

Audio Compositor



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This version of Audio Compositor is shareware; please consider [registering](#) your copy. Audio Compositor may be freely distributed so long as the entire kit (program, documentation, sample data, and other supporting files) is kept intact and unmodified. Registration codes used to unlock features of the program are assigned to individuals and may not, of course, be redistributed.

Introduction: What Audio Compositor Is

Audio Compositor is software for realizing standard MIDI files directly as digital audio. Scores that are not too demanding can be processed in real time and played on your computer's sound card. Alternatively, the output may be directed to a .WAV file for later playback--and in this mode Audio Compositor can deal comfortably with extremely complex scores and extraordinarily large collections of input samples.

Audio Compositor can also be driven in real time from your computer's MIDI IN port--that is, it can act as a sound module that you can play from your MIDI keyboard. Your sound card's wavetable capability, if it has any, is bypassed. The program's performance in this role will vary depending on processor speed and available physical memory; on less than a Pentium-class system it does not make for a really playable musical instrument, but can still be a useful aid to patch editing.

Audio Compositor's sounds are derived from instrument samples in individual .WAV files. It does not come with a ready-made sample library--there is only a small classical guitar patch to serve as an introduction to the program. Beyond that, you must do your own collecting, or design your own sounds with the assistance of other software. In addition to its own file formats, Audio Compositor reads patch banks in E-mu Systems's SoundFont 2.0 format, many of which are available on the Internet or from commercial sources.

Why Use Audio Compositor?

Music production on general-purpose computers is nothing new (it predates the commercial synthesizer), and a variety of software for the purpose can be found today--perhaps the best known is Csound. The spirit of most such software is to offer synthesis techniques that aren't, and perhaps never will be, implemented in the latest popular MIDI keyboard.

Audio Compositor does not quite belong with this distinguished company--it is simply the software equivalent of that most mundane of electronic music devices: the sample-playback machine. The value of software like this has been debated in various forums, the negative argument being that software wavetable synthesis uses a two thousand dollar computer to do the job of a two hundred dollar sound card, only more slowly.

It's true, as a little experience with its realtime functions will show you, that at today's processor speeds Audio Compositor is not a very capable instrument for live performance. Yet when sending its output to a .WAV file, it can do several things that a ordinary MIDI device cannot.

For one thing, Audio Compositor can deal comfortably with sample collections occupying hundreds of megabytes, using your hard disk as virtual memory and automatically shuttling samples between disk and RAM as needed. (Indeed it does this even when played from the keyboard--that is why you notice a slight delay the first time you play a given note.) Sample memory layouts that would be prohibitively expensive in hardware thus become quite practical. Of course, like any sample-based instrument, Audio Compositor is happiest with lots of physical memory as well, but the memory it uses is your computer's regular RAM--not memory installed on your sound card where it can be used for only one purpose.

Audio Compositor also works well with extremely dense or complex scores. When its output is a WAV file, it can produce practically an unlimited number of simultaneous voices. There is no serial interface at which MIDI data may be bottlenecked, and the timing of events in the output file is accurate to the nearest sample frame. Most of the small MIDI studio's usual sources of noise and distortion are eliminated--particularly if you have the means to move the output .WAV files to DAT or recordable CD for playback.

Having said all this, there is no use pretending that Audio Compositor is a good general-purpose substitute for a wavetable soundcard, especially if your goal is to hear General MIDI files at what might be termed the standard level of quality. It is designed as a tool for musicians who're trying to stretch the limits of MIDI and traditional sampling technology, by removing memory and polyphony barriers in exchange for speed of processing.

Midi Implementation

Audio Composer's internal architecture is quite simple. Samples (individual .WAV files) are mapped to the MIDI keyboard to form basic voices called *layers*. A *patch* in turn may contain any number of layers, which may sound simultaneously or be separated by velocity-based crossfades. Audio Composer provides a graphical environment for designing the keyboard mappings, envelopes, LFOs, controller routings, and so on.

Audio Composer's MIDI implementation includes the following:

Velocity, pitch bend, pan, volume, the sustain pedal, and tempo indications are recognized in the usual way.

Velocity, channel aftertouch, MIDI note number, pitch wheel, and all MIDI controllers are routable to a list of destinations which includes amplitude, pitch, filter cutoff, envelope parameters, LFO depths, etc.

Program changes are recognized for 128 patches, with no bank select. To use more than 128 patches you would need to process a score in multiple passes. (The new patch file format for Audio Composer version 2.0 is designed to accommodate multiple banks in a future release.)

Your Sample Library

Audio Compositor's raw material is your sample library, a collection of samples in Windows WAV format that you must somehow acquire. There is really no special physical "library"--the samples may be scattered around your hard disk or organized nicely into subdirectories (the latter approach is recommended!) Each sample that you intend to use in Audio Compositor must first be entered into a master list in the layer editor. If a sample must be looped, or if it will ever need to be pitch-shifted (this probably includes most of your samples), then you will need to add loop and pitch information to the WAV file using Audio Compositor's waveform editor, or any audio editor that understands this type of additional data.

Samples may have been recorded at any sampling rate (though it's generally best if they match the rate you will select for your output file). They can be mono or stereo; stereo separation will be maintained if you select stereo WAV output. The sample word size must be 16 bits--Audio Compositor does not know how to load 8 bit samples.

There are many sources of instrument samples these days. You can buy them on CD-ROMs designed for various samplers, though extracting these sounds to WAV files may tax your ingenuity. If your CD-ROM drive will read raw audio, you may find it more convenient (and more economical) to purchase samples on audio CDs and extract the contents using one of the audio grabbers available on the network. Less exotically, you can of course connect your audio CD player to your sound card and re-digitize: if your sound card is in the Turtle Beach class or better you may be surprised at how little quality you sacrifice compared to a direct digital transfer. As of this writing, CD-ROMs with samples already in .WAV format are beginning to appear on the market.

As an alternative to collecting and editing your own samples, you can also use prepared patch banks in E-mu Systems' SoundFont 2.0 format. Such banks are increasingly available on the Internet and from commercial sources. Audio Compositor is not an SF2 editor, but it can read these banks and use them for all of its realtime and MIDI file rendering functions.

I would like nothing better than to distribute Audio Compositor together with a full set of instrument samples, but the samples I use myself are from commercial sources and are not mine to give away. What's needed is a set that is (1) complete enough to be musically useful, (2) of very high quality, and (3) in the public domain. It's easy to satisfy any two of these conditions; please contact me if you can suggest how to manage all three.

Direct To Digital

Realizing a MIDI sequence directly as digital audio has certain advantages: for example, no electronic sources of noise or distortion are present, and there is no "noise floor" whatever. But before rejoicing over the absence of cables, mixer, and effects boxes, consider what you will have to do to replace their functionality. Mixing in the basic sense must be accomplished in the context of the MIDI file itself, using velocity, volume controller, etc. Effects can only be supplied after the fact, by loading Audio Composer's output into your favorite audio editor and applying whatever capabilities it may have. (Reverb will probably be at the top of your list.)

System Requirements

Audio Compositor is a demanding application, and it is designed primarily for what not so long ago were considered high-end systems. It runs on Intel or compatible platforms and requires Windows 95, or Windows NT beginning with version 3.51. Testing on NT has not been thorough.

Audio Compositor requires a floating-point unit--or at any rate its performance without one will be ridiculously poor--so a 486DX must be considered the minimum processor. Realtime output is awkward below about a Pentium 90, though success here is also heavily influenced by your sound card and its driver.

The memory requirement is harder to generalize about. Ideally, of course, you would want enough physical memory for all of the samples Audio Compositor will be using at a given time. On the other hand, if you had this much memory lying about, you might be better advised to install it in your sampler, and not use Audio Compositor at all. In other words, while paging memory to disk is normally to be avoided, it is precisely this capability that distinguishes your computer from your sampler, and Audio Compositor is designed to capitalize on it. Running it on eight megs of RAM may be painful, but trading speed and convenience for sample capacity is what Audio Compositor is all about. Sixteen is much more practical. There is no upper limit on how much memory the program can use, if you are building patches large enough to take advantage of it.

A sound card is a practical necessity, though the program will run without it. Wavetable capabilities are not necessary.

Packing List and Installation

Audio Compositor consists of a single executable called **Ac.exe**. The distribution also includes starter layer and patch files, and a small set of classical guitar samples, whose chief purpose is to give you something to look at when you first start up the program. There is also a small MIDI file with which you can do a test run.

There is nothing here to warrant a fancy installation program--just unzip the distribution, keeping the directory structure intact, and you will have something like:

```
c:\Ac\Ac.exe
c:\Ac\Ac.lay
c:\Ac\Ac.pat
c:\Ac\Leyenda.mid
c:\Ac\Samples\Guitar\Guitar p 2G
c:\Ac\Samples\Guitar\Guitar p 3C
(more for a total of 12 guitar samples . . .)
```

There is no reason not to use a different drive letter or directory name, but if you want the guitar patch to work you must keep the guitar samples in the same path relative to the program itself. That is, if Audio Compositor is installed in

```
h:\Dodgy-looking-software\Ac
```

then it will look for the guitar samples initially in

```
h:\Dodgy-looking-software\Ac\Samples\Guitar
```

If you find other samples in the kit, they'll also be installed in subdirectories under \Samples. When you add samples of your own, however, they may be installed anywhere on your computer (even on a different disk from Audio Compositor itself).

About the Example Files

Audio Compositor comes with a set of classical guitar samples, and the initial patch and layer files are preconfigured with a guitar patch that uses these samples. There is also a MIDI file containing an abbreviated version of Isaac Albeniz's *Leyenda*, which is designed to work with the guitar patch.

Including these example files was a tough decision. Since Audio Compositor is distributed over the network, the guitar sounds had to be compact, and they were not recorded with very good equipment in the first place. The MIDI file is quite simple. All of this is completely contrary to the main purpose of Audio Compositor, which is to allow you to put a very large array of high quality samples to use.

On the other hand, the guitar examples let you see something besides an empty window when you bring up Audio Compositor for the first time, and a few moments spent looking at them can probably tell you more about the program than an hour of reading this manual. Running *Leyenda* can also give you a quick idea of how well Audio Compositor will perform on your system. The guitar samples are also well suited to realtime control from your MIDI keyboard, since they fit into about 300 KB of memory and load quickly from disk.

The **Readme** file contains instructions for running the *Leyenda* example straight out of the box.

Registering Audio Compositor

Audio Compositor is shareware; if you like it please register it! Before registering, however, you should familiarize yourself with the program and understand how well it will run on your computer. For details see the [Registration Form](#).

