retrieve info about the first destination line in our example Mixer. Before calling mixerGetLineInfo() and passing it a MIXERLINE to fill in, you must initialize 2 fields. The ***cbStruct*** field must be set to the size of the MIXERLINE struct you're passing, and the ***dwDestination*** field must be set to the index value of the line whose info you wish to retrieve. Remember that the first destination line has an index of 0, so to retrieve its info, we set dwDestination to 0.

```
MIXERLINE     mixerline;
MMRESULT      err;

/* Get info about the first destination line by its index */
mixerline.cbStruct = sizeof(MIXERLINE);
mixerline.dwDestination = 0;

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_DESTINATION)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

When the above call returns, mixerGetLineInfo() will have filled in our MIXERLINE struct as per the "Speaker Out" (mixerline_SpkrOut) MIXERLINE struct shown above. (After all, the "Speaker Out" is the first line in our example Mixer -- it has an index of 0).

Now, if you want to fetch info about the second destination line in the mixer, the only thing different is the index value you stuff into the ***dwDestination*** field, as so:

```
/* Get info about the second destination line by its index */
mixerLine.cbStruct = sizeof(MIXERLINE);
mixerLine.dwDestination = 1;

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerLine,
MIXER_GETLINEINFOF_DESTINATION)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

When the above call returns, mixerGetLineInfo() will have filled in our MIXERLINE struct as per the "ADC Wave Input" (mixerline_WaveIn) MIXERLINE struct shown above. (After all, the "ADC Wave Input" is the second line in our example Mixer -- it has an index of 1).

Now, you should see how you can enumerate the destination lines by their indexes. Here is an example of printing out the names of all destination lines in our Mixer:

```
MIXERCAPS     mixcaps;
MIXERLINE     mixerline;
MMRESULT      err;
unsigned long i;
```

```
/* Get info about the first Mixer Device */
if (!(err = mixerGetDevCaps((UINT)mixerHandle, &mixcaps, sizeof(MIXERCAPS))))
{
    /* Print out the name of each destination line */
    for (i = 0; i < mixercaps.cDestinations; i++)
    {
        mixerline.cbStruct = sizeof(MIXERLINE);
        mixerline.dwDestination = i;

        if (!(err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_DESTINATION)))
        {
            printf("Destination #%lu = %s\n", i, mixerline.szName);
        }
    }
}
```

Now, we need to enumerate the source lines for each destination line. We also use an index value to reference each source line. The first source line, for a given destination line, has an index value of 0. The second source line, for that destination line, has an index of 1. The third source line has an index of 2. Etc. Remember that the "Speaker Out" had 4 source lines going into it: "Internal CD Audio", "Synth", "DAC Wave Out" and "Microphone Input". So their respective index values are 0, 1, 2, and 3. Let's look at the MIXERLINE structs for those 4 source lines. Assume that each one of them has 2 controls, a volume slider and a mute switch. Also, assume that each one is a stereo source.

```
MIXERLINE mixerline_CD = {
    sizeof(MIXERLINE),
    0,                              /* zero based index of destination line */
    0,                              /* zero based source index */
    0x00000000,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE|MIXERLINE_LINEF_SOURCE, /* state/information about line */
    0,
    MIXERLINE_COMPONENTTYPE_SRC_COMPACTDISC,  /* component type */
    2,                              /* # of channels this line supports */
    0,                              /* Not applicable for source lines */
    2,                              /* # of controls in this line */
    "CD Audio",                     /* Short name for this line */
    "Internal CD Audio",            /* Long name for this line */
    MIXERLINE_TARGETTYPE_UNDEFINED, /* MIXERLINE_TARGETTYPE_xxxx */
    0, 0, 0, 0x0100, "Example Sound Card",
};

MIXERLINE mixerline_Synth = {
    sizeof(MIXERLINE),
    0,                              /* zero based index of destination line */
    1,                              /* zero based source index */
    0x00000001,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE|MIXERLINE_LINEF_SOURCE,
    0,
    MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER,  /* component type */
    2,                              /* # of channels this line supports */
    0,                              /* Not applicable for source lines */
    2,                              /* # of controls in this line */
    "Synth",                        /* Short name for this line */
    "Synth",                        /* Long name for this line */
    MIXERLINE_TARGETTYPE_UNDEFINED, /* MIXERLINE_TARGETTYPE_xxxx */
    0, 0, 0, 0x0100, "Example Sound Card",
};
```

```
MIXERLINE mixerline_WaveOut = {
    sizeof(MIXERLINE),
    0,                              /* zero based index of destination line */
    2,                              /* zero based source index */
    0x00000002,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE|MIXERLINE_LINEF_SOURCE,
    0,
    MIXERLINE_COMPONENTTYPE_SRC_WAVEOUT,  /* component type */
    2,                              /* # of channels this line supports */
    0,                              /* Not applicable for source lines */
    2,                              /* # of controls in this line */
    "Wave Out",                     /* Short name for this line */
    "DAC Wave Out",                 /* Long name for this line */
    MIXERLINE_TARGETTYPE_WAVEOUT,   /* MIXERLINE_TARGETTYPE_xxxx */
    0, 0, 0, 0x0100, "Example Sound Card",
};

MIXERLINE mixerline_Mic = {
    sizeof(MIXERLINE),
    0,                              /* zero based index of destination line */
    3,                              /* zero based source index */
    0x00000003,                     /* unique ID # for this line */
    MIXERLINE_LINEF_ACTIVE|MIXERLINE_LINEF_SOURCE,
    0,
    MIXERLINE_COMPONENTTYPE_SRC_MICROPHONE,  /* component type */
    2,                              /* # of channels this line supports */
    0,                              /* Not applicable for source lines */
    2,                              /* # of controls in this line */
    "Mic",                          /* Short name for this line */
    "Microphone Input",             /* Long name for this line */
    MIXERLINE_TARGETTYPE_WAVEIN,    /* MIXERLINE_TARGETTYPE_xxxx */
    0, 0, 0, 0x0100, "Example Sound Card",
};
```

One thing that you'll note is, unlike with the destination lines, the MIXERLINEs for all source lines have their MIXERLINE_LINEF_SOURCE flag bit set. When this bit is set, you know that you have info on a source line. Secondly, note that the zero-based index for the destination line is 0. That's because all of the above source lines connect to the "Speaker Out" destination line, which is the first line in our mixer (and therefore has an index value of 0). Next, note that the zero-based source indexes for the 4 source lines are 0, 1, 2, and 3, respectively. Finally, note that the ID of each source line is unique -- unlike any other line, including any of the destination lines.

The mixer API mixerGetLineInfo() fills in a MIXERLINE struct for a source line too. Again, you can reference a source line by its index, but you also need to know the index of its respective destination line. You pass mixerGetLineInfo() the value MIXER_GETLINEINFOF_SOURCE to notify it that you want to reference the line by its index value. For example, here's how you would retrieve info about the first source line (of the "Speaker Out" destination line) in our example Mixer. Before calling mixerGetLineInfo() and passing it a MIXERLINE to fill in, you must initialize 3 fields. The ***cbStruct*** field must be set to the size of the MIXERLINE struct you're passing, the ***dwSource*** field must be set to the index value of the source line whose info you wish to retrieve, and the ***dwDestination*** field must be set to the index value of the destination line to which this source line connects.

```
MIXERLINE     mixerline;
MMRESULT      err;

/* Get info about the first source line (of the first destination line) by its index
*/
mixerline.cbStruct = sizeof(MIXERLINE);
```

```
mixerline.dwDestination = 0;
mixerline.dwSource = 0;

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_SOURCE)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

When the above call returns, mixerGetLineInfo() will have filled in our MIXERLINE struct as per the "Internal CD Audio" (mixerline_CD) MIXERLINE struct shown above. So for example, you can extract its line ID number from the MIXERLINE's dwLineID field.

Now, if you want to fetch info about the second source line of the "Speaker Out" destination line, the only thing different is the index value you stuff into the ***dwSource*** field, as so:

```
/* Get info about the second source line by its index */
mixerline.cbStruct = sizeof(MIXERLINE);
mixerline.dwDestination = 0;
mixerline.dwSource = 1;

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_SOURCE)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

When the above call returns, mixerGetLineInfo() will have filled in our MIXERLINE struct as per the "Synth" (mixerline_Synth) MIXERLINE struct shown above.

Now, you should see how you can enumerate the source lines (of each destination line) by their indexes. Here is an example of printing out the names of all destination lines, and their source lines, in our Mixer:

```
MIXERCAPS     mixcaps;
MIXERLINE     mixerline;
MMRESULT      err;
unsigned long i, n, numSrc;

/* Get info about the first Mixer Device */
if (!(err = mixerGetDevCaps((UINT)mixerHandle, &mixcaps, sizeof(MIXERCAPS))))
{
    /* Print out the name of each destination line */
    for (i = 0; i < mixercaps.cDestinations; i++)
    {
        mixerline.cbStruct = sizeof(MIXERLINE);
        mixerline.dwDestination = i;

        if (!(err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_DESTINATION)))
        {
            printf("Destination #%lu = %s\n", i, mixerline.szName);

            /* Print out the name of each source line in this destination */
            numSrc = mixerline.cConnections;
            for (n = 0; n < numSrc; n++)
```

```
            {
                mixerline.cbStruct = sizeof(MIXERLINE);
                mixerline.dwDestination = i;
                mixerline.dwSource = n;

                if (!(err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_SOURCE)))
                {
                    printf("\tSource #%lu = %s\n", i, mixerline.szName);
                }
            }
        }
    }
}
```

Getting info about a line by its ID

Once you know a line's ID number (which you can extract from the MIXERLINE's dwLineID after enumerating the line as shown above), you can later retrieve info on it by referencing its ID (instead of its index). If you're dealing with a source line, you do not need to know the index of the destination line to which the source line is connected. You merely initialize the MIXERLINE's ***dwLineID*** field to the ID number of the desired line, and then specify MIXER_GETLINEINFOF_LINEID when calling mixerGetLineInfo() as so:

```
/* Get info about the "Microphone Input" source line by its ID */
mixerline.cbStruct = sizeof(MIXERLINE);
mixerline.dwLineID = 0x00000003; /* The ID for "Microphone Input" */

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_LINEID)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

The above works with both destination and source lines. Once you know a line's ID, you can directly retrieve info on it without needing to know anything about indexes.

Getting info about a line by its Type

Often, you don't need to know about all of the lines in a Mixer. You may be writing a program that would deal only with a specific type of line. For example, let's say that you're writing a simple MIDI file player. Now, certain components of our example sound card are of no use to you at all. MIDI is not digital audio data, so the "DAC Wave In" component (and all source lines running into it) are of no concern to you. Likewise, the "Internal CD Audio", "DAC Wave Out" and "Microphone Input" source lines to the "Speaker Out" destination line are of no concern to you. The only component on our card which is capable of dealing with the playback of MIDI data is the "Synth" component that goes to the "Speaker Out". It is this line's controls that will affect the playback of MIDI data.

So, rather than enumerating all of the lines in the mixer until you come to the one with the MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER type, mixerGetLineInfo() lets you directly get info about a line that matches your desired type. You merely initialize the MIXERLINE's ***dwComponentType*** field to the desired type of line, and then specify MIXER_GETLINEINFOF_COMPONENTTYPE when calling mixerGetLineInfo() as so:

```
/* Get info about a "Synth" type of source line */
mixerline.cbStruct = sizeof(MIXERLINE);
mixerline.dwComponentType = MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER; /* We want a
```

```
Synth type */

if ((err = mixerGetLineInfo((HMIXEROBJ)mixerHandle, &mixerline,
MIXER_GETLINEINFOF_COMPONENTTYPE)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineInfo()\n", err);
}
```

This will fill in the MIXERLINE struct with info about the first line in the Mixer which has a type of MIXERLINE_COMPONENTTYPE_SRC_SYNTHESIZER. (If there are no such lines in the Mixer, a MIXERR_INVALLINE error is returned). Once you have that info, you can then directly manipulate its controls. This saves having to enumerate and search for a desired line when your needs are specific.

MIXERCONTROL structures, and enumerating controls

You use the mixer API mixerGetLineControls() to retrieve info about controls for a line. This API fills in a MIXERCONTROL struct with info about a control.

Let's examine the MIXERCONTROL struct. As mentioned before, our "Speaker Out" destination line has 2 controls; a volume slider, and a mute switch. Each one of these controls will have one MIXERCONTROL associated with it. Let's examine the MIXERCONTROL for each of these controls:

```
MIXERCONTROL mixerctl_Spkr_Vol = {
    sizeof(MIXERCONTROL),             /* size of a MIXERCONTROL */
    0x00000000,                       /* unique ID # for this control */
    MIXERCONTROL_CONTROLTYPE_VOLUME,  /* type of control */
    MIXERCONTROL_CONTROLF_UNIFORM,    /* flag bits */
    0,                                /* # of items per channels (used only if the
                                      MIXERCONTROL_CONTROLF_MULTIPLE flag bit is also
set) */
    "Volume",                         /* Short name for this control */
    "Speaker Out Volume",             /* Long name for this control */
    0,                                /* Minimum value to which this control can be
set */
    65535,                            /* Maximum value to which this control can be
set */
    0, 0, 0, 0,                       /* These fields are reserved for future use */
    31,                               /* Step amount for the value */
    0, 0, 0, 0, 0,                    /* These fields are reserved for future use */
};

MIXERCONTROL mixerctl_Spkr_Mute = {
    sizeof(MIXERCONTROL),
    0x00000001,                       /* unique ID # for this control */
    MIXERCONTROL_CONTROLTYPE_MUTE,    /* type of control */
    MIXERCONTROL_CONTROLF_UNIFORM,
    0,
    "Mute",                           /* Short name for this control */
    "Speaker Out Mute",               /* Long name for this control */
    0,                                /* Minimum value to which this control can be
set */
    1,                                /* Maximum value to which this control can be
set */
    0, 0, 0, 0,
    0,                                /* Step amount for the value */
```