Registration activates certain features that are unavailable in the shareware version. In this version of Audio Compositor the differences are as follows:

Feature	Unregistered	Registered
Maximum sample rate (WAV output)	22050	48000
Maximum realtime polyphony	6	32
Stereo WAV output	No	Yes
LFOs	No	Yes
Auxiliary envelope	No	Yes
Variable filters	No	Yes

Note that the increased realtime polyphony will not benefit all systems; see [Realtime MIDI Performance](#). Audio Compositor's "offline" polyphony (when rendering a MIDI file to WAV output) is unlimited even when unregistered. Keyboard-driven realtime output is always monophonic (this restriction will be lifted in a future version).

Audio Compositor Registration Form

When you register your copy of Audio Compositor, you will receive a personal registration code which adds functionality to the program. The registration code is also valid in updates and future versions of Audio Compositor, the latest of which may be downloaded from the Audio Compositor web site:

<http://www.eden.com/~mitchell>

Audio Compositor was first released in February 1996; the registration terms given here are good through May 1997. After that date, please check the web site for the latest news.

Registration costs \$40 US (US check or money order, or international money order drawn in US dollars), payable to Scott Mitchell. Mail with this form to:

Scott Mitchell
6350 N. New Braunfels Ave.
Suite 104
San Antonio, TX 78209
USA

Registration codes are delivered by postal mail. Please allow a reasonable amount of time for delivery, but if you have questions by all means get in touch via e-mail (mitchell@eden.com).

Thank you for registering!

Name: _____

Address: _____

Country: _____

E-mail: _____ (optional)

Optional:

Your O/S: Win95 WinNT

CPU Type: _____ Memory: _____

Sound Card: _____

Where did you find Audio Compositor?

Overview of the Program

If you drop down Audio Composer's **File** menu you'll see that it can open five types of files. The first three types are files that you can edit in various ways:

Sample files, which are standard Windows .WAV files containing individual instrument samples. You will need to add loop and pitch information to most samples using Audio Composer's wave editor (or a compatible editor). Some samples, such as unpitched percussion sounds, may need no preparation at all if they are already in 16-bit .WAV format.

The one and only **layer** file (AC.LAY), which has two functions. First, it keeps a master list of the samples available on your hard disk(s). Second, it maps groups of samples to notes on the MIDI keyboard to form intermediate objects called *layers*. For example, if you have a set of eight fortissimo trumpet samples at various pitches, you might create a layer called **Trumpet ff** in which the eight samples are mapped across the trumpet's range.

Patch files, with a .PAT extension. Each patch file can describe a set of up to 128 numbered patches. A patch corresponds to a MIDI program change, and is composed of any number of layers. Patches also contain most of the interesting synthesizer-like parameters that control musical expression. For example, if you have three trumpet layers called **Trumpet p**, **Trumpet mf**, and **Trumpet ff**, you might add them to a patch called **Trumpet** and specify velocity crossfades between them. Each layer could also have its own amplitude envelope, controller routings, and so on.

MIDI files, which must be in type 1 (multi-track) format. Audio Composer doesn't edit MIDI files; rather, you open a MIDI file when you are ready to render it. The rendering process creates a digital audio rendering of the MIDI file using the samples, layers, and patches created in the previous steps. Files that are not too complex can also be played through your sound card as they are rendered.

SoundFont files, which are self-contained patch banks in a format designed by E-mu Systems. A SoundFont can take the place of the whole combination of Audio Composer's patch, layer, and wave files; and it can be used for both MIDI file rendering and live MIDI. However, Audio Composer does not include an editor for SoundFonts.

Another component of Audio Composer is the Realtime Engine, represented by the bar at the bottom of the main window. When the realtime engine is started, Audio Composer becomes a MIDI module that you can play live from an external keyboard (or in a pinch, by clicking a special on-screen keyboard with your mouse). Its sound is derived from whichever wave, layer, patch, or SoundFont is currently active on the screen. The number of simultaneous voices you can play depends on the speed of your computer and your sound card; it's usually necessary to spend some time adjusting the settings to get clean output in this mode.

Preferences

The Preferences dialog (**Edit | Preferences**) is used to set some global parameters for Audio Compositor:

Audio Output Device

If you have more than one sound card in your system, you can choose from this list which one Audio Compositor will use. Another option is "Auto-select," which allows the system to find a device that can handle whatever sample rate is called for at the moment. If you only have one audio card, you don't need to worry about this setting.

Always Force Integer Loops

Audio Compositor's wave editor has the ability to create loops with fractional lengths. While this feature can be extremely useful at times, it can also get in your way--for example if you are editing a file for use with other software that requires whole-number loop lengths. You may also be bothered by an admittedly confusing inconsistency in Audio Compositor itself: that fractional loops are respected by the realtime engine and the MIDI-to-WAV renderer, but not by the wave editor when you press the **Play With Loop** button.

The wave editor allows you force integer-length loops for the current editing session by checking an item on its **View** menu. To make this behavior the default for all editing sessions, check **Always Force Integer Loops**.

Minimum Loop on Playback

This affects playback in the wave editor only. When you press the **Play With Loop** button, Audio Compositor has a choice between looping the sound itself, and asking your sound card to perform the loop. In many cases your sound card can do a better job of this, but using "driver" looping exclusively would defeat the wave editor's ability to let you hear the loop change as you drag the wave on screen. The compromise is to let the soundcard loop, but interrupt it at intervals with a new loop buffer. This setting defines the interval.

If you find that loops don't play smoothly in the WAV editor, try increasing this setting. (On the other hand, a few soundcards may not be able to do their own looping; if playing a looped sound hangs the program, try a setting of zero.) Some cards are happy with a value of 300; others require settings up to 6000 or above. A lower setting makes the WAV editor more responsive when you are dragging loop points.

This parameter has no effect on realtime playback.

Maximum Sample Cache

This parameter limits the amount of memory that Audio Compositor will use for caching recently-used samples. It affects both realtime MIDI and MIDI file rendering. The out-of-the-box default is 8,000,000 bytes, which is adequate for small projects. If you have sufficient pagefile space, you may increase it drastically (100 megs would not be at all unreasonable, though it would be unnecessary if the total size of your sample collection is not that large).

If you set this value higher than the amount of available virtual memory (free physical

memory plus available swapfile space) then you may get out-of-memory errors while running Audio Compositor. On Windows 95, you can avoid this by keeping this setting well below the amount of free space on the hard drive where your pagefile is located. For a more complete discussion, see [Memory Management](#).

The current size of the cache is indicated on the toolbar at the bottom of Audio Compositor's main window, along with the current pagefile status. The figure indicated here can be deceptive because the Layer Editor has its own cache, which is not included in the display. Wave Editor windows also use memory for samples which is not part of the cache.

The Wave Editor

The Wave Editor is used to prepare individual samples for use by Audio Compositor. While it resembles a general-purpose audio editor in some ways, it is specialized for looping sounds and defining their pitches--two operations which are particularly important to Audio Compositor. The loop and pitch information is imbedded in the .WAV file when you save it.

Audio Compositor's Wave Editor is intended for editing individual instrument samples, and doesn't perform well if asked to load a .WAV file that's larger than a few megs.

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Playing the Wave File



Play, Play With Loop, Stop

These transport-control buttons are used to play the sample through your sound card. Normally the entire file will be played, but if you have set start and/or stop markers (described shortly) then these will be observed. Also, if you have selected a region in the upper view (by dragging the mouse in it) then you will hear only that region when you press Play.

If a loop has been set, **Play With Loop** causes the sound to play continuously until you hit the **Stop** button. You can edit the loop points while this is going on. If no loop points have been set, then **Play With Loop** acts just like the regular **Play** button.

Important: When you press **Play With Loop**, the loop is performed by your sound card's driver. If you have defined a fractional loop length, it will be rounded off to the nearest sample. To hear an accurate rendition of a fractional loop you must use the [Realtime Engine](#).

Navigating the Wave File

When you open a .WAV file in the Object Editor, you will see the contents displayed in two windows, one above the other. The lower window always displays the entire length of the file; the upper window is the main editing view and can zoom in on as small a region as you like.

By dragging the mouse in the lower, full-length view, you select the region of the file that will be displayed in the upper view. This is the quickest way to close in on a particular portion of the file, and takes the place of the traditional scroll bar.

Dragging the mouse in the upper view will also highlight a region, but this has a different purpose than selecting in the lower view--it is usually done to define a region as the target of some editing operation. However, you can drag in the upper view with the *right* mouse button to slide its contents from side to side. This is a convenient way to make small adjustments to the view.

Stereo sounds are displayed differently than in most audio editors. Instead of placing the left and right channels in separate windows, Audio Composer displays them together on the same axis, in different colors. The left channel is always shown in black. (Sometimes, the right channel may not even be visible except at higher magnifications.)

Rescaling the View



Horizontal Zoom, Unzoom

The magnifying-glass buttons increase and decrease the horizontal scale of the upper window, generally by a factor of two. An exception is that if you select a portion of the wave in the upper view and then hit the "magnify" button, the selected portion will expand to exactly fill the upper view.

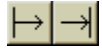
It's possible to zoom in so closely that spaces opens up on the screen between individual samples. When this happens, gray tick-marks appear along the bottom edge of the view to show you where individual samples are. The image of the wave is then drawn in a connect-the-dots fashion, which is mathematically dubious but convenient when you are doing precise looping.



Vertical Zoom, Unzoom

These buttons increase and decrease the vertical scale, allowing you to view waves with small amplitudes. The scale changes by a factor of two with each click.

Setting Start and Stop Markers



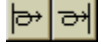
Set Start, Set Stop

When Audio Compositor uses a sample in its output, it normally assumes that it can use the entire length of the file. You can change this behavior by setting start and/or stop markers. Audio Compositor will ignore material that precedes the start marker or follows the stop marker. The Wave Editor will also respect the markers when you listen to the sample.

To set a start or stop marker, click the **Set Start** or **Set Stop** button (the mouse pointer changes to remind you what you're doing) and then click in the *upper* wave view at the appropriate spot. When the markers are set, Audio Compositor puts a gray background behind the portion of the wave that won't be played. The editor will annoy you with error messages if you attempt to do something irrational like placing the stop before the start, or placing either marker inside the sample's loop.

It doesn't make sense to have a lot of material outside the start and stop boundaries--it will be occupying memory at run time even though Audio Compositor will never use it. However, there may be times when you are uncertain about how much of an instrument's attack or decay you wish to include. Moving the start and stop markers gives you a chance to experiment without actually deleting material from the sample. When you have settled on the ideal start and stop points for a given sample, you can use **Truncate to Start/Stop Markers** (on the **Edit** menu) to discard the surplus material.

Setting Loop Points



Set Loop Start, Set Loop End

To introduce a loop into a new sample you must use these buttons, which work just like the **Set Start** and **Set Stop** buttons (see the previous page). The loop start and loop end points will be shown as green and red lines, respectively, in both the upper and lower views.

Use these buttons as a crude way of placing the loop approximately where you want it. You will then want to refine the loop using the Loop View described next.

Switching to Loop View



Loop View

Once loop points have been set, this button toggles between the normal display and a special **Loop View**. In Loop View mode, the upper window is divided in half by a vertical line. The loop start point is drawn first at the center line, and the display proceeds from there to the right-hand edge of the screen. Then it wraps around from the left in such a way that the loop end point falls again at the center line. The effect is to show you what the loop looks like as you roll across the critical junction between the loop points.

To change the loop points, simply drag the wave from side to side with your mouse. Dragging the right half of the screen changes the loop start, and dragging the left half changes the loop end. You can watch the loop markers in the lower view to see how the loop is moving with respect to the whole file.

It's important to note that the horizontal **Zoom** and **Unzoom** buttons still work in Loop View. If you need to make very small adjustments to the loop points, you will want to zoom in closely so that you can slide them back and forth in fine increments. Conversely, you can back off for a wider view when you want to make large adjustments quickly.

When making short loops, it frequently happens that you find the ideal length for a loop before you find its ideal placement. In this case you would like to be able to change the loop start and stop points in tandem, keeping the loop length constant while trying out different positions. Dragging in the upper view with the *right* mouse button provides just this feature by locking the left and right panes together.

See also:

[About Fractional Loops](#)
[More Notes On Looping](#)

About Fractional Loops

When you create a loop with Audio Compositor, the start point is always an integer, but the end point can include a fraction--that is, it can fall between two samples. This can be useful for tuning short, single-cycle loops, or for avoiding stubborn artifacts in longer ones. Creating a fractional loop is simple: put the display in Loop View, zoom in until there is space between individual samples (hash marks appear along the bottom of the window), and move the loop end point as usual by dragging the left-hand pane with the left mouse button. The end point will stay where you put it, even if it falls between two of the hash marks.

In contrast, you will notice that when you drag the right-hand pane to change the start point, the display always jumps exactly from one sample to the next. That is because the loop start point must always be an integer.

Important: To hear the effect of a fractional loop in Audio Compositor's wave editor, you must use realtime playback (see About the Realtime Engine). The wave editor's **Play With Loop** button sends output directly to your sound card, rounding the loop length off to the nearest sample.

Fractional loops are always respected when you play Audio Compositor from an external keyboard, and when rendering a MIDI file.

To prevent fractional looping for a particular sample, pull down the wave editor's **View** menu and check **Force integer loops**. If you would like this behavior to be the default for all wave editing sessions, check **Always force integer loops** in the Preferences dialog.

More Notes On Looping

Whatever you may have learned elsewhere about looping instrument samples will probably translate pretty well to Audio Compositor. If you can find very long samples, the importance of the whole subject may be diminished, since Audio Compositor may not have to resort to looping as often as a regular sampler with limited memory.

Audio Compositor's MIDI-to-WAV rendering process has another technique for reduced looping. A normal MIDI instrument doesn't know when you are going to take your finger off of a key, so once it begins looping a sound, it must continue to do so until the key is released. Audio Compositor, however, looks ahead in your MIDI file, and knows in advance how long each note is held. As a result, it can stop looping whenever it sees that there is enough material in the sample to finish the note without further looping. (Actually it is not quite omniscient and allows a safety margin against the possibility of an upward pitch bend, but that is a fine point.)

While the less-frequent looping is a good thing in general, it can lead to an occasional unpleasant surprise if you are accustomed to thinking that material following a loop end point will never be heard.


Setting the Pitch

Audio Composer will need to know the pitch of most samples. The pitch is specified as a musical note (C4 is middle C) plus or minus a fine-tuning amount expressed as so many cents. The cents value can range from -50 to +50 (a value of +51 would be expressed as 49 cents down from the next higher half-step). The pitch is displayed in the status bar at the bottom of the editing window, both in the musical notation just described and as a frequency in Hz.

It's not necessary to set a pitch value for unpitched percussion instruments or sound effects that are meant to be heard always at their original pitch. (You'll set the "Fixed Pitch" flag for these samples in the Layer Editor.)

Setting the pitch does not modify the sample's audio data in any way. You are telling Audio Composer what the pitch *is*, not what it should be. In other words, if the sample is nominally an A4 but was played 5 cents flat, the pitch should be set to A4 *minus* 5 cents so that Audio Composer can compensate when the sample is performed. Changing the pitch setting will have no effect on the sound you hear when the sample is played directly by the Wave Editor.

The **Edit** menu's Guess Pitch function is the simplest way to set a sample's pitch initially. If further adjustments are needed they can be made using the **Pitch Dialog**, which is called

up by the button shown here:  (that's supposed to be a tuning fork). The pitch dialog has three sliders which enable you to specify the sample's pitch in terms of octave number, note, and cents as described above.

The pitch dialog will remain visible until you close it, or until the current sample loses the input focus.

Compatibility with Other Audio Editors

Audio Compositor uses standard Windows .WAV files. Samples that have neither loop points nor definite pitch (most drums, for example) can be created using nearly any audio editor. Audio Compositor expects 16-bit samples at any sampling rate.

When you add pitch and/or loop points to a sample, they are stored in the .WAV file along with the audio data. The format of this information is again standard; programs from other manufacturers that are specifically designed for editing instrument samples will be compatible with Audio Compositor, though some of them may round fractional loop lengths to integer values. (If this frequently presents a problem, check **Always force integer loops** in the **Preferences** dialog to make Audio Compositor's wave editor consistent with your other software.)

Unfortunately the sampler data format (specifically I mean the "smpl" RIFF chunk) is not recognized by most general-purpose audio editors. Even the "looping" capability frequently seen is usually designed for playlist-style use and is saved in a format that's entirely different from that of sampler-oriented loops. Since these otherwise admirable products often discard information that they don't understand, you may find your loop and pitch information missing after using one of them to modify a sample. (Audio Compositor, by the way, makes a reasonable effort *not* to discard unfamiliar data placed by other editors.) Since software features are constantly changing, the best way to sort all this out is to experiment with the particular editors you have on hand.

The Edit Menu

The Wave Editor's **Edit** menu provides the following functions:

Gain Adjust

Normalize

Normalize L/R independently

Clear start/stop points

Clear loop points

Clear pitch

Truncate to start/stop points

Swap channels

Remix stereo

Reduce to mono

Crossfade loop

Resample

Change playback rate

Synthesize

Guess Pitch

Gain Adjust

Calls up the Gain Adjust dialog to increase or decrease the amplitude of all or part of the sample. This function will also do fade-ins and fade-outs.

Adjust Gain

This function adjusts the amplitude of the currently selected region, or of the entire sample if no region is selected. The adjustment is specified in decibels.

If you enter different values for the starting and ending adjustments, then the amplitude is changed by a value that changes smoothly across the affected region.

For stereo samples, you can specify different values for the left and right channels. The sliders are ganged for convenience--if you need to set them to different values you must work from right to left.

Normalize

Increases the amplitude of the entire sample by the greatest amount possible without clipping.

Normalize L/R independently

Increases the amplitude of both channels of a stereo sample, each by the greatest amount possible without clipping.

Clear start/stop markers

Removes the start and stop markers from the sample if they are present. (More accurately, it sets the start marker to zero, and the stop marker to the end of the sample.)

Clear loop points

Removes loop points from the sample if they are present.

Clear pitch

Removes the sample's designated pitch. An unpitched sample always sounds at its actual pitch, rather than being transposed according to the note that triggers it.

Truncate to start/stop points

Deletes any material in the sample file which precedes the start marker or follows the stop marker.

Swap channels

Quickly reverses the left and right channels of a stereo sample.

Remix stereo

Calls up the Remix stereo dialog, which allows you to adjust the relative levels of the left and right channels, or to reduce the stereo separation between channels.

Remix stereo

This function can be used to rebalance stereo samples or to reduce stereo separation. It replaces each channel with some mixture of the two original channels.

The two sliders on the left determine how the new left channel will be created. Setting this pair to 0 and -96 dB, respectively, will leave the left channel unchanged; raising the second value (and reducing the first as necessary) will introduce some right-channel content into the left channel. Applying this operation symmetrically to both channels at once (e.g., setting the sliders to -2, -12, -12, -2) has the effect of narrowing the stereo image of the sample.

This is probably not an operation you will want to perform very often, but it's proved useful in rare cases when a particular stereo sample sounds unusually "wide" and stands out from its neighbors.

Reduce to mono

Calls up the Reduce to mono dialog, which allows you to mix a stereo sample down to mono.

Reduce to mono

This function changes a stereo sample to mono, mixing the two channels together in the proportions you specify. To save only the left channel, set the sliders to 0 and -96 db; for an equal mixture of left and right, set both sliders to -6 dB (or a bit higher if the two channels contain dissimilar material.)

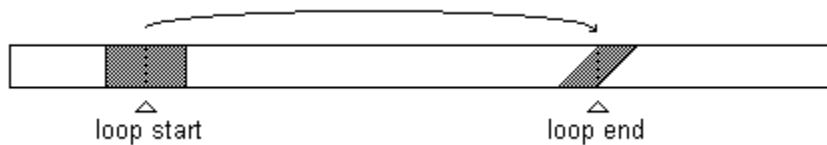
Crossfade loop

Calls up the Crossfade loop dialog, which allows you to perform a destructive crossfade around the current loop points.

Crossfade Loop

Audio Compositor does not perform crossfade looping on the fly. This function performs a "hardcoded" crossfade, permanently modifying your audio data. Obviously this is not an ideal solution, but as some samples simply cannot be looped without crossfading, this option is provided as a last resort.

The portion of the sample that is modified is centered around the loop end point:



The region surrounding the loop start point is used as source material. It is mixed into the region surrounding the loop end point, beginning at a low level, rising to 100 per cent at the loop end point, and then subsiding again to zero. The effect is that of a traditional crossfade loop when the sample is heard in loop mode. When the sample is played straight through, there are really two crossfades in quick succession as the transplanted material emerges and then recedes. Depending on the nature of the sound, this effect can be anywhere from imperceptible to totally unnatural.