```
    0, 0, 0, 0, 0,
};
```

There are several things to note above. First, note that each control has a unique ID. These ID numbers don't have to be unique with regard to the IDs of lines. (For example, the control ID of the mixerctl_Spkr_Vol control happens to be the same as the line ID of the mixerline_CD line). But, each control must have an ID unique from any other control, including the controls of other lines. (For example, the volume slider of the "Speaker Out" line can't have the same ID number as the mute switch of the "ADC Wave In" line).

I have set the MIXERCONTROL_CONTROLF_UNIFORM flag. What this means is that, although the "Speaker Out" is stereo (ie, it has 2 channels), there is not a separate volume control for each channel. (There are not individual, left channel and right channel volume settings). There is only one volume setting for both channels, and therefore both channels will be set to the same volume always. (Later, we'll study controls that aren't uniform).

Also note that each control has an appropriate type. The volume slider has a MIXERCONTROL_CONTROLTYPE_VOLUME type, and the mute switch has a MIXERCONTROL_CONTROLTYPE_MUTE type.

The MIXERCONTROL tells you what the min and max values can be for the control. For example, the volume slider can be set to any value inbetween 0 and 65,535. 0 is the minimum setting (ie, volume is lowest), and 65,535 is the maximum setting (ie, volume is loudest). So, does that mean that the volume slider has 65,535 discrete steps? (ie, Can it be set to any value from 0 to 65,535, inclusive)? Not necessarily. You also have to look at the step amount field. This tells you how many valid steps the control has. In this case, we have 31 valid steps. This means that the first valid setting is 0, but the second valid setting is 65,535 - (65,535/31) and the third valid setting is 65,535 - (65,535/(31*2)), etc. In other words, we have only 31 valid settings within that 0 to 65,535 range. (NOTE: the dwMinimum/dwMaximum fields are declared in a union with the lMinimum/lMaximum fields. You'll reference the former two when dealing with a control type with an unsigned value -- ie, a control type whose value is set with a MIXERCONTROLDETAILS_BOOLEAN or MIXERCONTROLDETAILS_UNSIGNED struct. You'll reference the latter two when dealing with a control type with a signed value -- ie, a control type whose value is set with a MIXERCONTROLDETAILS_SIGNED struct).

Enumerating controls is a bit different than enumerating lines. For one thing, you don't employ indexes with controls. Secondly, you can retrieve info about a single control only if you already know its control ID number. Otherwise, you must simultaneously retrieve info for all the controls for a given line.

Obviously, when you're first enumerating the controls for a line, you don't know the ID of each control. So you need to retrieve info on all of the controls with a single call to mixerGetLineControls(). In this case, you need to pass mixerGetLineControls() an array of MIXERCONTROL structs. There must be one struct for every control in the line.

For example, we know that our "Speaker Out" destination line has 2 controls; a volume slider, and a mute switch. (Remember that its MIXERLINE's cControls field is 2). So, to retrieve information about them, we must pass mixerGetLineControls() an array containing 2 MIXERCONTROL structs. We must also pass the value MIXER_GETLINECONTROLSF_ALL to indicate we want info on all the controls. We also pass a special structure called a MIXERLINECONTROLS which we must first initialize. This structure tells mixerGetLineControls() for which line we want to receive info about its controls. We also provide the pointer to our array of MIXERCONTROL structs via this additional struct. Here then is an example of retrieving info about all the controls in the "Speaker Out" line.

```
/* Let's just declare an array of 2 MIXERCONTROL structs since
   we know that the "Speaker Out" has 2 controls. For a real program,
   you typically won't know ahead of time how big an array you'll need,
   and therefore would instead allocate an appropriately sized array */
MIXERCONTROL       mixerControlArray[2];
MIXERLINECONTROLS  mixerLineControls;
MMRESULT           err;

mixerLineControls.cbStruct = sizeof(MIXERLINECONTROLS);

/* The "Speaker Out" line has a total of 2 controls. And
```

```
   that's how many we want to retrieve info for here */
mixerLineControls.cControls = 2;

/* Tell mixerGetLineControls() for which line we're retrieving info.
   We do this by putting the desired line's ID number in dwLineID.
   The "Speaker Out" line has an ID of 0xFFFF0000 */
mixerLineControls.dwLineID = 0xFFFF0000;

/* Give mixerGetLineControls() the address of our array of
   MIXERCONTROL structs big enough to hold info on all controls */
mixerLineControls.pamxctrl = &mixerControlArray[0];

/* Tell mixerGetLineControls() how big each MIXERCONTROL is. This
   saves having to initialize the cbStruct of each individual
   MIXERCONTROL in the array */
mixerLineControls.cbmxctrl = sizeof(MIXERCONTROL);

/* Retrieve info on all controls for this line simultaneously */
if ((err = mixerGetLineControls((HMIXEROBJ)mixerHandle, &mixerLineControls,
MIXER_GETLINECONTROLSF_ALL)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineControls()\n", err);
}
```

When mixerGetLineControls() returns above, our mixerControlArray[] array will have been filled in. mixerControlArray[0] will have been filled in as per the mixerctl_Spkr_Vol MIXERCONTROL struct shown above, and mixerControlArray[1] will have been filled in as per the mixerctl_Spkr_Mute MIXERCONTROL struct shown above. So, you can, for example, extract their control ID numbers from their respective MIXERCONTROL's dwControlID field.

It's possible to retrieve info on only one control, if you know its ID or Type. But it is not possible to retrieve info on more than one control, but less than the total number of controls. For example, assume that our "Speaker Out" had 5 controls (instead of only 2). You couldn't retrieve info on only the first 3, for example. You can either retrieve info on one (out of the 5) at a time, or all 5 of them at once. (ie, It's either one at a time, or all).

Getting info about a control by its ID

Once you know a control's ID number (which you can extract from its MIXERCONTROL's dwControlID field after enumerating the control as shown above), you can later retrieve info on just this one control by referencing its ID. You don't even need to know the ID of the line to which this control belongs. And you aren't forced to retrieve info for all of the other controls in this line. You merely initialize the MIXERLINECONTROLS's ***dwControlID*** field to the ID number of the desired control, and then specify MIXER_GETLINECONTROLSF_ONEBYID when calling mixerGetLineControls() as so:

```
/* We need only 1 MIXERCONTROL struct since
   we're fetching info for only 1 control */
MIXERCONTROL       mixerControlArray;
MIXERLINECONTROLS  mixerLineControls;
MMRESULT           err;

mixerLineControls.cbStruct = sizeof(MIXERLINECONTROLS);

/* We want to fetch info on only 1 control */
mixerLineControls.cControls = 1;

/* Tell mixerGetLineControls() for which control we're retrieving
   info. We do this by putting the desired control's ID number in
   dwControlID. The "Speaker Out" line's volume slider has an ID
```

```
   of 0x00000000 */
mixerLineControls.dwControlID = 0x00000000;

/* Give mixerGetLineControls() the address of the
   MIXERCONTROL struct to hold info */
mixerLineControls.pamxctrl = &mixerControlArray;

/* Tell mixerGetLineControls() how big the MIXERCONTROL is. This
   saves having to initialize the cbStruct of the MIXERCONTROL itself */
mixerLineControls.cbmxctrl = sizeof(MIXERCONTROL);

/* Retrieve info on only the volume slider control for the "Speaker Out" line */
if ((err = mixerGetLineControls((HMIXEROBJ)mixerHandle, &mixerLineControls,
MIXER_GETLINECONTROLSF_ONEBYID)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineControls()\n", err);
}
```

Getting info about a control by its Type

Often, you don't need to know about all of the controls in a line. You may be writing a program that would deal only with a specific type of control. For example, let's say that you're writing a simple MIDI file player and the only control you want to present to the enduser is the volume slider of the "Synth". You saw earlier how you could search for that MIDI playback component by Type and fill in a MIXERLINE struct with info about it, such as its line ID. You can then use that line ID to search for a particular type of control within that line, For example, you can search for a control with a type of MIXERCONTROL_CONTROLTYPE_VOLUME.

So, rather than enumerating all of the controls in the line until you come to the one with the MIXERCONTROL_CONTROLTYPE_VOLUME type, mixerGetLineControls() lets you directly get info about a line that matches your desired type. You merely initialize the MIXERLINECONTROLS's ***dwControlType*** field to the desired type of control, and then specify MIXER_GETLINECONTROLSF_ONEBYTYPE when calling mixerGetLineControls() as so. (Assume that you've already fetched the line ID for the "Synth" line, and stored it in "SynthID").

```
MIXERCONTROL       mixerControlArray;
MIXERLINECONTROLS  mixerLineControls;
MMRESULT           err;

mixerLineControls.cbStruct = sizeof(MIXERLINECONTROLS);

/* Tell mixerGetLineControls() for which line we're retrieving info.
   We do this by putting the desired line's ID number in dwLineID */
mixerLineControls.dwLineID = SynthID;

/* We want to fetch info on only 1 control */
mixerLineControls.cControls = 1;

/* Tell mixerGetLineControls() for which type of control we're
   retrieving info. We do this by putting the desired control type
   in dwControlType */
mixerLineControls.dwControlType = MIXERCONTROL_CONTROLTYPE_VOLUME;

/* Give mixerGetLineControls() the address of the MIXERCONTROL
   struct to hold info */
mixerLineControls.pamxctrl = &mixerControlArray;

/* Tell mixerGetLineControls() how big the MIXERCONTROL is. This
   saves having to initialize the cbStruct of the MIXERCONTROL itself */
```

```
mixerLineControls.cbmxctrl = sizeof(MIXERCONTROL);

/* Retrieve info on only any volume slider control for this line */
if ((err = mixerGetLineControls((HMIXEROBJ)mixerHandle, &mixerLineControls,
MIXER_GETLINECONTROLSF_ONEBYTYPE)))
{
    /* An error */
    printf("Error #%d calling mixerGetLineControls()\n", err);
}
```

This will fill in the MIXERCONTROL struct with info about the first control in the line which has a type of MIXERCONTROL_CONTROLTYPE_VOLUME. (If there are no such controls in the line, a MIXERR_INVALCONTROL error is returned). Once you have that info, you can then directly manipulate that control. For example, you can extract its control ID number from the MIXERCONTROL's dwControlID field. This saves having to enumerate and search for a desired control when your needs are specific.

Retrieving and setting a control's value

Now we're getting to the whole purpose of a mixer; to retrieve a control's value (so that you can display its current setting to the enduser), and to set a control's value (so that you can allow the enduser to adjust it).

To retrieve or set a control's value, you must know its control ID. You use mixerGetControlDetails() to retrieve its current value, and mixerSetControlDetails() to set it to a specific value. These functions utilize a MIXERCONTROLDETAILS struct. You initialize certain fields to tell mixerGetControlDetails()/mixerSetControlDetails() what control whose value you're retrieving/setting, and you also supply a pointer to another structure that will contain the actual value.

For example, let's consider retrieving the current value of the "Speaker Out" line's volume slider. By now, you know how to get info on that control, for example, its ID number. In order to retrieve a control's value, we need to supply a special structure into which its value is returned. What kind of structure do we use? Well, that depends upon what type of control it is. The volume slider is a MIXERCONTROL_CONTROLTYPE_VOLUME type. If you go back to the chart about the fader class of controls, it tells you that its value is retrieved using a MIXERCONTROLDETAILS_UNSIGNED struct. This struct has only one field into which the control's value is returned; dwValue. So we supply a MIXERCONTROLDETAILS_UNSIGNED struct to mixerGetControlDetails() (via the MIXERCONTROLDETAILS struct). Here then is an example, of retrieving and printing the current value of the "Speaker Out" line's volume slider:

```
/* We need a MIXERCONTROLDETAILS_UNSIGNED struct to retrieve the
   value of a control whose type is MIXERCONTROL_CONTROLTYPE_VOLUME */
MIXERCONTROLDETAILS_UNSIGNED value;
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerGetControlDetails() which control whose value we
   want to retrieve. We do this by putting the desired control's
   ID number in dwControlID. Remember that the "Speaker Out" line's
   volume slider has an ID of 0x00000000 */
mixerControlDetails.dwControlID = 0x00000000;

/* This is always 1 for a MIXERCONTROL_CONTROLF_UNIFORM control */
mixerControlDetails.cChannels = 1;

/* This is always 0 except for a MIXERCONTROL_CONTROLF_MULTIPLE control */
mixerControlDetails.cMultipleItems = 0;

/* Give mixerGetControlDetails() the address of the
```

```
   MIXERCONTROLDETAILS_UNSIGNED struct into which to return the value */
mixerControlDetails.paDetails = &value;

/* Tell mixerGetControlDetails() how big the MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Retrieve the current value of the volume slider control for this line */
if ((err = mixerGetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_GETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerGetControlDetails()\n", err);
}
else
{
    printf("It's value is %lu\n", value.dwValue);
}
```

To set a control's value, you simply fill in the special structure that contains the value, and pass it to mixerSetControlDetails(). You also specify MIXER_SETCONTROLDETAILSF_VALUE. Here is an example of setting the "Speaker Out" line's volume slider to a value of 31:

```
MIXERCONTROLDETAILS_UNSIGNED value;
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerSetControlDetails() which control whose value we
   want to set. We do this by putting the desired control's
   ID number in dwControlID. Remember that the "Speaker Out" line's
   volume slider has an ID of 0x00000000 */
mixerControlDetails.dwControlID = 0x00000000;

/* This is always 1 for a MIXERCONTROL_CONTROLF_UNIFORM control */
mixerControlDetails.cChannels = 1;

/* This is always 0 except for a MIXERCONTROL_CONTROLF_MULTIPLE control */
mixerControlDetails.cMultipleItems = 0;

/* Give mixerSetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED struct into which we place the value */
mixerControlDetails.paDetails = &value;

/* Tell mixerSetControlDetails() how big the MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Store the value */
value.dwValue = 31;

/* Set the value of the volume slider control for this line */
if ((err = mixerSetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_SETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerSetControlDetails()\n", err);
}
```

Multi-channel controls

As mentioned, when a control's MIXERCONTROL_CONTROLF_UNIFORM flag bit is set, then it doesn't have individual values for each one of its channels. For example, with the "Speaker Out", there is not a separate volume for the left and right channels of this stereo line.

But if a control's MIXERCONTROL_CONTROLF_UNIFORM flag bit is not set, and it has more than one channel, then each channel has its own value. For this reason, you'll need more than one of the special structures to retrieve/set the values for all channels. For example, let's assume that the volume slider for the "Speaker Out" does not have its MIXERCONTROL_CONTROLF_UNIFORM flag bit set. Since the Speaker Out line has 2 channels, that means that we need 2 MIXERCONTROLDETAILS_UNSIGNED structs to retrieve/set the value for the Left and Right channels' volumes respectively. We need to use an array of MIXERCONTROLDETAILS_UNSIGNED structs. The first MIXERCONTROLDETAILS_UNSIGNED struct will be for the first (ie, Left) channel, and the second MIXERCONTROLDETAILS_UNSIGNED struct will be for the second (ie, Right) channel. Here is an example of retrieving the values of the left and right channels of our volume slider for the "Speaker Out" line:

```
/* We need 2 MIXERCONTROLDETAILS_UNSIGNED structs to retrieve the
   values of a stereo control that is not MIXERCONTROL_CONTROLF_UNIFORM */
MIXERCONTROLDETAILS_UNSIGNED value[2];
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerGetControlDetails() which control whose value we
   want to retrieve. We do this by putting the desired control's
   ID number in dwControlID. Remember that the "Speaker Out" line's
   volume slider has an ID of 0x00000000 */
mixerControlDetails.dwControlID = 0x00000000;

/* We want to retrieve values for both channels */
mixerControlDetails.cChannels = 2;

/* This is always 0 except for a MIXERCONTROL_CONTROLF_MULTIPLE control */
mixerControlDetails.cMultipleItems = 0;

/* Give mixerGetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED array into which to return the values */
mixerControlDetails.paDetails = &value[0];

/* Tell mixerGetControlDetails() how big each MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Retrieve the current values of both channels */
if ((err = mixerGetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_GETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerGetControlDetails()\n", err);
}
else
{
    printf("The left channel's volume is %lu\n", value[0].dwValue);
    printf("The right channel's volume is %lu\n", value[1].dwValue);
}
```

To set the values of both channels, you fill in the values of both MIXERCONTROLDETAILS_UNSIGNED structs. Here is an

example of setting the left channel's volume to 31 and the right channel's volume to 0.