Two parameters must be specified:

Radius

The duration (in thousandths of a second) of the fade-in leading up to the loop end point. The total length of the effect is twice this value (or possibly less, since the second fade may be accelerated if there is not enough material after loop end to accommodate it at full length).

Contour

Crossfading a loop often creates a glitch in apparent sound volume around the point of the crossfade. If the fade is done linearly, then a relatively long crossfade on a complex sound like a string section can exhibit a dip in amplitude. An equal-power fade will correct this, but conversely may create a "bulge" when applied to other types of material. Set "Contour" to zero for a linear fade, to 100 for an equal-power fade, or to some value in between.

Since the Wave Editor has no "undo" function, be sure the original sample is saved to disk before experimenting with crossfade looping. Try a variety of settings to see which values work best, using the **File** menu's **Reopen** function each time to discard the changes. Be sure to listen to the result both in loop and straight-through mode. Short fades (a few milliseconds) are often useful for loops with stubborn clicks, while longer fades can help when looping unstable section or chorus sounds. When you have decided on the best settings, save your work.

It should be clear that if you create a crossfade in a sample, and later change the location of its loop points, the crossfade will no longer work properly. A good practice is to keep a backup copy of the original, unadulterated sample. Nondestructive crossfading is on the Audio Compositor "wish list."

Resample

Calls up the Resample dialog, which allows you to resample or pitch-shift the sample.

Resample

This function changes the sample rate of your audio data, and optionally changes the file's nominal playback rate to match the new sample rate. If the playback rate is not changed, the operation is referred to here as "pitch-shifting" and you must specify the new pitch. The result of this operation will only be correct if you have set the pitch of the original sample accurately. That is, if the current pitch is G4 and you request a shift to A4, the sound will be transposed up one step whatever its real frequency.

If the playback rate is changed along with the audio data, then the apparent pitch of the sound will remain the same and we use the term "resampling." In this case you simply specify the new rate.

The available resampling methods are described under [Resampling Algorithms](#).

You should be aware that some sound cards cannot play sounds at non-standard sampling rates. 44100, 22050, and 11025 Hz are of course the safest bets. There are also cards (including at least one prominent and expensive model) that play non-standard rates but at severely degraded quality.

Note that Audio Compositor never requires you to resample manually, since it can accept input material at any sample rate.

Change playback rate

Calls up the Change playback rate dialog, which changes the nominal playback rate of a sample without modifying its audio data.

Change Playback Rate

This function changes the file's nominal sample rate without modifying the audio data.

Synthesize

Calls up the Synthesize dialog, which can produce a few simple waveforms.

Synthesize

This dialog allows you to generate some simple waveforms. Calling this "synthesis" is going a bit far--it's included here mainly as a curiosity, though if you are patient and know exactly what you're doing you can manage a crude sort of additive synthesis with it. (One word: detune.) Expanding this feature for serious use really falls outside the scope of this version of Audio Compositor, but I'm open to suggestions.

Replace existing sample

If you check this box, the contents of the current WAV file will be discarded and replaced by the new material.

Add to existing sample

The new material will be mixed with your existing sample.

Length

The length of the new sample, specified either in milliseconds or in cycles. If you choose the latter option, the length will be so many periods of the frequency set in the **Pitch** box. The **Length** options are grayed out if you have checked **Add to existing sample**, because in this case the current length is never changed.

Pitch

This specifies the frequency of the new waveform, either as a musical pitch (e.g. "C 4" or "A#5") or in Hz.

Waveform

Pretty slim pickings.

Envelope

This determines the amplitude of the signal as a function of time. The envelope is drawn with the mouse--the procedure is identical to drawing an Amplitude Envelope in the Patch Editor.

Guess Pitch

Despite its tentative sound, the **Guess Pitch** function often works very well. It estimates the pitch of a small section of the sample. Normally this section is in the very center of the file, but if you select a portion of the wave with your mouse, Audio Compositor will focus its attention near the center of your selection.

Guess Pitch produces a message like the following:

The estimated pitch is D#3 -15.29 cents. Change to this value?

Answering "Yes" sets the pitch of the sample to the proposed value. Alternatively, you can tell **Guess Pitch** that it guessed too high or too low; in this case it will try to produce another estimate. Because **Guess Pitch** tends to be either wildly off the mark or quite accurate, the process usually takes only a few iterations, and you will not need an especially good ear to tell you whether a reasonable guess has been arrived at. Of course, **Guess Pitch** is not infallible, and certain types of sounds confuse it completely. In these cases your ears become more important!

Since **Guess Pitch** operates only on a small section of the file, it may not work well for samples whose pitch is unstable. For example, samples with vibrato can give varying results depending on where the selection is placed. Taking several estimates from different parts of the file may produce a useful average.

The Layer Editor

The Layer Editor does two related jobs. In the "Samples" area of the window, it keeps a list of the samples that are available on your computer. Each sample is assigned a "friendly" name, and the sample list displays this name rather than the full directory path to each sample's .WAV file.

The "Layers" area is used to create, copy, and delete layers. Clicking on the keyboard at the top of the screen will map the currently selected sample to the currently selected layer. Double-click to unmap a key.

Mapped keys can be "played" by clicking with the right mouse button. This requires some patience as a key will not speak immediately the first time it is pressed. A better way to audition a layer is to use the [Realtime Engine](#).

The Layer Editor stores all of its data in a single file called **Ac.lay**, in Audio Compositor's home directory.

[Maintaining the Sample List](#)

[Maintaining the Layer List](#)

[Mapping Samples to Layers](#)

[Auditioning a Layer](#)

[Menu Commands](#)

Maintaining the Sample List

The sample list is maintained using the **Add**, **Edit**, and **Delete** buttons, which hopefully are self-explanatory. Double-clicking the name of a sample is a shortcut for the **Edit** button. A fourth button toggles the sample list in and out of a special tree view.

When you click **Add**, you'll be greeted first with a normal Windows file-selection dialog; use this to locate the .WAV file for the new sample. If you'll be adding several samples from the same directory, you can select all of them at once. After you OK your selection, the sample parameters dialog will appear once for each of the .WAV files you selected, allowing you to adjust the settings listed below.

Each entry in the sample list has the following settings:

Path

The *full path* to the .WAV file containing this sample. You can type it in by hand, or use the **Browse** button to locate the file. If you move this file at a later time, you will need to edit this field accordingly, or Audio Compositor will not be able to find the sample.

When you're adding a new sample, the **Browse** button doesn't appear, because you will have just finished selecting the file anyway.

Name

Each sample is given a short name in addition to its full file specification. This makes the layer editor a little easier on the eye. The recommended practice is to name the sample after its file--that is, if full path of the WAV file is

e:\Samples\Trumpets\Trumpet F 5 mf.wav

then the sample should probably be named "Trumpet F 5 mf". For new entries, Audio Compositor will suggest default names according to this convention.

Level

This setting gives you a way to balance the sample's volume against others that will be mapped to the same layer; it is given in dB and is normally set to zero.

Fixed Pitch

Check this box if you don't want the sample transposed to match the particular note that triggers it. This is appropriate for unpitched percussion instruments, sound effects, etc.

Ignore Key-up

Check this box if you want the full length of the sample to be played out regardless of the length of the note that triggers it. This behavior can be useful with certain percussion sounds. You can get the same effect by setting a very long release phase in the amplitude envelope, but setting **Ignore Key-up** is slightly more efficient.

This setting is applied only during the realization of a MIDI file--it has no effect when you are listening in the editor. It has no effect at all on looped samples (since they would play forever if it did).

See also:

[The Sample List Tree View](#)

The Sample List Tree View

If your collection of sounds grows quite large, the sample list can become inconvenient to use, since you may have to scroll through hundreds of entries to find a particular sample. In a case like this you would like to be able to organize your samples into groups, with a group for each instrument or type of instrument.

The sample list's "tree view" mode is designed for this purpose. Pressing the **Tree View** button once activates this mode; pressing it again restores the usual flat alphabetical list. The tree view organizes your samples according to their positions on disk. It looks a bit like the left-hand side of the Windows Explorer, with everything but Audio Compositor's .WAV files missing. If you have established an orderly system of directories and subdirectories for keeping your samples, it will be reflected in the tree view. Conversely, if your samples are spread all over with no rhyme or reason, the tree view will look thoroughly incomprehensible!

Path names beginning with a dot are treated as a special case; the tree view shows them as belonging to the "Audio Compositor Home Folder," which represents the directory where Ac.exe is installed. The guitar samples that come with Audio Compositor are designated this way to make the installation process simpler, but you do not need to use this convention for your own samples.

Maintaining the Layer List

When you select a layer in the "Layers" box, its keyboard mapping is displayed across the top of the window. A couple of additional parameters are displayed inside the box. To modify the keyboard map, see [Mapping Samples to Layers](#) on the next page.

When you have modified a layer, you must press the **Save** button if you want to keep the changes. Alternatively you can press **Save As** to save the edited layer under a new name. Saving a layer does *not* write the changes to your hard disk--when you are finished editing and saving one or more layers, you must then save the entire layer file using the regular Windows **Save** command from the **File** menu. This two-level arrangement may sound hazardous, but Audio Compositor will prompt you if it thinks you're forgetting a step.

The layer list has no "Add" button. The way to create a new layer is to start with an existing one--usually the "Empty Layer," which is always present--modify it, and then save it under a new name using the **Save As** button.

To summarize the buttons in the Layers box:

Save

Commits any changes you have made to the currently selected layer.

Reset

Rolls back any changes you have made to the currently selected layer since you last pressed **Save**.

Save As

Saves the current layer, but under a new name. The original layer remains unmodified.

Delete

Deletes the current layer (but not its component samples).

Clear

Unmaps any samples from the keyboard, making a clean slate of the currently selected layer.

In addition to its keyboard map, a layer has the following properties:

Level

Use this to adjust the amplitude of the layer relative to other layers. It is given in dB and is normally set to zero.

Comment

A plain text field that you can use in any way you like.

Mapping Samples to a Layer

Clicking on the keyboard maps the currently selected sample to the currently selected layer. A sample can of course be mapped to any number of individual keys. Double-click on a key to remove the mapping.

Audio Compositor regards middle C as C4.

You'll see that as you add samples to the keyboard, it changes color. Gray keys have no sample mapped to them; the other colors have no individual significance, but they alternate to show you the split points between different samples. If you move the mouse over a mapped key, the name of its sample is displayed just below the keyboard.

Auditioning a Layer

The best way to audition a layer is to start realtime input and play it from your MIDI keyboard. Apart from being more "musical," this method plays looped samples cleanly, even with fractional loop lengths. The first time you play a note in each zone of the keyboard map, you will notice a slight delay as the corresponding sample is loaded from disk.

If you don't have a MIDI controller attached to your computer, try the Virtual MIDI Keyboard.

If realtime output simply doesn't work on your computer, you can audition the layer by clicking the layer editor's own keyboard with the *right* mouse button. Again, there may be a slight delay, and if you release the button before the sample is loaded, you won't hear anything at all until you right-click again on that key.

Samples played using the layer editor's on-screen keyboard accumulate in memory: the amount of memory tied up in this way is displayed in the "Cache" box. If you overdo it and your system becomes paralyzed, hit the **Clear Cache** button to purge some of the samples from memory. For a more thorough housecleaning, press **Clear All**.

If realtime playback is running then memory management is handled by the realtime engine, and controls in the "Cache" box have no effect.

Right-clicking on a key that has no sample mapped to it (a gray key) does nothing.

Layer Editor Menu Options

The Layer Editor has just a couple of its own menu options:

Edit | Reparent file pointers

Reports | All samples

Reports | Samples for current layer

Reparent File Pointers

If you have collected a large number of samples, you will have invested a great deal of time in telling Audio Compositor where to find each one. If you later decide to move some or all of your .WAV files to a different directory, or to a new disk drive, you could be faced with the rather mind-numbing task of editing the file path specification for hundreds of samples.

This function is intended to ease such situations. Please note that it does *not* cause your .WAV files to be moved from one place to another. It's used when you have moved the .WAV files by hand, and need to correct the now-invalid pointers in the sample list.

Reparent file pointers is very simple and rather dangerous: it simply scans the entire sample list for .WAV file paths that begin with the "Old Root," substituting the "New Root" in its place. A typical case might be:

Old Root: c:\Ac\Samples
New Root: d:\Samples

This is entirely a string-replacement chore, and Audio Compositor will not try very hard to stop you from entering completely irrational values. Be sure to check the results thoroughly before saving your work.

Reports | All Samples

This produces a report describing all of the samples known to the layer file. It gives the name of each sample and the full path to its .WAV file, and in addition it opens each .WAV file and reports information about its pitch, loop points, etc. For large sample collections this can be a fairly slow process.

If you have an entry for a sample whose .WAV file has been deleted or moved, this will be noted in the report. Thus, running a report is a convenient way to validity-check your sample list.

The report is called Samples.txt and is created in the current directory. For convenience, Audio Compositor will try to open it for you when it's ready using Write.exe (which on Win95 usually means WordPad).

Reports | Samples for current layer

This report is identical to the [All Samples](#) report, but includes only samples referenced by the currently selected layer.

The Patch Editor

Patches are at the top of Audio Compositor's "architecture"--they are the objects that correspond to MIDI program changes. Use the Patch Editor to create a patch and assign layers to it (the layers must already have been created in the Layer Editor). The Patch Editor also assigns amplitude envelopes, controller routings, and the like.

Patches are stored in a patch file, whose filename has a .PAT extension. A single file may contain up to 128 patches, numbered 0 through 127, plus a special patch at program 128, which is automatically assigned to MIDI channel 10 if present--this is for a General MIDI drum patch. When you render a MIDI file with Audio Compositor, you will tell it which patch file to use (or alternately, which [SoundFont](#)). If your MIDI files are geared for several different setups then you might want a patch file to match each of them--otherwise one patch file might be all you ever need. Storing multiple banks of patches in a single .PAT file is not yet supported.

When the patch editor is open, you can audition patches using an external MIDI controller or the [Virtual MIDI Keyboard](#).

Details on using the patch editor:

[The Patch Editor Tree View](#)

[The Patch Editor Toolbar](#)

[Patch Parameters](#)

[Auditioning a Patch](#)

[Per-Layer Parameters](#)

The Patch Editor Tree View

The patch editor is divided vertically into two panes. The left-hand side gives you a "tree view" of the patch file's contents, and follows a style familiar from many Windows applications. Expanding a patch item reveals its component layers; expanding a layer reveals the samples that are mapped to it.

The right-hand pane shows details about the currently selected patch and layer, and is described in more detail on subsequent pages.

The Patch Editor Toolbar

The Patch Editor's toolbar has four buttons. Note that these functions are duplicated on the **Edit** menu.



Create Patch

Click this button to create a new patch. The Patch Editor creates an empty patch called "New Patch" at the nearest convenient patch number. To change the name and number to the values you really wanted, edit them in the right-hand pane of the editor window, and press the Apply button in the top-right corner.



Add Layer

Add a layer to an existing patch. The layer is chosen from a list of layers that have been defined in the layer editor.

You can control the order of layers within a patch by remembering that new layers are always added at a point immediately following the selected item in the tree. If you select a patch and press **Add Layer**, the new layer will become the first layer in that patch. If you select an existing layer, the new layer will be inserted just below it.



Delete

If the currently selected item is a patch, delete the patch. If a layer, unmap that layer from its "parent" patch. You can't delete a sample using this view (use the Layer Editor to unmap a sample from a layer).



Change Layer

Place a different layer in the position occupied by the currently selected one. (The new layer will assume all the per-layer parameters that were applied to the previous one.)

Patch Parameters

There are only a few parameters that apply to a patch as a whole, and these are visible at the top of the Patch Editor's right side. Note that this row of items has its own **Apply** button. A patch (not a layer or sample) must be selected in the tree view before these fields can be edited.

Number

The program change number for the patch, which must be between 0 and 127. You may change this number, so long as the new number is not already in use.

Name

The name of the patch, entirely for your own information.

Level

This is used to modify the patch's amplitude relative to other patches. It is given in dB and is normally zero.

Changes made to these items do not become effective until you click the **Apply** button.

The [Audition](#) feature is described on the next page.

Auditioning a Patch

The best way to try out a patch is to start the realtime playback engine, and play the patch using an external MIDI keyboard. If you haven't got an external controller handy, try the Virtual MIDI Keyboard. (The Patch Editor's built-in keyboard, seen in earlier versions of Audio Compositor, became redundant and has been removed.) When auditioning a patch, note that changes to the layer parameters won't be effective until you have pressed the **Apply** button below the Layer Parameters tab sheet.

Per-Layer Parameters

The Per-Layer Parameters occupy a tabbed sheet in the lower-right part of the Patch Editor. This is where most of the interesting synth-like parameters in Audio Compositor are set. You can edit these only when a layer (not a patch) is selected in the tree view.

When you work with this part of the program it is tempting to think that you are editing a layer. Bear in mind, however, that the settings here apply only to an instance of the layer: if you include the same layer in more than one patch, you can assign it different parameters for each patch. You can even assign the same layer twice to the same patch, with two sets of parameters--if you can think of a reason to do so.

The parameters are divided into several groups:

[General Layer Parameters](#)

[Amplitude Envelope](#)

[Auxiliary Envelope](#)

[Key Routings](#)

[Controller Routings](#)

[Crossfade](#)

[LFOs](#)

See also:

[Routing Examples](#)

[Crossfade Examples](#)

General Layer Parameters

Level

The relative output level for this instance of the layer, expressed in dB. Use this value to balance the layer against other layers assigned to the same patch. The normal value is zero.

Enable Filter Modulation (registered copies only)

When requested to do so, Audio Compositor employs fixed anti-aliasing filters to improve the quality of its output. Checking **Enable Filter Modulation** frees these static filters to act in a musically expressive way: you can set the cutoff frequency (see below) and then modulate it with, for example, key velocity or aftertouch. This capability is, of course, standard on most MIDI instruments today. Audio Compositor requires you to activate it deliberately because it incurs a very significant performance penalty.

Base Filter Cutoff (registered copies only)

The initial cutoff frequency for the lowpass filter. It can be the object of a key or continuous controller routing. The default value is 20000, though this is something of a fiction at low sampling rates, since the actual cutoff frequency never exceeds half the rate.

The filter options mentioned here have no effect unless filters are active; see [About Filters](#) for more information.

Amplitude Envelope

The amplitude envelope is specified by a series of coordinates, each with an amplitude expressed in dB and a time value expressed in thousandths of a second. On the screen, the envelope is displayed as a set of line segments connecting points which you can move with the mouse. Each of these movable points is marked with a small circle. The rules are:

Any point can be moved by clicking and dragging, though the first point is "glued" to the left-hand edge of the window.

You can create a new point by clicking anywhere except over an existing point. The envelope can have any number of segments.

Delete a point by double-clicking on it.

The last (rightmost) point is regarded as a "hold" value; it determines the amplitude of the note between the time the last envelope segment plays out and the time the note is released. It's displayed as a horizontal dashed line extending to the right-hand edge of the display.

The envelope's release stage is *not* shown. Its length (again in milliseconds) is simply typed into the "Release Length" box.

With the magnifying-glass buttons, the horizontal scale can be changed to accommodate envelopes of various lengths. A problem arises when the view is scaled to show longer periods of time: the resolution of the graph becomes finer than the resolution of your monitor, so that it may become impossible to position the mouse on exactly the value you want. For example, you may want to set a segment at exactly 9000 ms, but the mouse skips from 8978 to 9007. You probably shouldn't be worried about such a minute detail, but if it really matters, try dragging with the *right* mouse button. This will cause the point to move by single units, lagging behind the mouse cursor if necessary.

Other values set on this page:

Release Length

The length of the single release segment, which is never drawn on the graph.

First Segment Linear

If you're not up to reading the long next paragraph, just check the box.

On your screen, the envelope's y-axis is marked in decibel units, uniformly spaced, and the straight lines representing the segments are accurate within this context. That is, what looks like a straight line on the graph will be an exponential curve if you examine the output with an audio editor. In general, this type of response works well because it sounds like a smooth progression to the human ear, but the attack stage of the envelope is more often than not an exception because the attacks of real-world instruments do not often resemble exponential shapes. Most synthesizer designs treat the attack stage of the amplitude envelope as a special case, and give it a more linear than an exponential shape. Checking **First Segment Linear** causes Audio Composer to behave this way.

Auxiliary Envelope (registered copies only)

The auxiliary envelope is very similar to the [Amplitude Envelope](#). In fact, the only differences are these:

The auxiliary envelope is defined in percentage terms rather than dB.