```
/* We need 2 MIXERCONTROLDETAILS_UNSIGNED structs to set the
   values of a stereo control that is not MIXERCONTROL_CONTROLF_UNIFORM */
MIXERCONTROLDETAILS_UNSIGNED value[2];
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerSetControlDetails() which control whose value we
   want to set. We do this by putting the desired control's
   ID number in dwControlID. Remember that the "Speaker Out" line's
   volume slider has an ID of 0x00000000 */
mixerControlDetails.dwControlID = 0x00000000;

/* We want to set values for both channels */
mixerControlDetails.cChannels = 2;

/* This is always 0 except for a MIXERCONTROL_CONTROLF_MULTIPLE control */
mixerControlDetails.cMultipleItems = 0;

/* Give mixerSetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED structs into which we place the values */
mixerControlDetails.paDetails = &value[0];

/* Tell mixerSetControlDetails() how big each MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Store the left channel's value */
value[0].dwValue = 31;

/* Store the right channel's value */
value[1].dwValue = 0;

/* Set the left/right values of the volume slider control for this line */
if ((err = mixerSetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_SETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerSetControlDetails()\n", err);
}
```

Of course, a control may have even more than 2 channels. You always need an array of special structures large enough to accomodate all channels for a given control, so typically, you'll allocate the array as needed.

It is not legal to retrieve or set only some of the channels. For example, if a control has 8 channels, it's not legal to try to retrieve the values of only the first 2 channels. You must always retrieve/set the values of all channels simultaneously. But there is one caveat to this rule. It concerns setting a value. If you set only the value for the first channel, then mixerSetControlDetails() automatically treats the control as if it was MIXERCONTROL_CONTROLF_UNIFORM. The net result is that all channels get set to that one value. So, you can quickly set all channels to the same value just by setting the value of only the first channel.

Multi-item controls

You won't often encounter multi-item controls. A multi-item control is one that has several values associated with each channel. An example could be a graphic equalizer control. Let's study a simple example. Assume that you have the following 3

band graphic equalizer built into a sound card:

This control has three values associated with it -- the value for the "Low" band, the value for the "Mid" band, and the value for the "High" band. (ie, It is assumed that each band can be set to a different value, otherwise it would be a fairly useless graphic EQ). The way that this would be represented is a a multi-item control. It has 3 items (ie, values) associated with it.

Let's further assume that this control is in the "Speaker Out" line.

Let's examine its MIXERCONTROL struct. A multi-item control has the MIXERCONTROL_CONTROLF_MULTIPLE bit set of its MIXERCONTROL's ***dwControlType*** field. The MIXERCONTROL's ***cMultipleItems*** field will also tell how many items are in each channel.

```
MIXERCONTROL mixerctl_EQ = {
    sizeof(MIXERCONTROL),           /* size of a MIXERCONTROL */
    0x00000002,                     /* unique ID # for this control */
    MIXERCONTROL_CONTROLTYPE_EQUALIZER, /* type of control */
    MIXERCONTROL_CONTROLF_UNIFORM|MIXERCONTROL_CONTROLF_MULTIPLE,   /* flag bits */
    3,                              /* # of items per channel */
    "EQ",                           /* Short name for this control */
    "Graphic Equalizer",            /* Long name for this control */
    0,                              /* Minimum value to which this control can be
set */
    65535,                          /* Maximum value to which this control can be
set */
    0, 0, 0, 0,                     /* These fields are reserved for future use */
    31,                             /* Step amount for the value */
    0, 0, 0, 0, 0,                  /* These fields are reserved for future use */
};
```

First of all, note that the ID is different than all of the other controls for this Mixer. Also, note that its MIXERCONTROL_CONTROLF_MULTIPLE flag bit is set. The ***cMultipleItems*** field is set to 3 to indicate that there are 3 items per channel. (But since I made this control MIXERCONTROL_CONTROLF_UNIFORM, it still has only 3 values total, even though the "Speaker Out" line is stereo. In other words, the value for each band affects both channels equally).

To query the values for all 3 bands, we need an array of 3 structures to fetch the values. What kind of structures? Well, the MIXERCONTROL_CONTROLTYPE_EQUALIZER control type is of the fader class, and you'll remember that all the types in that class use a MIXERCONTROLDETAILS_UNSIGNED struct to set/fetch values. Here's how we query the current values of the 3 bands:

```
/* We need 3 MIXERCONTROLDETAILS_UNSIGNED structs to retrieve the
   values of this control */
MIXERCONTROLDETAILS_UNSIGNED value[3];
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerGetControlDetails() which control whose value we
   want to retrieve */
mixerControlDetails.dwControlID = 0x00000002;
```

```
/* It's a MIXERCONTROL_CONTROLF_UNIFORM control, so the values
   for all channels are the same as the first */
mixerControlDetails.cChannels = 1;

/* There are 3 items per channel */
mixerControlDetails.cMultipleItems = 3;

/* Give mixerGetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED array into which to return the values */
mixerControlDetails.paDetails = &value[0];

/* Tell mixerGetControlDetails() how big each MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Retrieve the current values of all 3 bands */
if ((err = mixerGetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_GETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerGetControlDetails()\n", err);
}
else
{
    printf("The Low band is %lu\n", value[0].dwValue);
    printf("The Mid band is %lu\n", value[1].dwValue);
    printf("The High band is %lu\n", value[2].dwValue);
}
```

To set the values of all 3 bands, you fill in the values of the MIXERCONTROLDETAILS_UNSIGNED structs. Here is an example of setting the Low band to 31, the Mid band to 0, and the High band to 62.

```
/* We need 3 MIXERCONTROLDETAILS_UNSIGNED structs to set the
   values of the 3 bands */
MIXERCONTROLDETAILS_UNSIGNED value[3];
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerSetControlDetails() which control whose values we
   want to set */
mixerControlDetails.dwControlID = 0x00000002;

/* The values for all channels are the same as the first channel */
mixerControlDetails.cChannels = 1;

/* We're setting all 3 bands */
mixerControlDetails.cMultipleItems = 3;

/* Give mixerSetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED structs into which we place the values */
mixerControlDetails.paDetails = &value[0];

/* Tell mixerSetControlDetails() how big each MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Store the Low band's value */
value[0].dwValue = 31;
```

```
/* Store the Mid band's value */
value[1].dwValue = 0;

/* Store the High band's value */
value[2].dwValue = 62;

/* Set the values of the 3 bands for this control */
if ((err = mixerSetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_SETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerSetControlDetails()\n", err);
}
```

Now let's remove that MIXERCONTROL_CONTROLF_UNIFORM flag bit of the MIXERCONTROL. Now, each channel's items have individual values. Since the "Speaker Out" has 2 channels, that means we have a total of 2 (channels) * 3 (items), or 6 values total for this control. Our graphic EQ now looks like this:

Left Channel **Right Channel**

We're going to need 6 MIXERCONTROLDETAILS_UNSIGNED structs to fetch the values of all items in all channels. Oh, and in the preceding example, I assumed what the labels were for those items. What you really should do is ask the Mixer to provide you with the labels if you wish to print them out. To do this, you must supply an array of MIXERCONTROLDETAILS_LISTTEXT structs, just like how you use an array of MIXERCONTROLDETAILS_UNSIGNED structs to fetch the values of all items in all channels

```
/* We need 6 MIXERCONTROLDETAILS_UNSIGNED structs to retrieve the
   values of this control */
MIXERCONTROLDETAILS_UNSIGNED value[6];
/* We need 6 MIXERCONTROLDETAILS_LISTTEXT structs to retrieve the
   labels of all items */
MIXERCONTROLDETAILS_LISTTEXT label[6];
MIXERCONTROLDETAILS          mixerControlDetails;
MMRESULT                     err;

mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);

/* Tell mixerGetControlDetails() which control whose value we
   want to retrieve */
mixerControlDetails.dwControlID = 0x00000002;

/* Our "Speaker Out" has 2 channels */
mixerControlDetails.cChannels = 2;

/* There are 3 items per channel */
mixerControlDetails.cMultipleItems = 3;
```

```
/* Give mixerGetControlDetails() the address of the
   MIXERCONTROLDETAILS_UNSIGNED array into which to return the values */
mixerControlDetails.paDetails = &value[0];

/* Tell mixerGetControlDetails() how big each MIXERCONTROLDETAILS_UNSIGNED is */
mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_UNSIGNED);

/* Retrieve the current values of all 3 bands for both channels */
if ((err = mixerGetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_GETCONTROLDETAILSF_VALUE)))
{
    /* An error */
    printf("Error #%d calling mixerGetControlDetails()\n", err);
}
else
{
    unsigned long   i,n;

    /* Let's fetch the labels of all the items */

    mixerControlDetails.cbStruct = sizeof(MIXERCONTROLDETAILS);
    mixerControlDetails.dwControlID = 0x00000002;
    mixerControlDetails.cChannels = 2;
    mixerControlDetails.cMultipleItems = 3;

    /* Give mixerGetControlDetails() the address of the
       MIXERCONTROLDETAILS_LISTTEXT array into which to return the labels */
    mixerControlDetails.paDetails = &label[0];

    /* Tell mixerGetControlDetails() how big each MIXERCONTROLDETAILS_LISTTEXT is */
    mixerControlDetails.cbDetails = sizeof(MIXERCONTROLDETAILS_LISTTEXT);

    /* Retrieve the labels of all items for both channels, Note
       that I specify MIXER_GETCONTROLDETAILSF_LISTTEXT */
    if ((err = mixerGetControlDetails((HMIXEROBJ)mixerHandle, &mixerControlDetails,
MIXER_GETCONTROLDETAILSF_LISTTEXT)))
    {
        /* An error */
        printf("Error #%d calling mixerGetControlDetails()\n", err);
    }
    else
    {
        /* Print the values of all items */
        for (i = 0; i < 2; i++)
        {
            printf("Channel %lu:\n", i+1);