One may think of the amplitude envelope as being inert or neutral when at its maximum value, which is 0 dB. But it is usually best to think of the auxiliary envelope's neutral value as being the minimum, or 0%.

While the output of the amplitude envelope is always applied to volume (and only volume), the auxiliary envelope may be used to modulate either pitch, filter cutoff, or both.

In keeping with the last point, this page includes two fields not found among the Amplitude Envelope settings:

To Pitch

The degree to which the auxiliary envelope modulates the pitch when at its maximum excursion of 100%. It is expressed in cents. The envelope may be used either to raise or lower the normal pitch; in the latter case this value will be negative.

To Filter

The degree to which the envelope will modulate the filter cutoff at its maximum excursion of 100%. It expressed in octaves, and modifies the cutoff frequency as defined on the [General Layer Parameters](#) page.

This value is meaningful only when filters are present. See [About Filters](#) for more about modulating the filter cutoff frequency.

Key Routings

A *routing* is a linkage between incoming MIDI data and some aspect of a note's performance; the implementation here is similar to that in most MIDI instruments. Audio Composer distinguishes between "key" routings and "continuous" routings. Key routings are evaluated once at the beginning of each note, and may affect such things as its volume, pitch, and envelope, but once the note has begun to sound they have no further effect. Continuous routings are those that can alter the note while it is sounding. To keep the distinction clear, the two kinds of routings are assigned separate pages in the parameters dialog, but they are very similar in most respects.

Routings are one area in which Audio Composer somewhat outshines the average synthesizer. The usual method is to dial in a set of parameters that say, for example, "Please vary the attack stage of the amplitude envelope from two seconds to zero as velocity increases." The attack length will thus decrease in a straight line as you go from minimum to maximum velocity, so at a velocity of 64 the attack has no choice but to be one second long. In contrast, Audio Composer allows you to draw the routing graphically, so that you are no longer confined to straight-line relationships.

To add a key routing, click the **Add** button, and select a source and destination. If the source is a MIDI controller, you will have to specify its number; if the destination is a stage of the amplitude envelope, you will have to specify which stage. When the routing is first added, it will have a neutral shape--that is, Audio Composer draws a "do nothing" function to get you started. To make the routing useful, modify the function by drawing with the mouse (the procedure for drawing is the same as for drawing the [amplitude envelope](#)).

The routing source is displayed on the x-axis, usually with a range of 0 to 127 to represent the range of possible velocity or MIDI controller values. When the pitch wheel is used as a modulation source, its value is "coarsened" to a number in the same range by dropping the least significant bits. You do not need to route pitch bend expressly to pitch, as Audio Composer performs this by default (without dropping any bits).

The source called *Key Number* refers to the position of the note on the 128-note MIDI keyboard. When this source is used, the vertical lines on the graph are spaced at 12-note intervals (that is, at octaves). All of these lines represent C's on the keyboard, and the one drawn in red is middle C (MIDI note number 60).

The y-axis units depend on the destination of the routing, and represent a quantity to be *added* to the current value.

Volume in dB.

Pitch in cents.

Amplitude Envelope Level (for a given segment) in dB.

Amplitude Envelope Hold Level in dB.

Amplitude Envelope Duration (for a given segment) in percent change.

Amplitude Envelope Release in percent change.

These additional destinations are available in registered copies of Audio Composer:

Filter Cutoff in octaves.

LF01 to Volume in dB.

LF02 to Volume in dB.

LF01 to Pitch in cents.

LF02 to Pitch in cents.

LF01 to Filter Cutoff in octaves.

LFO2 to Filter Cutoff in octaves.

Auxiliary Envelope to Pitch in cents.

Auxiliary Envelope to Filter Cutoff in octaves.

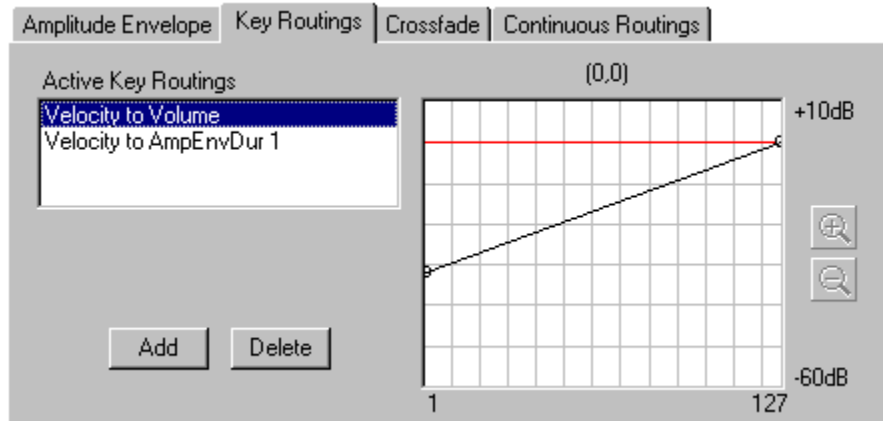
Auxiliary Envelope Duration (for a given segment) in percent change.

Auxiliary Envelope Release in percent change.

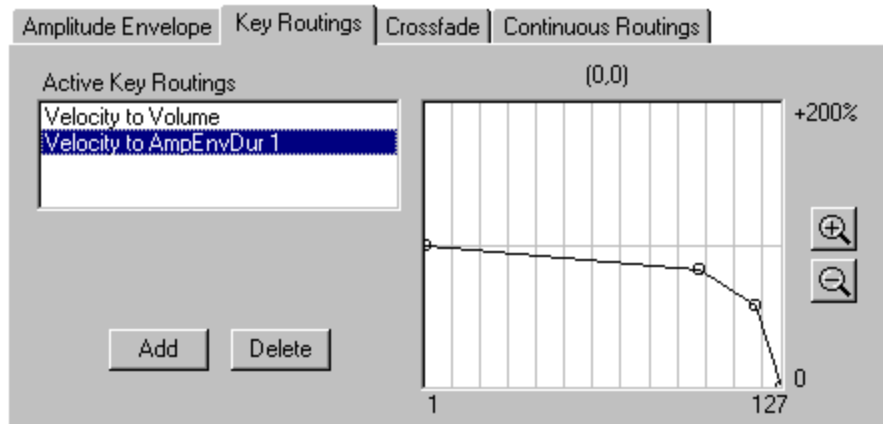
See also [Routing Examples](#).

Routing Examples

For most layers you will probably want route velocity to volume (amplitude) as in this first example. If you want Audio Compositor to mimic the "feel" of a particular MIDI keyboard, you might find that an upwardly-convex shape is more accurate than the straight line shown here. Note also that while commercial keyboards often have 60 dB or more of dynamic range, their factory presets often use less than half of that.



In the next example, velocity is used to shorten the first stage (stage 1) of the amplitude envelope. The effect is slight at lower velocities, and increases steeply at the highest velocities.



In the last example, controller number 7 is routed to volume. Since Audio Compositor will always perform this routing by default, you don't really have to add it unless you want a response that's different from the built-in function--which happens to be exactly the curve shown here. Note that the examples above were "key" routings, but this is a "continuous" routing:

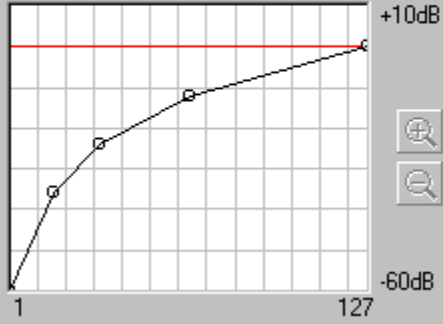
Amplitude Envelope | Key Routings | Crossfade | Continuous Routings

Active Continuous Routings

- Controller 7 to Volume
- Channel Aftertouch to Pitch

Add

Delete



Continuous Routings

Continuous Routings are used in very much the same way as Key Routings; in fact if you have read about the latter there is not much to add. To repeat a distinction made earlier, key routings ignore input which is received after the beginning of a note, while continuous routings can modify the note as it plays. Consequently, the sources and destinations available for routing are somewhat different in the two cases. For example, velocity cannot be the source of a continuous routing because it is measured only once, at key-down. The length of an envelope segment can't be the destination of a continuous routing, because that segment may already have been played out at the time a control value is received.

Note that Audio Compositor routes the Pitch Wheel to Pitch by default. If you add this routing yourself (for example, you want to invert the normal action of the wheel) then the new routing will replace the default behavior. Unless you have something really bizarre in mind, you should take care to cross the x-axis exactly at a value of 64.

The available continuous routing destinations are:

Volume in dB.

Pitch in cents.

These additional destinations are available in registered copies of Audio Compositor:

Filter Cutoff in octaves.

LFO1 to Volume in dB.

LFO2 to Volume in dB.

LFO1 to Pitch in cents.

LFO2 to Pitch in cents.

LFO1 to Filter Cutoff in octaves.

LFO2 to Filter Cutoff in octaves.

Auxiliary Envelope to Pitch in cents.

Auxiliary Envelope to Filter Cutoff in octaves.

Crossfade

This is a velocity crossfade in the usual sense. The fade is between the current layer and the one after it (this happens to be the only way in which the order of the layers within a patch is ever important). Note that applying a crossfade to the last layer in a patch does not really make sense: the layer will fade out over the specified velocity range, but there will be no following layer on which to perform a corresponding fade-in.

The crossfade is specified by its:

Center

The velocity at which both layers are attenuated equally. A value of zero disables the crossfade altogether, and is the default setting.

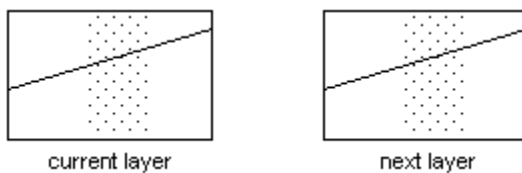
Width

The number of velocity steps over which the crossfade takes place. Half of this range lies on either side of the center velocity. A value of zero causes an abrupt transition from one layer to the next (sometimes called a "cross-switch" in synth parlance).

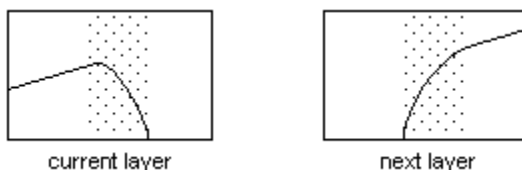
Type

The algorithm used to calculate the crossfade. *Equal Power* uses a constant-sum-of-squares method and is the normal setting. The *Exponential* option, which is of questionable value and is not even very accurately named, yields a slightly different shape. The *Linear* option provides a straight-line fade, and may be appropriate in certain special circumstances.