            for (n = 0; n < 3; n++)
            {
                printf("\tThe %s item is %lu\n", label[3 * i + n].szName, value[3 * i
+ n].dwValue);
            }
        }
    }
}
```

It is not legal to retrieve or set only some of items of a control. For example, if a control has 8 items, it's not legal to try to retrieve the values of only the first 2 items. You must always retrieve/set the values of all items for all channels simultaneously. The one caveat to this rule is that if you set only the items for the first channel, then mixerSetControlDetails()

automatically treats the control as if it was MIXERCONTROL_CONTROLF_UNIFORM. The net result is that the items for all channels get set to the same values as the first channel. So, you can quickly set all channels to the same values just by setting the items of only the first channel.

Notification of changes

In my above examples, I have shown that the mixer device is always opened with mixerOpen() and its handle retrieved for use with other mixer functions. This is not always necessary. In fact, the mixer API has been designed so that, instead of passing a handle to an open mixer wherever any mixer function specifies such an arg, you can instead pass the mixer ID number of the desired mixer. So, you do not need to open a mixer explicitly.

But there are advantages to explicitly opening a mixer (with mixerOpen()) for as long as you need to do operations upon it. First, this prevents the mixer from somehow being "unloaded" (presumably by the audio card's driver). Secondly, when you have a mixer open, you can instruct Windows to send you a special message (to the Window procedure of some window that you've created) whenever any line's state has changed (for example, if the line is muted), or the value of some control has been changed. You get sent such a message not only when you change a line's state or a control's value, but also when any other program also opens that mixer (ie, more than one program can open the same mixer simultaneously) and changes a line or control. So, you can keep your program in sync with any changes made to the mixer by any other program.

When you call mixerOpen(), you specify the handle to your own window which you wish Windows to send its special "mixer messages". Pass your window handle as the third arg to mixerOpen(). Also specify CALLBACK_WINDOW as the last arg.

There are 2 special "mixer messages". MM_MIXM_LINE_CHANGE is sent to your window procedure whenever a line's state is changed. MM_MIXM_CONTROL_CHANGE is sent whenever a control's value is changed.

For MM_MIXM_LINE_CHANGE, the WPARAM argument to your window procedure is the handle of the open mixer whose line has changed. The LPARAM argument is the ID number of the line whose state has changed.

For MM_MIXM_CONTROL_CHANGE, the WPARAM argument to your window procedure is the handle of the open mixer whose line has changed. The LPARAM argument is the ID number of the control whose value has changed.

Conclusion

The Mixer API is one of the most complicated APIs regarding Windows multi-media. It may take awhile to absorb this tutorial and apply it to your needs. But the Mixer API gives you a way to make adjustments to the settings of any sound card without needing to be specifically written for that card.

For more information about structures and APIs about mixers, see Microsoft Developer Network's reference upon audio mixers

Microsoft makes a freely downloadable example of using the Mixer API. But I found this code to be too terse in its use of comments, and also had a lot of code not really related to the Mixer API and not needed. I have pared down this example to mostly code pertinent to the Mixer API, and profusely commented the code. You can download my version of Microsoft's Mixer Device Example to show how to display information on all Mixer devices and their lines/controls, as well as adjust controls' values. Included are the Project Workspace files for Visual C++ 4.0, but since it is an ordinary C Windowed app, any Windows C compiler should be able to compile it.

This information is not yet completed.

Windows' MultiMedia (File) I/O (MMIO) functions were created to provide programmers with some convenient routines to help read and write RIFF files, such as WAVE files. (You should read the article upon Interchange File Format before proceeding).

The MMIO functions allow file buffering (ie, much like the standard C functions such as fopen, fread, and fwrite), which is very handy given the fact that RIFF files often involve numerous reads and writes of a few bytes. Furthermore, the Windows functions take care of this file buffering completely transparently. (ie, Your program simply tells Windows to use file buffering, and Windows takes care of all details, including allocating a buffer if you do not wish to supply one yourself).

The MMIO functions allow quick and relatively easy locating, reading, and writing of chunks. That spares your program from having to do a lot of file seeking and parsing.

The MMIO functions can work with files on disks, or file images that have been completely loaded into memory.

Finally, you can install your own "custom I/O" procedures (that are called by Windows MMIO functions such as mmioRead or mmioWrite) to support streaming data to devices other than disks or memory files. In that way, any program that uses MMIO functions to read/write data can be made to transparently stream that data to/from your device.

Opening a RIFF file for reading or writing

NOTE: For the sake of demonstration, the following examples will be shown with a WAVE file, although there are other types of RIFF files to which the MMIO functions are applicable.

The first thing you must do is open a file for reading, writing, or both. This is accomplished by calling mmioOpen(). The first argument is a pointer to a filename.

The second argument is a pointer to an MMIOOPEN structure. This does not need to be supplied unless you desire some special operations such as parsing a file image in memory, or forcing Windows to use a file buffer that you've allocated for I/O buffering, or supplying the address of a custom I/O procedure.

The third argument is some flags OR'ed together. To open a file for reading, specify the flag MMIO_READ. To create a new file to which you'll be writing data, specify the flag MMIO_WRITE. To open a file that you'll be both reading and writing, specify the flag MMIO_READWRITE.

The MMIO_EXCLUSIVE, MMIO_DENYWRITE, and MMIO_DENYREAD flags are file-sharing flags. For example, to prevent another program from simultaneously writing to the file while your program has it open, specify MMIO_DENYWRITE. MMIO_EXCLUSIVE denies other programs read or write access simultaneously. MMIO_DENYREAD prevents other programs from reading (but not writing) to the file simultaneously

Specify the MMIO_ALLOCBUF flag if you wish Windows to allocate a buffer for buffered file I/O. The default size will be 8K, but if you wish a different size, then you must supply a MMIOOPEN structure with its cchBuffer field set to the desired buffer size in bytes. If cchBuffer is 0, the default buffer size is used. (If you are providing your own I/O buffer, do not set the MMIO_ALLOCBUF flag).

There are some other flags that will be discussed later.

If successful, mmioOpen() returns a handle to the open file. This handle can be used only with MMIO functions.

So, to open the file C:\Windows\Chord.wav for reading:

```
HMMIO hmmio;
```

```
/* Open the file for reading with buffered I/O. Let windows use its default internal
buffer */
if (!(hmmio = mmioOpen("C:\\WINDOWS\\CHORD.WAV", 0, MMIO_READ|MMIO_ALLOCBUF)))
{
        printf("An error opening the file!\n");
}
```

Locating and reading the Group Header

The first thing that you'll want to do after you open a file for reading is to check that it contains the desired type of data within it. For example, let's say that you want WAVE data. To check that there is a WAVE format within the open file (ie, it could be a collection of various types of data -- a CAT or LIST), and to locate the file pointer to the Group header (ie, the WAVE's RIFF header) and read in that Group Header, you need to allocate a "chunk information" (MMCKINFO) structure, set its fccType field to the ID that you wish to seek out (here it's 'WAVE'), and pass that structure to mmioDescend() with the flag MMIO_FINDRIFF. mmioDescend() then dives into the file and seeks to that RIFF WAVE header (if there is one in the file) and reads it in. (Windows docs refer to reading a chunk as "descending" into it).

Note that there is an MMIO function, mmioFOURCC(), to help you initialize the fccType field with an IFF ID. An IFF ID is 4 characters, but treated as one unsigned long in Motorola (big endian) order.

Here is an example of this:

```
MMCKINFO mmckinfoParent;   /* for the Group Header */

/* Tell Windows to locate a WAVE Group header somewhere in the file, and read it in.
 This marks the start of any embedded WAVE format within the file */
mmckinfoParent.fccType = mmioFOURCC('W', 'A', 'V', 'E');
if (mmioDescend(hmmio, (LPMMCKINFO)&mmckinfoParent, 0, MMIO_FINDRIFF))
{
        /* Oops! No embedded WAVE format within this file */
        printf("ERROR: This file doesn't contain a WAVE!\n");
}
else
{
        /* Here you may seek to, and read in other chunks */
}
```

Locating and reading a chunk

After you locate to the Group Header, the next thing that you'll likely want to do is read in the chunks of that group. For example, with the WAVE format, you'll at least want to read in the **fmt** chunk to get information about the WAVE, and then locate to the **data** chunk and read in the waveform data.

You locate the file pointer to a chunk header in much the same way as locating to a Group header. But, you'll need a second "chunk information" structure. You set its fccType field to the chunk ID that you wish to seek out (here we'll seek out a 'fmt ' chunk), and pass that structure to mmioDescend() with the flag MMIO_FINDCHUNK, and also pass the MMCKINFO that you filled in above (with the Group Header). mmioDescend() then dives into the file and seeks to that chunk's header (if there is one in the file), and read it in. (NOTE: Only the 8 byte chunk header is read in -- not the chunk's data).

Here is an example of reading the fmt chunk header:

```
MMCKINFO mmckinfoSubchunk;   /* for finding chunks within the Group */

/* Tell Windows to locate the WAVE's "fmt " chunk and read in its header */
mmckinfoSubchunk.ckid = mmioFOURCC('f', 'm', 't', ' ');
if (mmioDescend(hmmio, &mmckinfoSubchunk, &mmckinfoParent, MMIO_FINDCHUNK))
{
        /* Oops! The required fmt chunk was not found! */
        printf("ERROR: Required fmt chunk was not found!\n");
}
else
{
        /* Here you may read in the chunk's data */
}
```

The mmckinfoSubchunk structure's cksize field now contains the chunk's chunkSize.

Reading a chunk's data

After reading in a chunk's header as above, you can then read in its data using the mmioRead() function.

It just so happens that a WAVEFORMATEX structure can hold all the data in WAVE's fmt chunk verbatim. So, you need to make only one call to mmioRead() to load in that chunk's data, as so:

```
WAVEFORMATEX waveFormat;   /* for reading a fmt chunk's data */

/* Tell Windows to read in the "fmt " chunk into a WAVEFORMATEX structure */
if (mmioRead(hmmio, (HPSTR)&waveFormat, mmckinfoSubchunk.cksize) !=
(LRESULT)mmckinfoSubchunk.cksize)
{
        /* Oops! */
        printf("ERROR: reading the fmt chunk!\n");
}
```

Ascending out of a chunk

After you've used mmioDescend() to locate and read in a chunk's header and/or data, if you plan to read any other chunks in the file, you need to "ascend" out of any chunk that you've mmioDescend()'ed into. You can't mmioDescend() into another chunk without first ascending out of any chunk you're currently reading. You use the mmioAscend() function to do so. You pass it the MMCKINFO that you used to mmioDescend() into the current chunk.

For example, to ascend out of that fmt chunk whose data we were reading above:

```
mmioAscend(hmmio, &mmckinfoSubchunk, 0);
```

The purpose of this function is to allow the mmio functions to do some necessary, internal bookkeeping with the file position pointer inbetween you moving it from chunk to chunk via mmioDescend().

You can then use the same MMCKINFO with mmioDescend() to locate/read another chunk. (ie, You don't need a separate MMCKINFO with every chunk that you wish to read).

Closing a file

After you're done reading/writing to a file, you must close it. You do this by passing the handle which you obtained from mmioOpen() to the function mmioClose() as so:

```
mmioClose(hmmio, 0);
```

You can download my WaveParse C example to show how to use the MMIO functions to open, read, and display some information about a WAVE file. Included are the Project Workspace files for Visual C++ 4.0, but since it is a console app, any Windows C compiler should be able to compile it. Remember that all apps should include MMSYSTEM.H and link with WINMM.LIB (or MMSYSTEM.LIB if Win3.1). This is a ZIP archive. Use an unzip utility that supports long filenames.

MIDI/Audio cards for a PC

The Old Days

Soon after the IBM "personal computer" (ie, PC) first appeared in the consumer market in 1984, companies began to make cards that plugged inside of the computer in order to play and record music/audio/speech (since the IBM PC didn't have that capability built in). Some of the cards dealt only with digital audio playback and recording. Some other cards dealt with only MIDI input and output. Some cards even did both. (Remember that digital audio and MIDI are two entirely different things, although both typically serve the purpose of recording and playing musical performances).

At the time, the only operating system much in use for the IBM PC was called ***MS-DOS***. It was Microsoft's precursor to Windows, and it was a text-based (ie, no graphical interface like we have now) operating system. The MS-DOS operating system was very primitive when it came to supporting audio/MIDI hardware. In fact, all that it supported was the internal PC speaker. If an MS-DOS program wanted to use any other audio hardware, such as a MIDI interface like the Roland MPU-401, or a sound card like the Creative Labs' Sound Blaster (ie, back then, these were the two most popular add-on cards for MIDI, and digital audio, respectively), the program had to manipulate the audio hardware directly, controlling it in order to make sound. If a particular program didn't directly support your particular audio card, then you wouldn't get sound from that program.

Because the very first, widely accepted MIDI interface for a PC was the Roland MPU-401, almost all MS-DOS MIDI programs directly supported this card. Because the very first, widely accepted Sound Card for a PC was the Creative Labs Sound Blaster, almost all MS-DOS game programs directly supported this card. (MS-DOS MIDI programs didn't support the SB because MIDI users weren't at all interested in the awful FM synthesis and very poor MIDI handling of the early SB. Likewise, MS-DOS game programs didn't support the MPU-401 because game players weren't interested in buying expensive external MIDI sound modules just to hear their game music. Furthermore, game programs needed to play digitized voice and sound effects on a card with a Digital to Analog converter, or DAC, which the SB had and the MPU-401 didn't. So, it was almost impossible to find audio hardware that could be used with both game software and music software).

Lots of companies making MIDI interfaces subsequently offered MPU-401 hardware compatibility so that their products would work with MS-DOS MIDI software that directly manipulated an MPU-401. For example, MusicQuest made the MQX-16. Likewise, lots of companies making Sound Cards offered Sound Blaster hardware compatibility so that their products would work with MS-DOS game software that directly manipulated an SB. For example, MediaVision made the PAS-16.

But typically, if you wanted both MIDI and digital audio support in your computer, then you had to not only buy two different cards to use with your software, but one of the cards had to be 100% hardware compatible with an MPU-401 and the other card had to be 100% hardware compatible with an SB. If you bought an MPU-401 clone from some company other than Roland, or an SB clone from some company other than Creative Labs, you often ran into hardware compatibility problems which made the MS-DOS software not work with the card. So, in the old days, hardware incompatibilies were very

common. Also, the market was prevented from innovating beyond the MPU-401 and SB "standards" because the hardware was tied to the software and vice versa. It was a catch-22 situation. The hardware manufacturers couldn't innovate because then their new hardware designs wouldn't work with any software. And the software programmers couldn't innovate because the hardware manufacturers were too reluctant to invest in making new hardware that didn't already have a lot of software support. (Consumers shunned hardware that wasn't supported by a lot of software, for good reason -- you can't do much with hardware that has little software support). So consumers were limited to buying hardware that had only the features and performance of an MPU-401 or SB, and software that supported only one or the other.

Windows to the rescue

Some years later, the Windows operating system was released. This time, Microsoft realized that some standards were needed for digital audio and MIDI support right in the operating system itself. Windows has specific functions in it to do various things with audio and MIDI. For example, it has a function to play a WAVE (ie, digitized audio) file on a sound card's Digital-to-audio Converter (ie, DAC). It has a function to send a MIDI message out of a MIDI interface's MIDI OUT jack. Etc. This group of audio functions is known as Windows *Media Control Interface* (ie, MCI). (In fact, Windows MCI also includes functions for video, as well as audio operation of CD-ROM drives. But as MIDI/audio enthusiasts, we don't care about that). So rather than manipulate the audio hardware directly, a Windows program will instead call these audio functions in Windows. For example, if a Windows program wants to send a MIDI message out of a MIDI interface's MIDI OUT, rather than directly manipulating the card's MIDI OUT port like an MS-DOS program would do, the Windows program passes that MIDI message to the Windows function that deals with outputting MIDI messages.

So what does Windows do with that MIDI message? Well, if your sound card or MIDI Interface has a Windows MCI driver for it (and virtually all audio hardware ships with a Windows MCI "driver"), and the card itself has MIDI playback support, then Windows will be able to pass that MIDI message to the card's driver which will actually send the MIDI message out the card's MIDI OUT jack (or maybe play the MIDI message on a built-in MIDI sound module).

NOTE: If you can hear the Windows system sounds upon your audio card, then you definitely already have it setup with a Windows MCI driver (and a card that supports digital audio).

In this way, a Windows program (including Windows game programs) can support any audio hardware that ships with an MCI driver. The Windows program doesn't care what audio hardware design is used; MPU-401, Sound Blaster, or anything else. If the program needs to do MIDI playback, all that is required is audio hardware that actually can do MIDI playback, and its Windows MCI driver. If the program needs to do digital audio playback, all that is required is audio hardware that actually can play digital audio, and its Windows MCI driver. For example, CakeWalk Pro Audio does its MIDI and digital audio playback and recording via Windows MCI functions. Therefore, CakeWalk can use the digital audio and/or MIDI features of either some fancy high-end audio hardware or some cheap "game card". (Actually, CakeWalk is smart enough to use MCI drivers itself, rather than just via Windows MCI functions, and can therefore use more than one card simultaneously for MIDI

play/record. Other Windows programs strictly go through Windows MCI functions, and therefore their access to audio hardware is determined by the settings in Window's "MultiMedia" or "Sound" Control Panel utility, or Win3.1's MIDI Mapper).

So what Microsoft did was essentially make hardware/software incompatibilities a thing of the past. Consumers no longer have to worry about whether a particular software program works with their hardware, and vice versa. They are free to buy any software and use it with any hardware, and be confident that it should work. (This is not to say that all cards are created equal. Sure, a game card can be used to record a digital audio track with Cakewalk, just like a high-end card can. But the game card will typically create a track with a lot more audible "hiss" in it, and maybe roll off more of the high and low frequencies). Hardware manufacturers and software programmers were free to innovate, and since then, we have moved well beyond the MPU-401's MIDI limits and the SB's digital audio limits.

The trade-off

There's always a price to be paid. Sure, now consumers no longer have to ask "Is this hardware compatible with an MPU-401 or Sound Blaster?", nor "Does this software program require MPU-401 or Sound Blaster hardware compatibility?". The hardware incompatibility issues are pretty much irrelevant today, and today's software no longer requires specific audio hardware (although now it may require specific driver support, as I'll discuss below). Things have definitely improved there.

But, we have a new level of software support that we have to worry about, namely, driver support. Whereas it is completely irrelevant today whether the digital audio card you're considering is 100% hardware compatible with an SB, or the MIDI interface you're considering is 100% hardware compatible with an MPU-401, you do have to consider whether the card has good, complete driver support. If it doesn't, you could find yourself stuck with a useless piece of hardware, just like the folks back in the old days were sometimes stuck with useless hardware when it wasn't 100% compatible with an MPU-401 or SB, or they couldn't get software that supported their hardware.

As mentioned, virtually all audio cards made since the late 80's ship with a Windows MCI driver that offers the basic functionality of MIDI play/record and/or digital audio play/record. But over the years, Microsoft has added new features to the Windows operating system which drivers may utilize to give added functionality above the basic features. For example, some cards offer mixing capabilities that any advanced program can tap into (ie, if the program is likewise written for these new Windows features). For example, you may be able to pop-up a graphical "mixer" in Cakewalk to adjust the microphone recording level or speaker output (ie, Master Volume), and mute inputs/outputs, and control other aspects of the card. With older or "low quality" drivers, you may have no such features, or you may have to use some special software that ships with the card to adjust these settings. (ie, You can't fully control such settings on the card via software, and/or from one program like Cakewalk).

Another example is that some drivers are "multi-client" (or "multi-instance") whereas others are "single-client". Multi-client drivers allow more than one simultaneously running program to use the hardware. With a single-client driver, if you're already running one program that is using the audio hardware, and you try to simultaneously run a second program which wants to use that audio hardware, the second program will pop up an error message saying that "The device is busy". You

have to close down the first program in order to run the second program. This can be annoying. (On the other hand, a properly written Windows program should avoid this error even with a single-client driver. Unfortunately, too many audio/MIDI programmers do not know how to write such software. If you encounter a program that gives you this error, send the following URL in an email to the tech support for that product: http://www.borg.com/~jglatt/tech/share.htm).

Microsoft has developed some new functions especially useful for new Windows software. These new functions are called DirectX, and they offer extra flexibility that MCI doesn't such as realtime mixing of many channels of digital audio. In particular, DirectSound is applicable to cards that play digital audio, and DirectMusic is applicable to MIDI. A Windows program has to be specially written to use these new functions, and many new Windows games now use DirectX for video and sound. So, if you have an MCI driver for your card, does that mean that you have DirectSound support, and that a program using such will work with your card? Well, yes and no. To take advantage of DirectX, you need an MCI driver for your card which also has some extra support for DirectSound/DirectMusic in it. If your sound card driver doesn't directly support DirectSound/DirectMusic, Windows can "manipulate" things such that a program using DirectSound/DirectMusic will be able to use your sound card, but there may be speed penalties involved, or some sound features may not work very well. Or the sound may even not work at all if the driver is really questionable. In conclusion, you really do want DirectSound and/or DirectMusic support in your card's driver if you plan to use programs that use DirectX (ie, mostly new Windows games at this point, but audio/MIDI software using DirectSound/DirectMusic may likely be on its way. You may be especially interested in DirectX support if you're using a software synthesizer -- a program running on your computer which emulates a MIDI synth. Software synths tend to work best with programs and drivers that directly support DirectX. And some of the audio programs that use "DirectX plug-ins" to add new features to the program may also benefit from drivers that support DirectX).

But by far the greatest source of problems will be related to Windows XP. Why? Windows XP is a very different operating system than Windows 3.1, Windows 95, Windows 98, and Windows Millennium (ME). Those latter 4 operating systems can all use the same driver. **But a driver written for any of those 4 will not work for Windows XP**. Windows XP is based upon the operating systems of Windows NT and Windows 2000, and it uses a completely different type of Windows driver than does Windows 3.1/95/98/ME. Now it is true that a driver written for Windows NT or Windows 2000 will very likely work for Windows XP. (So, if you can't find an XP driver for your hardware, look for NT or 2000 drivers). But, many sound cards do not ship with a Windows driver that supports NT, 2000, and XP. Those cards ship with a driver that supports only Windows 3.1, 95, 98, and ME. Windows NT/2000/XP drivers are more complicated to write, and a lot of programmers do not know how to write them (whereas 3.1/95/98/ME drivers are "old hat"). So, many music companies have very spotty NT/2000/XP driver support, if any support at all. Before you buy some audio/MIDI hardware to use with XP, make sure that you can get driver support for XP (or at least NT/2000).

For all of the above reasons, I want to stress that it's important to choose a sound card from a manufacturer who is good about providing updated drivers. You may be able to save a few bucks by buying a cheap sound card from "Joe's Sound Card Company", but if you can't get uptodate drivers for it, it may never function as well as a card with good driver support. Years back, I used to have to tell people that they were stuck buying Creative Labs cards due to hardware/software compatibility issues. Now that Windows MCI has made the hardware issues irrelevant, is that no longer the case? Sadly, no.

To be perfectly frank, as perhaps the largest music company making computer audio cards, Creative Labs can afford to hire numerous, decent programmers, and so CL's driver support is typically a lot better than most other companies. And today, driver support is what really matters. A case in point: when I moved to Windows 2000, I couldn't get my Roland RAP-10, or expensive Mediator MIDI interface, or Turtle Beach audio card, or some other multi-port MIDI interfaces working. (These were all considered "better" than Creative Labs cards back when I purchased them). Why? Because they never had Windows NT/2000/XP drivers. Sure, they had drivers for 3.1/95/98/ME, but remember that those drivers will not work for NT/2000/XP. And the manufacturers never wanted to spend the money hiring developers to provide updated support for hardware that had been around a few years and was no longer selling (or selling very much). But what did work on Windows 2000? -- a Sound Blaster 16, because Creative Labs did release updated drivers for it. And so history repeats itself.

Now, a lot of companies like to "promise" future support, but you should take this with a big grain of salt. For example, one company rep I spoke with mentioned that they expected to have NT drivers available for their product within "a couple months". It is now 3 years since that conversation and they still have no such NT drivers (even though their web site still actually claims "Drivers for WinNT are being worked on". I don't think so). My rule of thumb is that, if appropriate drivers are not available within one month of release of the hardware, the odds are unlikely that you're ever going to see the driver support you want. It is unlikely that a competent, "inhouse" programmer would be more than one month behind the actual public release of the hardware, so the company has either hired someone questionable, or they've "farmed out" the work to an outside contractor. In either case, my experience is that such a situation forebodes a questionable commitment to driver updates. So don't hold out on any promise of future driver support longer than one month. (And wait a month before you buy if you're waiting on some "promise").

So, the important questions to ask about driver support for audio hardware is:

1. If you're using Windows NT, 2000, or XP (or plan to soon) -- "Does this card ship with a driver that works under Windows XP?". (If you want to get really technical, you can ask if it ships with a Windows NT/2000/XP kernel mode driver or miniport driver. If the person doesn't know what you're talking about, then you may be talking to someone who can't give you a definitive answer as to whether the card really does have drivers that work under XP). If the answer is no, then the card won't work at all under NT/2000/XP.
2. "Does the driver for this card support the Windows mixer API?" (pronounced "aye-pee-eye", like the individual letters). If the answer is no, then you can't use the standard Windows Volume control (in the task bar) or mixer software with it, and probably can't control such settings from other software. You had better hope that the card at least ships with some special software that lets you do things like adjust the volumes of various inputs/outputs, etc.
3. "Is the driver for this card multi-client?" If the answer is no, then you can run only one program at a time which uses the card.
4. "If this card does digital audio playback/recording, does its driver implement DirectSound functionality? If this card does MIDI playback/recording, does its driver implement DirectMusic functionality?". If the answer is no, then you'll find that the card doesn't work with game software that requires DirectX sound support, or works with very limited sound. And if music software arrives with DirectX support, you may also wish that your driver supported DirectX too.

Different cards for different needs

Now, this is not to say that, just because MCI allows a Windows program to perform MIDI and/or digital audio with most all cards on the market nowadays, you'll get the same performance from all. As mentioned, a game card can be used to record a digital audio track with Cakewalk, just like a high-end card can. But the game card will typically create a track with a lot more audible "hiss" in it, and maybe roll off more of the high and low frequencies. A cheap card may have only a 16-bit DAC (and ADC), rather than 24-bit, and so you may hear more "distortion" as you mix numerous audio tracks in realtime (for example, if you take a program that can record numerous, separate "tracks" of digital audio, such as Cakewalk or Cool Edit Pro, and then play all of them back simultaneously. Also, software synths, and/or DirectX software that does realtime wave mixing, can benefit from 24-bit, rather than 16-bit, DACs. Essentially, any time you're mixing more than 2 digital audio waveforms in realtime, you can benefit from higher resolution than 16-bit DACs. But playing back only stereo, 2 track audio shouldn't make any audible difference between 16-bit and 24-bit hardware).

Furthermore, you should be aware that some cards don't fully support (in hardware) all that is possible under Windows MCI operation. For example, some cards don't offer *full duplex* operation. Full duplex means that when you're recording a digital audio track, you can simultaneously hear previously recorded digital audio tracks being played back. Although you can use CakeWalk with a half duplex card, you won't hear previously recorded digital audio tracks when recording a new track, even though that's what CakeWalk could do with a full duplex card under Windows MCI. (But most all cards nowadays are full duplex). As another example, Digital Audio Labs CardD only supports digital audio playback and recording. It doesn't handle MIDI at all. CakeWalk could use a CardD to play/record digital audio tracks. But CakeWalk can't play/record MIDI tracks on a CardD (ie, you'd have to use a separate card for MIDI, or use a "Software Synth" as described below).