In effect, a crossfade modifies the velocity-to-volume routing of the current layer and the one following it. You could redraw these routings yourself, but setting a crossfade is more convenient. If the original routings look like:



Then setting a crossfade will make them sound like this:



Note that if the current and following layers have different velocity-to-volume curves to begin with, the crossfade may not work normally.

See also [Crossfade Examples](#).

Crossfade Examples

Crossfades may be "stacked," so if you have sampled a piano at four different dynamic levels you might have an arrangement like this:

<u>Layer</u>	<u>Crossfade Center</u>	<u>Crossfade Width</u>
Piano p	60	0
Piano mf	84	0
Piano f	116	0
Piano ff	0	0

If a layer with no crossfade appears in the middle of the order, the layer after it plays normally (unless it has a crossfade of its own). In the following example, the first layer crossfades into the second, but the third layer is always heard:

<u>Layer</u>	<u>Crossfade Center</u>	<u>Crossfade Width</u>
Violins p	80	20
Violins f	0	0
Solo Violin	0	0

Finally, this rather spectacular example shows three different solo trumpets, each sampled at three dynamic levels, combined to form a unison section patch. At low velocities we'll hear the three **mp** samples, and then with increasing velocity cross through the three **mf**s to the three **ff**s. The center velocities are staggered slightly to further disguise the transitions:

<u>Layer</u>	<u>Crossfade Center</u>	<u>Crossfade Width</u>
Trumpet 1 mp	72	16
Trumpet 1 mf	104	16
Trumpet 1 ff	0	0
Trumpet 2 mp	76	16
Trumpet 2 mf	108	16
Trumpet 2 ff	0	0
Trumpet 3 mp	80	16
Trumpet 3 mf	112	16
Trumpet 3 ff	0	0

If the logic here seems obscure, keep in mind that a crossfade suppresses its own layer at velocities above the fade, while suppressing the *following* layer at velocities below it. Since no crossfade is set on the **ff** layers, the effect "starts over" with each of the **mp** layers.

LFOs (registered copies only)

Two low-frequency oscillators may be defined, and either may be routed to volume, pitch, filter cutoff, or any combination of the three.

The routings defined on this page are constants; they cause the LFO to work at the same depth all the time. It is often useful to leave all of these routings set to zero, while setting up a key or controller routing that will activate the LFO only in response to some MIDI value (e.g. mod wheel, or aftertouch).

Rate

The LFO's frequency in Hz. This parameter will accept a non-integer value, e.g. 3.5 Hz. A value of zero disables the LFO.

Delay

The length of time between the beginning of a note and the onset of the LFO, expressed in milliseconds.

To Pitch

The degree to which the LFO will modulate the pitch at its maximum excursion, expressed in cents. A value of 100 means that the pitch will oscillate between a half step over and a half step under its normal value.

To Volume

The degree to which the LFO will modulate the volume at its maximum excursion, expressed in dB. A value of 3 means that the volume will oscillate between 3 dB above and 3 dB below its normal value.

To Filter

The degree to which the LFO will modulate the filter cutoff at its maximum excursion, expressed in octaves. A value of 1 means that the volume will oscillate between an octave above and an octave below its value defined on the [General Layer Parameters](#) page.

This value is meaningful only when filters are present. See [About Filters](#) for more about modulating the filter cutoff frequency.

About Filters

Audio Compositor does not employ any digital filters unless specifically requested. This is for efficiency--filters consume a far greater share of your processor than any other task that Audio Compositor performs. For each voice, a simple lowpass filter may be brought into play, either as a fixed anti-aliasing filter whose only job is to improve the quality of pitch-shifting operations, or as a time-variant filter which may be used for musically expressive purposes. There are then three possible cases:

If you are rendering a MIDI file without the "Bandlimited Resampling" option, or using realtime MIDI output without checking the "Filters" option, then there are no filters, period.

If you activate the filtering options mentioned above, then every voice has at least a fixed anti-aliasing filter applied.

If you also check "Enable Filter Modulation" in a particular layer's General Layer Parameters, then the filter's cutoff value becomes a routing destination (just as volume and pitch are). The filter's initial cutoff value is defined among the layer parameters, and it can then be modulated by the auxiliary envelope, either of the LFOs, by MIDI key or velocity, or by a continuous controller or aftertouch. (These options are available only in registered copies of Audio Compositor.)

When modulating the filter cutoff, remember that its value is constrained in certain ways. For example, if your output sample rate is 22050 Hz, a filter cutoff above 11025 Hz is not meaningful because the output can contain no information above this frequency anyway. At the lower end, Audio Compositor constrains the filter cutoff to a frequency no lower than five octaves below the sample rate. At a sample rate of 44100 Hz, this frequency is $44100 / 2^5$ or about 1400 Hz. A filter modulation that strays outside these limits will effectively "clipped," possibly giving unexpected results.

Note also that it never makes sense to define a filter cutoff unless you have some expressive effect in mind: for example, modulating the cutoff frequency with MIDI velocity, to get a brighter sound on louder notes. Setting a fixed cutoff merely to correct a sound that is too bright overall is a terrific waste of CPU cycles--in this case, filter the original sample once and for all (using your favorite audio editor) rather than asking Audio Compositor to do it each time the sound is played.

About SoundFont Files

"SoundFont" is a trademark of E-mu Systems, Inc.

While Audio Compositor doesn't come packaged with a great many instrument sounds, it does give you access to what is potentially a wide variety of samples in E-mu's SoundFont 2.0 format. SoundFont 2.0 files, with the extension SF2, descend from the old SBK files familiar to Sound Blaster AWE32 owners. As of this writing (October 1996) SF2 banks are not particularly common on the Internet, though it appears they soon could be. In the meantime, the older SBKs are everywhere, and utilities are available that can convert them into the SF2 format that Audio Compositor can read. The SoundFont capability is provided for users who prefer using ready-made patch libraries to designing their own in Audio Compositor's own patch-editing environment.

[SoundFonts vs. Audio Compositor's Own File Formats](#)

[Limitations of SoundFont Compatibility](#)

[How To Use SoundFonts with Audio Compositor](#)

SoundFonts vs. Audio Composer's Own File Formats

With one exception (see the discussion of ROM samples below), the SoundFont format is entirely self-contained. That is, a single file contains both the audio data and all of the patch-definition data necessary to constitute an entire bank of patches (or multiple banks, for that matter). As such, if you elect to use an SF2 file it replaces not only Audio Composer's layer and patch files, but also the WAV files that comprise Audio Composer's usual sample "library." This underscores a difference in design philosophy: SoundFont banks are designed to fit into RAM on a sound card or other MIDI module, and as of this writing an 8 megabyte SoundFont file is still considered "large." If you are accustomed to using Audio Composer with a sample collection in the hundreds of megabytes, you can see that the single-file approach might be somewhat unwieldy. Size considerations aside, however, you will not be making any compromises if you use SoundFont files in place of Audio Composer's usual formats.

Note that Audio Composer does not provide any tools for editing SoundFont files (though it can display their contents, and extract individual samples into WAV files). In general, AC's SoundFont capabilities are intended for users who prefer pre-edited patches. However, at least one SoundFont editor is available and there is no reason not to use it in conjunction with Audio Composer.

Limitations of SoundFont Compatibility

Audio Compositor is not warranted to be compatible with every detail of the SoundFont specification, particularly in this early release. One incompatibility, which is permanent, concerns the use of "ROM samples." In a SoundFont file, there are two kinds of "samples": those that reside in the file itself, and those that merely point to a location in your AWE32's ROM. If you haven't got an AWE32 or compatible card, the latter type of sample is of course unusable. Furthermore, Audio Compositor never concerns itself with the details of any particular sound card, so even if you do have a Sound Blaster these samples are still unusable from AC (which marks them with a special icon when you open the file). Happily, a great many of the SBK and SF2 files that you will come across probably will not reference any ROM samples.

A second limitation of Audio Compositor concerns certain types of effects processing that may be called for by SoundFont presets. Audio Compositor currently has no reverb or chorus effects. Audio Compositor does observe the lowpass filters that may be defined in SoundFont presets, but only when you select the "Bandlimited Resampling" option for MIDI file rendering, or check the "Filters" option for realtime MIDI playback. The SoundFont parameter modifying the filters' Q or resonance is not observed.

How To Use SoundFonts With Audio Compositor

SF2 files are opened from the File menu in the usual way. The contents of the file (banks, presets, samples, etc.) can be viewed but not edited. Audio Compositor is not a SoundFont editor, and cannot display or modify the parameters in the SoundFont file.

When an SF2 file is the active window it can be played via external MIDI (or the on-screen keyboard) like other Audio Compositor files. In the MIDI file renderer, the Files tab in the Job Setup dialog allows you to select an SF2 file as an alternative to the usual Audio Compositor PAT file. Apart from that, the rendering process is the same whether you are using a patch file or a SoundFont file.

At the present time there is no way to convert SoundFont files directly to Audio Compositor layer and patch files. You can extract the audio for an individual SoundFont sample into a WAV file, by double-clicking the sample's entry in the SoundFont window. Note that samples are the entries at the lowest level of the SoundFont hierarchy, after banks, presets, and instruments. (Double-clicking on one of these parent objects has no effect.)

About the Realtime Engine

When you click the **Play** buttons in Audio Composer's wave editor, the current wave is sent in large blocks directly to your sound card. In this case, Audio Composer behaves much like any conventional audio application.

However, Audio Composer is also capable of producing a continuous stream of audio that reacts to MIDI events--that is, it can work more or less like normal MIDI module that you can play "live" from an external keyboard (or from a special on-screen keyboard). The component that does this job is called (for want of a better term) the *realtime engine*, and it is represented by the control bar at the bottom of Audio Composer's main window. The **Start** and **Stop** buttons turn the realtime engine on and off. You wouldn't want it running all the time, because it puts a considerable load on your CPU.

When the realtime engine is running, any wave, layer, patch, or SoundFont that's currently active on your screen can be played from a MIDI keyboard attached to your computer. In fact, if a .PAT or SoundFont file has the active window, Audio Composer even responds multitimbrally to program changes on all 16 MIDI channels. If you don't have an external MIDI controller, the **Keyboard** button pops up an on-screen keyboard that you can play with the mouse.

Audio Composer's realtime performance varies widely depending on the speed of your processor and the latency characteristics of your sound card, so the realtime engine must be adjusted to run on your particular system. The Settings button opens a window with controls for this purpose. Stuttering and distorted playback are signs that you need to experiment a little (or maybe a lot) here. The controls allow you to limit the quality of realtime output--e.g. polyphony, sample rate, and keyboard responsiveness--in exchange for more reliable operation.