In other words, although MCI allows Windows programs to support a wide variety of cards, not all cards are created equal, and therefore there still are performance differences and limitations that you have to consider.

Finally, it should be noted that not everyone wants the same thing. For example, if you find the sounds and flexibility (ie, controller routing, patch programming, etc.) of the internal sound modules built into computer sound cards too limited, and instead prefer external MIDI modules, then you'll probably want a card without a built-in sound module, and a solid MIDI interface built in. On the other hand, if you play games, you'll very likely want a card with digital audio playback, and may not care about a MIDI interface at all.

In conclusion, you have to know what you want to do with audio/MIDI, and what kind of performance you require, before you can pick the card for you. If you just want a card to use with game software, then it's very likely that the "best" card for you is different than the "best" card for someone who wants to master a CD of his own compositions as played back on his audio card. And if you're a guy who wants to do both, then you may have to decide whether to invest in 2 cards, or find the best single-card compromise. Finally, since driver support is so important to the ultimate performance of a card under modern operating systems, you'll want to buy from a company that offers good, periodic driver updates.

MIDI Interfaces

If you're going to do any MIDI work, you definitely need hardware that supports MIDI input and output. I refer to hardware that deals only with the transfer of MIDI data between a computer and some external MIDI units as a "MIDI Interface". (Yeah, it's sort a misnomer since the last "I" in "MIDI" itself means "Interface"). MIDI interfaces usually have the MIDI IN and OUT DIN jacks right on the hardware itself. Some MIDI Interfaces take the form of a card that plugs into an ISA or PCI slot inside of your computer. Others take the form of an external box that attaches to the computer's serial (COM) or parallel (printer) or USB port. A MIDI Interface does not produce sound by itself. It needs some other, external MIDI sound module connected to it's MIDI IN and OUT jacks.

Because MIDI Interfaces are designed for only MIDI work, sometimes they have special features that appeal to professional musicians who need optimum MIDI performance. For example, some interfaces have multiple MIDI busses to allow MIDI data to be input/output more efficiently to many external devices. (See the article Multiple MIDI outputs for more information about such). Some interfaces offer SMPTE (and other forms of) synchronization. Some interfaces have very fast hardware-buffering of input/output (moreso than cards that aren't dedicated to handling MIDI). Without such buffering, MIDI playback can really hamper a fast computer, and cause it to perhaps slow down when it tries to simultaneously do other things. The absense of such buffering may even result in MIDI playback with weird timing fluctuations (ie, the rhythm of the music sounds "off").

Examples of MIDI Interfaces in ISA card format are Roland's MPU-401 (or MPU-IPC), the SuperMPU, MusicQuest MQX-16 and MQX-32, and certain Voyetra cards. Of these, the MQX-32 and SuperMPU each have 2 MIDI busses, as well as SMPTE sync. These are all older cards that are either out of production or not supported. There have not really been many new MIDI interfaces in the form of an ISA/PCI card. Most MIDI interfaces sold nowadays are external boxes that attach to the serial, parallel, or USB port, with USB being the most popular and best supported now.

Note: The original MusicQuest MQX-32 offered two independent MIDI busses, but the second bus (as well as SMPTE support) was disabled in Uart mode, and that's the mode that most all software runs an MPU-401 compatible in, including Windows software and MPU-401 drivers. MusicQuest also released the MQX-32M (which had 2 independent MIDI inputs) in 1989. A few months later in April 1989, MusicQuest, realizing that most software was running MPU-401 compatibles in Uart mode, redesigned the MQX-32 and MQX-32M to allow the second MIDI bus and SMPTE features to be used in Uart mode. I'm not sure of the firmware version that offered these changes, but the important thing to note is that only the redesigned MQX-32 offers multiple busses and SMPTE under Windows.

Examples of MIDI Interfaces that attach to the parallel port are Midiman's Portman series (comes in 4, 2, or 1 buss) and Mediator's MP-128S (8 MIDI busses, SMPTE). Some serial interfaces are Mediator's MS-124 (multiple busses) or MS-101. Examples of USB interfaces are Edirol's UM-880 (8 MIDI busses, IN and OUT), UM-2 (2 MIDI busses), UM-1S (1 MIDI bus), and MidiMan's MidiSport series (comes in 8, 4, 2, or 1 MIDI buss). Parallel, serial, and especially USB port interfaces are very useful for laptops running Windows. Some of the USB port interfaces are quite nice. They have multiple MIDI outputs (ie, busses) and even SMPTE sync. I'd avoid the serial port interfaces unless you know

that your computer has at least a 16550 chip in it. Less capable chips won't be able to keep up with the MIDI baud rate required by the interface. And forget about serial port interfaces with multiple outputs. There's still a single serial buss (from your computer) feeding the interface. And my experience with parallel port interfaces is that they can sometimes not work with notebooks (although they seem to do fine with most desktop PC parallel ports). Most companies seem to have moved to making USB interfaces, and so the parallel and serial interfaces may not come with driver support for XP (whereas a lot of the USB interfaces do support NT/2000/XP. Edirol seems to have pretty good driver support, as do Midiman). USB is the way to go now, and can work with both PCs and Macs (given driver support).

Some external MIDI modules have a connector to which the module can be directly connected to a computer serial (ie, COM) port. So you don't need a computer MIDI interface at all for such a module. Besides eliminating the need for a MIDI interface, the module typically also has MIDI jacks to which you can attach more external MIDI units which can then pass MIDI data to and from the computer via this module directly attached to the computer. (ie, A MIDI keyboard controller attached to the module's MIDI In sends its MIDI messages along to the computer's COM port for input into the computer. Other sound modules daisy-chained to the module's MIDI out receive MIDI messages going out of the COM port). In other words, such a module functions as a MIDI interface by itself. Examples of such modules are the numerous Roland SC line sold through Edirol, as well as many Roland pro units such as the JV line. These units can even be attached to a Mac serial port (with the proper cable). Furthermore, the units can be attached directly to a keyboard controller, and used without the computer. They even have a headphone jack, making them ideal for playing MIDI scores on notebooks. Of course, you do need a serial MIDI driver for your operating system. (Windows 3.1/95/98/ME drivers, as well as some Windows NT drivers which work on 2000/XP, are available for Roland modules on Edirol's web site). The Roland JV-5030 can be directly attached to a USB port and function as a MIDI interface, as can some of the new Edirol models. USB will likely become the preferred port for MIDI sound modules that attach directly to some computer port.

Digital Audio Cards

If you're going to do any digital audio work, then you need audio hardware with a Digital to Analog Converter (ie, DAC) to play digital audio, and an Analog to Digital Converter (ie, ADC) to record digital audio (ie, digitize an audio signal running into the card's Microphone or Line Input jacks). I refer to such a card as a "Digital Audio Card" or "Audio Card".

The typical (ie, most supported) use of digital audio is for game sound effects and voice (ie, sound effects and voice are usually WAVE files played on the card's DAC), and also recording/playing a finished musical mixdown (such as creating a WAVE or MP3 mixdown of a song). See the Digital Audio on a computer FAQ for further information.

Most digital audio hardware for a computer takes the form of a PCI card (that plugs inside of your computer and has jacks on the back of your computer for speaker output, microphone input, etc). But some digital audio hardware comes in external packages such as Roland's Audio Canvas UA-100 which connects to a USB bus. (Parallel and serial ports are too slow for digital audio transfer in realtime, so products are not available for those ports).

Most all sound cards on the market today offer digital audio playback/record. Examples of cards that offer digital audio playback/record are Event's Darla, Gina, or Layla cards, Digital Audio Labs CardDeluxe or V8, Aardvark Direct Pro 24/96, EMagic Audiowerk8, etc, as well as "game cards" like the SBLive or Audigy, or Santa Cruz, or most anything that you'll see on the shelves of your typical computer store. The more professional cards may offer more than 2 discrete digital audio tracks. (But note that most programs, such as CakeWalk, offer "virtual digital audio tracks" in order to support more than 2 tracks of digital audio on cards with only 2 tracks, so extra digital audio tracks aren't necessary and aren't useful unless supported by software. They prove most useful when using software that can access multiple "devices" simultaneously such as Cakewalk Pro Audio. Then you can use a unique hardware audio "channel" for each sequencer track, and have Cakewalk not need to do any software mixing in realtime. In other words, Cakewalk doesn't need to implement virtual tracks because you have enough actual hardware tracks for all of your sequencer tracks. The hardware does the mixing. And that can help reduce any "artifacts" such as distortion upon playback). The Event, DAL, and Aardvark units are geared for serious semi-pro use (although all of the above non-gamer cards are good digital audio performers), whereas the game cards typically have lower signal-to-noise ratios (ie, produce more "hiss" upon recording) or poorer frequency response (ie, can roll off the lower and higher frequencies more).

There are some high-end ($$$) digital recording packages that feature both hardware and software (typically Windows 3.1/95/98/ME programs). Typically, these systems offer multiple digital audio tracks (usually discrete, with multiple inputs/outputs, digital I/O, SMPTE sync, etc) and features that make them full-fledged production studios comparable to using a standalone digital audio machine such as an ADAT. Such packages are CreamWare's TripleDat, DigiDesign's Session (and earlier Session 8), Spectral's Prisma, Studio Audio & Video's SADIE, Micro Technology Unlimited's Microsound Krystal, and Merging Technology's Pyramix. But for most people, I think that you'd be much better off using off-the-shelf software such as Cakewalk with a good digital audio card and MIDI interface.

Lately, many cards have been offering Digital I/O, meaning that the output of the card's DAC can be run directly to another digital recording medium such as a DAT recorder. In this way, you bypass the card's analog audio output stage (which isn't needed if you're going to create your master mix on a DAT deck anyway), and eliminate any extra noise from that section. Also, transferring digital audio between your computer and other devices that support digital I/O is quicker and results in bypassing the card's analog input stage which can also introduce noise. Most all of the serious audio cards have Digital I/O. Even some of the cheap cards, such as the SBLive or Audigy or Santa Cruz, have digital I/O, but sometimes they don't always support the DAT bit rate as well as the more professional cards do. For example, a cheap card like the Audigy may do something called "resampling" which can introduce distortion. On some cards, the digital I/O is an add-on board that you must buy. Some manufacturers, such as Zefiro, make relatively inexpensive, digital I/O only cards (ie, without the analog audio stage -- ie, you can't actually hear the tracks playing until you hook up some analog stage to them). This is useful if you already have another card that can play digital audio. You use that card just to monitor the audio tracks. But you use the Zefiro to record the tracks, and then play them during the final DAT mixdown. There is also a digital I/O only version of the CardD.

Digital Audio and MIDI support in one card

Most sound cards have both digital audio support, as well as MIDI support. The card may have a built-in hardware MIDI sound module which will play any MIDI data that a program sends to the card. This is akin to attaching one external MIDI sound module to a MIDI Interface card. Such a card actually can produce MIDI generated sound all by itself, unlike a MIDI Interface. So, the one card offers a "complete" audio/MIDI system capable of being used by itself to produce a musical project.

Most all built-in modules nowadays support the General MIDI standard. This means that the module is multi-timbral and can therefore play entire, complex MIDI arrangements. (See the article entitled What's Multitimbral? for more details). Some cards even have built-in hardware effects such as digital reverb, delay, and chorus to enhance the sounds on the card. (Ideally, you want these effects to also be applicable to the digital audio playback, as well as the built-in MIDI sound module. But on some cards, this may not be the case).

One advantage to having such a sound card as opposed to just a MIDI Interface, is that you can easily use the sound card to compose and score music without needing to run MIDI and audio cables to external MIDI sound modules and effects devices, and twiddling knobs on a mixer. You just run the stereo outputs of the card to a tape/DAT deck, and monitor with amp/speakers or headphones (since most cards have a headphone jack too). Many Windows sequencer programs such as CakeWalk offer a software "mixer" that can adjust the panning, volume, reverb level, etc, of each playing "instrument" via MIDI messages that are (hopefully) understood by the card's driver. So, you can setup a finished, stereo "mix" without moving from your computer.

NOTE: Many "game cards" have a poor MIDI implementation that doesn't support MIDI control over certain sound parameters. For example, maybe a better sound card lets you add vibrato to an instrument using MOD Wheel MIDI messages, whereas a cheap card won't do anything with that message.

Some cards also allow you to replace the waveforms used in the GM sound module. You load some WAVE file into RAM on the card itself, set loop points, and can play the wave back polyphonically as an "instrument". In other words, these cards are starting to be "MIDI samplers" (albeit without the bells and whistles of fancy VCAs, VCFs, multi-sampling, etc). Of course, you need a software program in order to choose which waves to load and how to play them back (ie, map them out to particular MIDI note numbers if supported, set looping, etc). Support software for such, which often is released with important features not yet implemented, makes this one of those features that you may want to wait for it to mature. (Such software also tends to be written for Windows 3.1/95/98/ME, but not Windows NT/2000/XP which are much stricter about allowing access to hardware by a program). Many cards offer the ability to install lots of RAM on the card itself which can translate to loading lots of waveforms and creating very rich-sounding patches or unusual custom patchsets which sound totally unlike the original patch set in ROM. Companies are releasing patchsets that are fairly extensive and nice-sounding, which you can further modify, such as Creative Labs' 8 MEG EMU Patchset. Of course, uploading that much data to the card does take a little while (less than a minute) but only has to be done once whenever you turn on your computer. If you do get a card with this feature, you'll want one that supports "downloadable soundfonts" (ie, DLS), preferably version 2.0 of the specification.

Most sound cards with MIDI support also contain a MIDI Interface (for external MIDI modules) too,

because after all, the internal sound module is usually controlled by MIDI messages anyway, and it's not much more circuitry to allow the MIDI messages to be sent on out to external modules. On the other hand, most sound cards are designed more with MultiMedia use in mind rather than MIDI. For this reason, they don't have actual MIDI IN and OUT DIN jacks on the card itself. Instead, the cards have a joystick port into which you connect a special box that contains the actual MIDI IN and OUT DIN jacks. (The box should also contain an optoisolator on the MIDI input). Most companies refer to this box as a "MIDI Adapter Cable" (or sometimes a "MIDI Adapter", or even by the inaccurate designation of "MIDI Cable" which invites confusion with a simple MIDI cable that connects any two MIDI devices together). Since the Sound Blaster was the most common card with this type of "connect the MIDI Adapter to the joystick port" arrangement, most manufacturers have adopted the same pinout (15-pin) as the SB for their joystick ports. In this way, any MIDI Adapter Cable for the SB will work with any other Sound Card.

And although most MIDI Adapters are made to work with any SB compatible joystick port (also called a "gameport"), that doesn't mean that they're all alike. Some adapters have an actual box with surface-mounted female DIN jacks, including even a MIDI THRU jack. You then connect MIDI cords between the Adapter and your MIDI module. The box may even have its own joystick connector so that after you plug the adapter into the card's joystick port, you can then plug a joystick into the adapter's joystick connector. Thus, you don't have to disconnect the adapter every time that you want to use a joystick. (You can use the joystick at the same time that MIDI is going through the card since the joystick uses different pins than the MIDI in/out). An example of such an adapter is Mediator's JAM52/M. Other, cheaper adapters simply chop a MIDI cable in half, solder the bare ends to a connector that plugs into the card's joystick port, and you plug the dangling male DIN connectors directly into your MIDI module's MIDI jacks. Obviously such an arrangement isn't as versatile (nor dependable since cables tend to break) as the box with the surface-mounted jacks. An example of these cheaper adapters is Mediator's MG-2. Many other companies make these MIDI adapters.

There is one thing to know if you're looking for a sound card that offers MIDI support (in addition to digital audio). Many of the new cards don't have a hardware MIDI sound module on the card itself. Rather, the card uses a "Software Synth" for MIDI playback. To cut costs, the card may use the "Microsoft GS Wavetable SW Synth" that is included with the Windows operating system. (The "SW" stands for "Software"). What is a Software Synth? It is part of the card's (software) driver. A MIDI program such as Cakewalk sends the driver a MIDI message to play a C note with a Piano sound, for example, and the driver translates that one MIDI message into a stream of digital audio values which represent the sound of a piano playing a C note. The driver likewise translates other MIDI messages as it receives them. It uses its own set of digital audio waveforms for its built-in "patches" and mathematically mixes them into a stereo, digital audio "mixdown". This digital audio mixdown is played on the card's DAC (ie, digital audio section) immediately (ie, while the driver is doing the translation). In effect, a Software Synth translates MIDI to WAVE, in realtime (ie, while receiving the MIDI messages). This is an inefficient approach toward MIDI playback that is much more CPU intensive for your computer (than use of a real hardware MIDI sound module built into the card). And since the software synth typically needs to load its "soundfont" of waveforms into your computer's system RAM, it eats up your RAM. You'll likely find that, if you need to do any work with MIDI, you'll have to supplement such a sound card with a MIDI interface. One telltale sign that a card's built-in MIDI support is limited to only a Software Synth is if it doesn't support MIDI recording (but rather, only playback).

There are many cards on the market that support both digital audio and MIDI. A few of them are the Creative Labs SBLive (older) or Audigy (newer), Turtle Beach Santa Cruz (sometimes branded as a "Videologic SoundFury"), Acoustic Edge, Lynx Studio Technology LynxONE, the Terratec DMX 6fire LT, etc.

Audio on the Motherboard

Some computers have audio chips directly soldered onto the motherboard, and therefore don't need a separate sound card. In particular, notebook computers usually have motherboard audio because ISA/PCI cards cannot be put into a notebook, and notebooks are cramped for space.

These chips typically offer basic, digital audio recording/playback of the quality of a typical "game card". Besides having a DAC and ADC so that you can record and play digital audio (and play game sound effects and voice), these chips also tend to have at least support for a Software Synth for MIDI playback. A good chip should even support MIDI IN and OUT through a MIDI Adapter attached to a joystick port. But in notebooks without a joystick port, you typically have to get an external MIDI Interface for MIDI support beyond a Software Synth.

I've never seen motherboard audio using a chip that offers much better specs than your typical game card. So, for people looking for higher quality (ie, musicians), motherboard audio is usually to be avoided.

Personally, I don't like motherboard audio because you can't upgrade it easily. If you want to buy and install a new audio card, you can't remove that chip. Therefore you have to setup your new card such that it doesn't conflict with the settings of that onboard chip. Typically, the computer's BIOS has settings to disable the onboard chip, but besides being a waste (ie, at least you can sell an old audio card when you buy a new one, or reuse it in another computer -- not so with these embedded chips), I've discovered that PnP issues and other BIOS bugs can result in problems. Sometimes you can't get that onboard chip to stop using valuable resources (ie, IRQ lines) that you want for your new card, or otherwise can't make it completely "disappear". It lingers like body odor, causing you grief.

I think that an audio card is a better deal than motherboard audio, unless you're absolutely sure that you want a typical game card quality and don't ever want to upgrade for the lifetime of that motherboard. An exception to this is a notebook computer since you can't install a separate ISA or PCI soundcard, and notebooks aren't noted for their upgradability anyway.

What's best for me?

As mentioned, that depends upon what you want to do with audio/MIDI, and what kind of performance you require. Let's examine a few test cases:

I want only a MIDI Interface. I'm going to use only external MIDI sound modules (ie, I like the sound and features of dedicated MIDI modules better than the internal sound modules on Sound Cards). I want to use Windows MIDI programs such as CakeWalk. I'm not going to need digital audio tracks (ie, I'm dealing only with MIDI music or using a dedicated external digital audio unit such as an ADAT). I don't care about game audio support.

A very good choice are the USB port devices with multiple MIDI busses and SMPTE. See the listing of USB interfaces. They offer lots of flexibility and can handle a large, professional MIDI system that needs to sync to other equipment well.

Note that some external MIDI modules also have a connector allowing you to attach the device directly to the computer's serial (COM) port, and more recently, the USB port. Examples are most of Roland's Sound Canvas and JV lines. These external units have MIDI IN and OUT jacks to which you can attach further units. So, in effect, they're like serial port MIDI Interfaces with a built-in GM module. See serial (COM) port modules.

Like the guy above, I'm going to use only external MIDI sound modules with Windows MIDI programs such as CakeWalk. But I want to record digital audio tracks too. I also don't care about game audio support.

If you want top performance, you could buy 2 items; an audio card that is made especially for clean digital audio such as the DAL CardD or V8, Aardvark Direct Pro 24/96, the Event Gina, Layla, or Darla, or other high-end digital audio cards, and a MIDI Interface as mentioned in the preceding answer. Configure your software (and Windows) to use the former for digital audio and the latter for MIDI.

Or, you can buy a sound card that has decent digital audio (although not quite as good as the above digital audio cards) plus at least a built-in MIDI interface for both output and input, such as the Turtle Beach Santa Cruz. This is actually a fairly good all-around card. Even the SB Audigy isn't bad, except for if you want to use its digital I/O with a DAT machine.

Stay away from the no-name sound cards as typically sold in large computer retail chains. They typically use the sort of chip that you'd find used for motherboard audio.

I'm interested in a card only for games, and I want something that's compatible with all games.

If you're a game player, it's mostly all about drivers, specifically DirectX support. You definitely want a card that supports DirectSound, and supports it well. If you want to play it safe, buy a Creative Labs Audigy. If you do, you probably won't be left high and dry without any driver support. Don't buy "Joe's Sound Card" unless there's a web site where you can download driver updates for it, or you're sure it comes with the DirectSound drivers (for your version of the Windows operating system -- XP needs an NT/2000/XP driver) you need in the box. You also want to pay attention to the details of the driver's DirectSound support -- whether it supports "3-D sound modeling" and various effects algorithms, and especially how well it supports realtime mixing of "channels" of audio.

Whether a card uses a Software Synth for MIDI playback is pretty much irrelevant to you, since most games nowadays do not use MIDI at all. (For game music, they use CD audio tracks, which is digital audio. And of course, sound effects and voice are digital audio too).

What may be more important to you is whether the card supports "surround sound" (ie, has 4 independent speaker outputs for "front" and "back" speakers).

I'm interested in a card for games which will work well with all of my game programs. But, I also want to use it for music projects, so it has to have high quality digital audio and MIDI performance, and a great sounding internal GM module. In other words, I want something that sounds better than Creative Labs' line, but is every bit as supported by game programs.

Now you've got a problem. You need a card that is designed for both gamers and musicians, and these are very different markets. You'll likely have to make compromises (ie, maybe give up surround sound for gaming, or accept worse S/N ratios and frequency response for music work).

I have a notebook computer.

Since you can't put ISA/PCI cards into notebooks, a lot of the aforementioned cards are not useful here. As mentioned, a USB port MIDI Interface is one solution for MIDI recording/playback. Of course, you then need an external MIDI module. See the listing of USB interfaces.

External MIDI units that directly attach to a serial (COM) port or USB port are quite useful here, as they eliminate the extra box (and are hence more portable).

If you want to improve upon the digital audio specs of the motherboard audio on your notebook, you may want to look into the Roland Audio Canvas UA-100, or if you want both digital audio and MIDI, check out some of the new USB Edirol models that support both digital audio recording/playback and MIDI.

I need a card that works under Windows XP.

Do not buy anything that doesn't specifically come with a Windows XP driver, or at least has a Windows NT or Windows 2000 driver for it (which should work). Windows 95, 98, SE, ME, or Windows 3.1 drivers will not work with Windows XP.

MIDI Machine Control (MMC) is a protocol specifically designed to remotely control hard disk recording systems, and other machines used for record or playback, over a MIDI cable. The only way to do this is with System Exclusive messages, and so several specific SysEx messages were defined in order to implement MIDI Machine Control. Many devices support this protocol (although a more elaborate protocol was later created called **MIDI Show Control**, which features a command set to control non-musical equipment such as lights and effects devices).

The information below is only a small portion of the complete specification, gleaned from material sent to me by various sources. Some of this information was obtained via trial and error, so some important details may be missing.

Device ID

Every device which can respond to MIDI Machine Control messages should have a unique (ie, individual) ID number. For example, a hard disk recorder may have an ID of 1. A MIDI sequencer controlling the hard disk recorder's record and playback may have an ID of 2. Usually, a device will allow the user to set its individual ID, so that any conflicts between devices can be resolved. The range of allowable ID numbers is 0 to 127 inclusive. By having unique ID numbers, you can use one of these ID numbers in a System Exclusive message, and then the various devices that are all daisy-chained together via MIDI can determine which device a particular System Exclusive message is meant for. Of course, even the master controller which is being used to control the entire MIDI Machine Control network can have its own, unique ID, in case slaved controllers wish to create and send messages back to the master.

NOTE: It is possible to have two (or more) devices set to the same ID number. What this means is that both devices always respond to the same MIDI Machine Control messages with that ID number, and you completely lose individual control over each. There is also no limit as to how many individual ID numbers a given device can respond to. If desired, a device can even respond to all 127 individual ID numbers, but this would be akin to a sound module that only operates in Omni mode (ie, not too useful if you have other units daisy-chained). As you'll see below, the All-Call Controllers ID make it a moot point to have a controller respond to more than one individual ID number.

An ID number of 127 (referred to as the "All-Call ID number") is reserved to mean "all devices respond". When a MIDI Machine Control message with an ID of 127 is sent along the MIDI bus, all devices should respond to this message if appropriate (ie, if they support the particular command of that message), regardless of their individual ID numbers.

The general form of an MMC message

The general form of an MMC message (that is sent to, or generated by, an MMC device) is:

0xF0 0x7F <deviceID> 0x06 <command> 0xF7

The third byte is the Device ID.

The fifth byte is the <command>. It is one of the following values:

0x01 Stop
0x02 Play
0x03 Deferred Play
0x04 Fast Forward
0x05 Rewind
0x06 Record Strobe (Punch In)
0x07 Record Exit (Punch out)
0x09 Pause

The Goto MMC message

The Goto message allows recording or playback to be cued to a specific position in terms of SMPTE time (ie, a specific hour, minute, second, SMPTE frame number, and subframe number). The format of the message is as follows:

0xF0 0x7F <deviceID> 0x06 0x44 0x06 0x01 <hr> <mn> <sc> <fr> <ff> 0xF7

Identity Request

It is possible to query an MMC device to find out it's identity. To do so, you send the device an Indentity Request message as follows:

0xF0 0x7E <channel> 0x06 0x01 0xF7

The reply is device/manufacturer specific. For example, a Fostex D-160 will return the following System Exclusive message:

0xF0, 0x7E, <channel>, 0x06 0x02 <ID> <fc1> <fc2> <fn1> <fn2> <v1> <v2> <v3> <v4> 0xF7

where the above message contains the following parameters:

<ID> - Device's ID
<fc1> <fc2> - Device's family code
<fn1> <fn2> - Device's family number
<v1> <v2> <v3> <v4> - Software Version

Consult the manual for a specific MMC device to see what particular values to use for the above parameters.

Other MMC messages

There are likely more MMC messages that allow other functions. The format of these messages is not documented here.

Programming the MPU-401 in UART mode

NOTE: This document assumes that your program needs to manipulate the MPU-401 hardware directly. Under MS-DOS, you need to do this.

For operating systems such as Windows and OS/2, an application should instead use the operating system functions to do I/O to the MPU-401 (ie, for Windows, see the article Windows MIDI and Digital Audio Programming). In this case, the information contained herein is only useful if you're writing an MPU-401 device driver for these operating systems.

Background

The Roland MPU-401 is a MIDI interface card for the IBM PC. It has 2 modes of operation: **Intelligent** and **Uart** modes.

In Intelligent mode, the MPU uses its onboard circuitry to provide lots of services to an application. For example, the MPU has an onboard hardware timer that it uses to time-stamp incoming MIDI events, and can be used to determine when to output MIDI events for sequenced playback. The MPU-401 also has an internal metronome that can be set to automatically beep at a rate relative to this timer. There are lots of other useful services that Intelligent mode provides such as managing FSK tape and MTC/SMPTE sync (on some models) and data filtering.

Intelligent mode does quite a bit, and therefore requires the app to supply a rather elaborate interrupt handler, since the MPU needs to interrupt the computer for so many various purposes. Furthermore, the MPU is designed to act as the controller. The MPU uses its onboard timer to tell the app when it's time to output the next MIDI message. It doesn't easily let the app use that timer for its own purposes (ie, the app has no direct access to the timer). And when the MPU tells the computer to fed it another byte, the app must do so immediately. The MPU will refuse to do anything else, including even allowing the app to send the MPU a command to do something, until the MPU gets exactly what it asked for. Furthermore, the MPU doesn't have lots of ports for controlling it. It only has 2 bi-directional ports which must serve various purposes, so figuring out what the MPU is doing can be a convoluted process. Finally, unless you're writing a MIDI sequencer, many of the MPU's Intelligent mode functions aren't needed, and in fact, can make things more complicated than need be. Because of all this, Intelligent mode is often not too flexible for an app. It was really designed to help old, slow 8086 computers, by letting the MPU do some of the work of a sequencer. It wasn't designed for flexibility.

UART mode discards all of the above features, and turns the MPU into a simple device that is a slave to the computer. The MPU merely outputs each MIDI byte whenever the app writes that byte to the MPU, and lets the app read incoming MIDI bytes whenever the app chooses to read from the MPU. The app has a lot more control over when it chooses to output and input bytes. Of course, a sequencer app has to manage its own timer for sequencing, and implement a metronome, and do many of the other things that Intelligent mode does. But most modern computers run fast enough that it's no problem for sequencer software to take on all of these duties, and modern OS's like Windows even provide APIs that help manage these duties. Add to the fact that the interrupt handler for UART mode is typically much simpler, and very few other computer cards implement Intelligent mode (so an app that works with other cards has to do all of these duties anyway), are some of the reasons why Intelligent mode is hardly ever used today.

Because of the success of the original MPU-401, many ISA computer cards that offer a built-in MIDI interface,

are designed to be hardware compatible with the MPU-401, but most implement UART mode only. (PCI cards, as well as interfaces that attach to the serial, parallel, or USB ports have no MPU hardware compatibility whatsoever, and the techniques shown below will not work with those cards).

MPU Ports

The MPU-401 is an 8-bit card. It has 2, 8-bit ports: **DATA**, and **STATUS\COMMAND**. The MPU can be set to various base (I/O) addresses. If it is at the default base address of 330 (hex), then the DATA port is at 330, and the STATUS\COMMAND port is at 331.

The DATA port is where you input and output MIDI bytes. This port is bi-directional. That is, you write to this port when you output MIDI bytes to the MPU, and you read from this same port when you input MIDI bytes from the MPU. The MPU also sends its acknowledge byte (for commands) to the DATA port, so that's also where your app reads any command acknowledge from the MPU. When in Intelligent mode, the MPU also sends other non-MIDI bytes to the DATA port for you to read and decipher in order to discover something the MPU wants you to know about, such as when the MPU's timer has overflowed. I'll call these "MPU Operation bytes". It's this intermixing of MIDI and Operation bytes to the DATA port which makes Intelligent mode so convoluted to utilize. Fortunately, in Uart mode, you never read anything but MIDI bytes from the DATA port, with the exception of an ACK if you send a Reset command.

The STATUS\COMMAND port is also bi-directional. When you write to this port, it's the COMMAND port. This is where you write MPU command bytes (ie, not MIDI bytes) to the MPU. The MPU's Intelligent mode has a large command set. For example, the command byte 85 (hex) starts the MPU's metronome beeping (at a rate that is set by other MPU commands). Most all of these commands are ignored and not implemented when the MPU is in Uart mode. In Uart mode, the MPU recognizes only 1 command; the command to put it back into Intelligent mode, the Reset command. So there. After you write a command byte to the COMMAND port, the MPU acknowledges this command. How does the MPU acknowledge? It sends an FE (hex) byte to its DATA port. So, after you write a byte to the COMMAND port, you need to keep reading bytes from the DATA port until you encounter an FE byte. Keep in mind that any other bytes you read before this FE are MIDI bytes (or Operation bytes if the MPU is in Intelligent mode) which you should be prepared to handle while you're waiting for that FE.

When you read from the STATUS\COMMAND port, it's the STATUS port. The 8-bit byte that you read from this port gives you information about the status of the MPU. The highest bit (7) of this byte is the state of the DATA SET READY (DSR) line. This bit will be clear (0) if the MPU has some incoming MIDI byte (or Ack byte or Operation byte) waiting for you to read. The DSR line will remain low until you've read all of the bytes that the MPU has waiting in its hardware input buffer. You should continue reading bytes from the DATA port until the DSR bit of the STATUS port toggles to a 1 (ie, sets). When that happens, the MPU has no more bytes waiting to be read. The next highest bit (6) is the state of the DATA READ READY (DRR) line. This bit will be clear whenever it's OK for you to write a byte to the MPU's DATA or COMMAND ports. The MPU sets bit 6 of the STATUS port whenever it is not ready for you to write a byte to the DATA or COMMAND ports. You should always make sure that DRR is clear before writing each byte to the MPU's DATA or COMMAND ports. (The MPU will not clear DRR if it has an incoming MIDI byte waiting for you to read, ie, whenever DSR is set. Keep this mind if you try to do polled input instead of using an interrupt handler for input. If you don't install an interrupt handler for MPU MIDI input, you must always check DSR in all of your program loops, and be prepared to read the MPU's DATA port whenever DSR is set).

Uart mode

When an MPU powers up, it defaults to Intelligent mode (ie, assuming it supports such. Devices which only implement Uart mode typically power up in Uart mode). So, in order to use Uart mode, you need to put the MPU into Uart mode. You do this by sending the command byte 3F (hex) to the MPU's COMMAND port.

Before you do this, it's a good idea to reset the MPU by sending the command byte FF to the COMMAND port. This is the Reset command. Besides clearing out the MPU's input buffer, and running status, this command actually puts the MPU into Intelligent mode. After you send the MPU a command byte, you need to wait for that acknowledge byte from the MPU (ie, at the DATA port). After you get that acknowledge, you can follow up by writing the 3F command byte to kick the MPU into UART mode. Normally, you'd wait for an acknowledge to each written command byte, but the 3F command is an exception. The MPU does not send an acknowledge for this one command, and instead launches right into Uart mode. You're home free now. All you need to do is install your interrupt handler to read incoming MIDI bytes (assuming you want interrupt-driven input as opposed to doing polled input). With output, you need to do polling. The MPU doesn't interrupt the computer after its done outputting a byte.

Here are some 80x86 functions (for MASM) to help you get started using the MPU.

```
DataPort dw 330h ;Change this to whatever base address your MPU is at
StatPort dw 331h ;Change this 1 greater than whatever base address your
                 ;MPU is at

;************************** is_input() ***************************
; Checks if there is a byte waiting to be read from the MPU. Clears
; the Z flag if so. Sets the Z flag if not.
;
; Uses registers AL DX

is_input proc near
     ;---Return state of DATA SET READY. MPU clears this line when it
     ;   has a byte waiting to be read from its DATA port.
     mov   dx,StatPort
     in    al,dx
     and   al,80h
     ret
is_input endp

;************************** get_mpu_in() ***************************
; Reads a byte from the MPU's DATA port. Returns the byte in AL.
;
; Uses register DX

get_mpu_in proc near
     mov   dx,DataPort
     in    al,dx
     ret
get_mpu_in endp

;************************** is_output() ***************************
; Checks if it's OK to write a byte to the MPU's COMMAND or DATA ports.
; Clears the Z flag if so. Sets the Z flag if not.
```

```
;
; Uses registers AL DX

is_output proc near
     ;---Return state of DATA READ READY. MPU clears this line when it's
     ;   OK for us to write to the MPU's ports.
     mov   dx,StatPort
     in    al,dx
     and   al,40h
     ret
is_output endp

;************************* put_mpu_out() ***************************
; Writes a byte to the MPU's DATA port. The byte is passed in AL.
;
; Uses register DX

put_mpu_out proc near
     mov   dx,DataPort
     out   dx,al
     ret
put_mpu_out endp

;************************** set_uart() *****************************
; Sets the MPU into Uart mode. If an interrupt handler is already
; installed for the MPU, then you should disable that interrupt
; before calling this.
;
; Uses registers AL DX

set_uart proc near
     ;---Wait until it's OK to write to the MPU's ports. Note: if there's
     ;   something wrong with the MPU, we could be locked in this loop
     ;   forever. You really should add a little "escape code" within this
     ;   first loop, or at least a timeout of let's say 1 second.
sr1: call  is_output
     jnz   SHORT sr1
     ;---Send FF command to the MPU.
     mov   al,0FFh
     out   dx,al
     ;---Wait for the MPU to make a byte available for reading from its
     ;   DATA port. You could also lock up here if you're dealing with
     ;   a game card that doesn't even implement a bi-directional
     ;   COMMAND/STATUS port. Therefore, another few seconds time-out is
     ;   appropriate here, and if a time-out occurs, just jump to sr3.
again:
     call  is_input
     jnz   SHORT again
     ;---Get the byte and check for an ACK (FE) to the cmd we sent. If
     ;   not an ACK, discard this and keep looking for that ACK.
     call  get_mpu_in
```

```
     cmp   al,0FEh
     jne   SHORT again
     ;---Wait until it's OK to write to the MPU's ports.
sr3: call  is_output
     jnz   SHORT sr3
     ;---Send 3F command to the MPU
     mov   al,03Fh
     out   dx,al
     ret
set_uart endp
```

First of all, before you install any interrupt handler for MIDI input, put the MPU into Uart mode by calling set_uart(). When this call returns, you can then install an interrupt handler for MPU input. (Consult an MS-DOS book for how to write and install an interrupt handler. By default, the MPU-401 uses IRQ #9, although modern versions have jumpers for setting IRQ). Whenever the MPU receives an incoming MIDI byte, it will interrupt the computer. In your interrupt handler, you should keep reading bytes from DATA port while DSR is low. Here's how you do that:

```
    ;---Is there another byte waiting to be read?
more:
    call  is_input
    jnz   SHORT done
    ;---Yes there is. Read it.
    call  get_mpu_in

    ;OK. We got the byte in AL. Normally, we'd store it somewhere. But,
    ;I'm just going to throw it away because this is a do-nothing tutorial.

    ;---Now go back and see if there's another byte waiting to be read.
    jmp   SHORT more
done:
```

There's just one thing I want to stress. Never, never end your MPU interrupt handler without making sure that you've read all waiting bytes. (ie, Make sure that DSR is clear). If you end your handler leaving DSR set, then the MPU will not cause any further interrupts. If you do lots of other things in your MPU interrupt after you read all bytes, it's best to make one last, further check with is_input() before ending your interrupt handler.

Now whenever you want to output a MIDI byte to the MPU, you need to check if it's OK to do so, and then write it to the DATA port. Here's how you do that:

```
    ;---Is it OK to write the MPU ports?
wait:
    call  is_output
    jnz   SHORT wait ;No it isn't. We have to wait.
    ;---Write the byte. Just for illustration, I'll output a MIDI Clock
    mov   al,0F8h
    call  put_mpu_out
```

Note that in the above example, we wait for the MPU to be ready to accept a MIDI byte. The MPU may be in the process of outputting a previously written MIDI byte. After all, MIDI transmission is really slow. You may

not want to wait for the MPU to finish with that other byte before it lets you write a new byte. Instead, whenever is_output sets the Z flag, you may want to postpone MIDI output, go do something else, and then come back to calling is_output.

Here are the above functions in C.

```
unsigned short DataPort = 0x330; /* Change this to whatever base
                         address your MPU is at */
unsigned short StatPort = 0x331; /* Change this 1 greater than whatever
                         base address your MPU is at */

/************************** is_input() ***************************
 Checks if there is a byte waiting to be read from the MPU. Returns
 0 if so. Returns non-zero if not.
*/

unsigned char is_input(void)
{
    /* Return state of DATA SET READY. MPU clears this line when it
       has a byte waiting to be read from its DATA port. */
    return(inp(StatPort) & 0x80);
}

/************************** get_mpu_in() ***************************
 Reads a byte from the MPU's DATA port. Returns that byte.
*/

unsigned char get_mpu_in(void)
{
    return(inp(DataPort));
}

/************************** is_output() ***************************
 Checks if it's OK to write a byte to the MPU's COMMAND or DATA ports.
 Returns 0 if so. Returns non-zero if not.
*/

unsigned char is_output(void)
{
    /* Return state of DATA READ READY. MPU clears this line when it's
       OK for us to write to the MPU's ports. */
    return(inp(StatPort) & 0x40);
}

/************************** put_mpu_out() ***************************
 Writes the passed byte to the MPU's DATA port.
*/

void put_mpu_out(unsigned char data)
{
     outp(DataPort, data);
}
```

```
/************************** set_uart() ****************************
 Sets the MPU into Uart mode. If an interrupt handler is already
 installed for the MPU, then you should disable that interrupt
 before calling this.
*/

void set_uart(void)
{
     /* Wait until it's OK to write to the MPU's ports. Note: if there's
      something wrong with the MPU, we could be locked in this loop
      forever. You really should add a little "escape code" within this
      first loop, or at least a timeout of 1 second. */
    while(is_output());

     /* Send FF command to the MPU. */
     outp(StatPort, 0xFF);

again:
     do
     {
        /* Wait for the MPU to make a byte available for reading from
           its DATA port.
           Note that you could also lock up here if you're dealing with
           a game card that doesn't even implement a bi-directional
           COMMAND/STATUS port. Therefore, a few second time-out is
           appropriate here, and if a time-out occurs, jump to skipit. */
        while(is_input());

        /* Get the byte and check for an ACK (FE) to the cmd we sent.
           If not an ACK, discard this and keep looking for that ACK. */
     } while (get_mpu_in() != 0xFE);

skipit:
     /* Wait until it's OK to write to the MPU's ports. */
     while(is_output());

     /* Send 3F command to the MPU. */
     outp(StatPort, 0x3F);
}
```

Here's an example of reading input:

```
unsigned char data;

     /* Is there another byte waiting to be read? */
     while(!is_input())
     {
       /* Yes there is. Read it. */
       data = get_mpu_in();

       /* OK. We got the byte. Normally, we'd store it somewhere. But,
```

```
    I'm just going to throw it away because this is a do-nothing
    tutorial. */
}
```

Here's an example of writing a byte:

```
/* Is it OK to write the MPU ports? */
while(is_output());

/* Write the byte. Just for illustration, I'll output a MIDI Clock */
put_mpu_out(0xF8);
```

Errata

As mentioned, most cards implement only UART mode. But some cheap game cards don't even bother implementing a bi-directional COMMAND/STATUS port. They implement only the STATUS port. (The assumption is that, because UART mode doesn't support any commands except the FF command to knock an MPU back into intelligent mode, there's no reason to have a COMMAND port in a UART only card. Instead, the manufacturer saves a few pennies on parts). The implication here is that if you call set_uart(), it will lock up forever in the again loop (ie, while waiting for the ACK). Why? Because the card will never process any command and therefore never ACK to something that the card completely ignores. That's why I recommend adding the timeouts as noted above to set_uart() if you're not sure that you're dealing with a properly designed MPU Uart mode.

Most modern sound cards with built-in wavetable sound modules have their patch sets (and drum kit) configured to be General MIDI compliant. For example, a Roland RAP-10 offers a GM compliant sound module. So too do the SCD-10 and SCD-15 daughterboards. Yamaha's cards offer GM (as well as a larger patch set known as XG).

Game cards typically offer two "modes"; a GM compliant mode (ie, a "Windows Sound System" mode which on really cheap cards may be only partially GM compliant) which incorporates the wavetable section of the card, and an SB compatible mode which incorporates FM synthesis. The Ensoniq and Tropez are examples.

Not all of the above have the same quality sound, but they all are GM compliant sound modules (at least when in their "Windows Sound System" mode). So it's important to understand both the MIDI Specification and General MIDI in order to know how to make music on the wavetable modules of these cards.

You should have a "General MIDI" option for game music. This option means that your game music will take the form of simply writing MIDI messages to an MPU-401 compatible MIDI card (in Uart mode. For details and example code on programming the MPU-401 Uart mode, see the article Programming the MPU-401). It's assumed that a GM module is attached to the MPU's MIDI OUT. This is the case with most sound cards that offer MPU-401 compatibility plus a wavetable module. The built-in wavetable module is internally attached to the MIDI OUT of an MPU compatible port. The one notable exception to this is Creative Labs. (So what else is new?) In order to save a few pennies in parts, Creative Labs did not design the SB32 and AWE32 wavetable modules to be accessed through an MPU compatible MIDI port. The built-in wavetable sound module is accessed via proprietary ports, and not through an MPU-401 port which understands MIDI. So, you still have to incorporate support for the SB's proprietary way of accessing its wavetable module. But the good news is that a General MIDI option supports most everybody's wavetable sound except for Creative Labs. Oddly enough, if you add a daughterboard to an SB, the daughterboard is accessed through an MPU-401 MIDI port, so that does upgrade the SB32 and AWE32 to work just like everyone else's wavetable module (ie, it is accessed in the same way -- by writing MIDI messages to an MPU-401 MIDI OUT port).

It's best to have your game music in MIDI file format. (Of course, for cards whose internal modules are not attached to an MPU compatible port, such as the SB, its "control bytes" may also not be MIDI. So, you may have to do some translation of the MIDI bytes when playing back a musical score on an SB, or for cards that offer only an SB FM synthesis section (ie, no wavetable attached to an MPU port). But, I'd still recommend using MIDI for the musical score's file format. It's easy to deal with when creating/editing game music). For details, see MIDI File Format.

I think that the best test setup for a game programmer would be an AWE32 with either a GM daughterboard such as an SCD-10 or some external GM module attached to the AWE32's MIDI OUT. You can then test playback of sound effects upon the SB's digital audio section, as well as test playback of music upon the SB's FM synth and also the WaveTable Synth (which is GM compliant, but not accessed via the MPU compatible MIDI port). Then, you can test your game music's "GM" option via the daughterboard (or external unit) connected to the AWE32's MPU Uart compatible

MIDI OUT port. This covers testing the SB family, and all cards that offer either MPU-401 compatibility or SB compatibility (which is just about everything else, except maybe for a GUS -- I'm not sure if that one is completely proprietary).