Realtime output is intended mainly as an aid to patch editing, though on some systems it makes for quite a playable musical instrument. In certain cases it is even possible to drive Audio Composer from a sequencer running on the same computer, via a MIDI loopback device. Arrangements like this will become more practical as processor speeds increase.

[Starting Realtime Output](#)
[Realtime MIDI Performance](#)
[Realtime MIDI Settings](#)
[MIDI IN Channel Routing](#)
[The Virtual MIDI Keyboard](#)

Starting Realtime Output

Realtime output is controlled by the buttons at the bottom of Audio Composer's main window. The **Start** button is grayed out unless some playable object (wave, layer, patch, SoundFont) is currently open. Once you have started realtime output, you can change the focus from one window to another; the realtime engine will play whichever window is active at note-on.

The realtime engine's behavior varies a bit depending on what kind of object you are playing:

Wave Files

When the Wave Editor has the current focus, Audio Composer maps the wave file across your entire keyboard. There is no velocity response (because that is not a property of an individual sample), but the sample's loop and pitch will be observed. If no pitch is set, the sample will be played at its actual pitch no matter which note is played. The Wave Editor responds to all 16 MIDI channels.

Layer Files

When the Layer Editor has the focus, realtime MIDI input is directed to the currently selected layer. It responds to all 16 MIDI channels.

Patch Files

When the Patch Editor is the active window, MIDI input is directed to the patch *or* layer that is currently selected in the left-hand pane. Clicking a patch or layer causes it to be selected on all 16 MIDI channels. However, program changes received via external MIDI are channel-specific. That is, the Patch Editor acts like a multi-timbral synth if you send it program changes via the MIDI IN port. (If you change to a program number that is not defined in your patch file, you will get silence!)

The first time you play a given note on your keyboard, Audio Composer must load the corresponding sample from disk. This often introduces a noticeable delay. You can "preload" the patch by touching a key in each sample's key-split zone, and then waiting a few seconds. (If the patch has velocity splits, remember that it may load different samples depending on how hard you strike the key.) Patches which require more than your computer's available physical memory will probably not be playable--if your system begins to bog down, close the active window to unload the samples.

If you have several windows open at once, be aware that the Layer and Patch editors use cached copies of individual samples. If you open and edit a .WAV file, the change may not be reflected in a layer or patch that was already open elsewhere on the screen. After saving a .WAV file, you can force it to be reloaded it by highlighting the layer or patch editor window, and selecting **Cache->Clear**.

Realtime MIDI Performance

Audio Composer does not use the synthesis facilities on your sound card, only its digital-to-analog converters. All aspects of the sound production are performed on your computer's CPU. Consequently, when played from an external keyboard, Audio Composer often cannot produce as many simultaneous voices as a dedicated MIDI module. (Note that this is different from MIDI file rendering, where Audio Composer's polyphony is unlimited.)

Furthermore, while a normal synthesizer knows in advance how many separate voices it can generate, Audio Composer doesn't. The number depends on the speed of your computer, on certain characteristics of your sound card and its driver, and on the amount of any unrelated activity on your system (even mouse movement). When you push Audio Composer too far (by playing too many notes at once) the result is stuttering, distortion, or an interesting dive-bomber effect that may constitute a whole new synthesis method. When you first experiment with this feature, keep the volume turned down low!

Audio Composer manages this situation in two ways. First, a slider in the Realtime MIDI Settings window allows you to set a hard limit on Audio Composer's polyphony. Once you have learned where the boundaries are on your system, you can use this to keep Audio Composer from overtaxing itself if you accidentally play too many notes at once.

Second, a variety of other controls in the same **Settings** window allow you to reduce Audio Composer's workload in exchange for sound quality and musical expression. With many sound cards (especially the less expensive kind) you may need to increase the **Scan Interval**. You can also reduce the output sample rate, or tell Audio Composer not to perform amplitude envelopes or to ignore certain types of MIDI data.

Realtime MIDI Settings

This window controls Audio Composer's response to live input from your MIDI keyboard. Most of the parameters here are meant to help you tailor the program's performance to the capabilities of your computer. For additional information see [Realtime MIDI Performance](#).

Select MIDI IN

This is a drop-down list of all the MIDI input devices that Audio Composer can find on your system. You must select the appropriate one (probably an external MIDI IN port) before Audio Composer can receive data from an external controller.

Output sample rate

The sampling rate at which audio will be sent to your sound card. At low rates you will be able to sustain more simultaneous voices, at the expense of sound quality. In any case, there is no point in using a rate higher than that at which your samples were recorded.

MIDI receive

Here you can selectively discard various types of MIDI data (controllers, pitch bend, channel aftertouch, and program change messages). To discard a given type of message, *clear* the corresponding check box. The effect on Audio Composer's efficiency will depend on how the MIDI events are used in the patch you're playing. These settings usually do not have a dramatic effect on performance.

Poly aftertouch is always ignored in this version of Audio Composer.

Amp Envelope

This check box tells Audio Composer whether to apply the patch's amplitude envelope (it therefore has no effect in the wave or layer editors). Turning off envelopes can improve the program's response somewhat--obviously at the expense of musical expression.

Scope

This checkbox activates the MIDI status box at the bottom of the window, which reports each MIDI message as it arrives (subject to the "MIDI receive" filters). It is a useful gadget when you are troubleshooting a problem, but turning it off will help performance.

Priority

Adjusts the scheduling priority of the realtime playback thread. A setting of +1 will make realtime output more resistant to interference from system activity such as mouse movement. Higher settings may be necessary if you want to minimize Audio Composer while driving it from another application such as a sequencer.

Output level

Adjusts the output level within a range between 0 and -24 dB. This is a digital gain control for controlling clipping in Audio Composer's output, *not* a volume control tied to your sound card's amplifier. If you are hearing distorted output, try reducing this

value before concluding that you are overdriving the CPU--particularly if you are playing lots of simultaneous voices.

Polyphony

As you will soon discover, realtime MIDI polyphony is self-limiting, in the sense that the audio output goes to pieces if you push it too far. It can be useful to ask the program to limit its polyphony deliberately to some specific value, since this can save you from inadvertently over-driving the system by playing too many notes at once. It also helps by cutting off the decay tails of notes as you play in order to stay within the limit. This voice-stealing proceeds on a first-in, first-out basis.

As on most synthesizers, multi-layered patches use up multiple voices in the available polyphony.

The unregistered shareware version of Audio Composer has a maximum polyphony of 6.

Scan interval

Note: this parameter has no effect on Audio Composer's MIDI file rendering process, which is always carried out with single-sample accuracy even when audible output is turned on.

This is the interval at which Audio Composer scans the MIDI input for new MIDI events. It's expressed in milliseconds. Ideally you would like to set it to 1, so that Audio Composer could respond that quickly when you play a note on the keyboard.

Unfortunately, the "scan" is also the interval at which Audio Composer interrupts your sound card to pass it fresh output, and many sound cards fall into conniptions when beaten on at anything close to a 1-millisecond rate. Even with a speedy card, a figure of 10 is more realistic. At this setting, the delay before a played note sounds becomes noticeable, but not hopelessly objectionable. A MIDI sequencer with a resolution of 75 ppq at a tempo of 80 would degrade timing to about the same extent.

Even more unfortunately, the figures mentioned above are still out of reach for many sound cards, particularly the most affordable ones. To overcome stuttering or distorted output, you may need a setting of 30, 50, or even 80. Unless you are a cathedral organist, you will find the delay introduced by these settings very disagreeable, though realtime output can still be a useful aid to patch editing.

Default pitch bend

The default pitch-bend range, expressed in half-steps. (A value of four means two up and two down). It's currently not possible to modify this value via MIDI (write if this is important to you). It has no effect on efficiency.

Buffers x1K

Some soundcard drivers appear to perform much better if output is passed to them in chunks that are a multiple of 1K bytes. Checking this box enforces that condition, possibly at the expense of keyboard responsiveness. Check this box only if it helps.

Slow driver

Some soundcards and drivers respond more quickly than others to system requests. If you get stuttering or distorted playback, and increasing the Scan Interval doesn't help, try checking this box. The downside of this feature is that it will loosen up Audio Compositor's timing a bit.

Experience has shown that most garden-variety sound cards are "slow," so this box is now checked by default when Audio Compositor is first installed.

Save settings

If you check this box, Audio Compositor will remember the current settings (except for the **Save settings** box itself) the next time you start it up.

MIDI IN Channel Routing

This window allows you to route each channel of incoming MIDI data (i.e. from your keyboard) to Audio Compositor, or to a MIDI OUT port of your choice, or to both at once. This makes it possible, for example, to layer Audio Compositor with your sound card's internal synth. The default setting is to send all MIDI channels to Audio Compositor, and none to MIDI OUT.

If you select a MIDI OUT port but do not assign any channels to it, and then close the Channel Routing window, Audio Compositor will close the MIDI OUT port. This avoids tying up a port for no reason.

The Virtual MIDI Keyboard

Surely there is a special place in the next world for people who misuse the word "virtual," but it seems inescapable here. Pressing the **Keyboard** button at the bottom of Audio Compositor's main window pops up a keyboard control that you can play with your mouse--not a wholly satisfying musical experience, but useful at times if you do not have the real thing connected to your computer.

From Audio Compositor's point of view, clicking on the Virtual MIDI Keyboard is exactly equivalent to playing notes on an external MIDI controller. Both send MIDI events to the realtime engine, which you should read about before using this feature. Until you have tailored the realtime engine's performance for your computer and sound card, its output may not be acceptable.

The keyboard responds differently to the left and right mouse buttons: the left plays individual notes, while the holding down the right button plays any note that the mouse passes across. The "sustain" checkbox works like a pedal--this is particularly useful with the wave editor, since it allows you to leave a note playing while you drag loop points around. Uncheck the box to turn off any held notes. The **Velocity** slider determines the velocity that's sent with each note-on.

When you are editing samples, layers, and patches, the best way to audition them is via MIDI--using an external controller if one is present, or the "virtual" keyboard described here. Playing Audio Compositor's sounds via MIDI is more memory-efficient than other methods, and it's also the only way outside the MIDI file renderer to hear the effects of fractional loop points.

Sounds can also be played directly from the wave editor (using the control buttons) and from the layer and patch editors (using the clickable keyboards that appear in those windows). Use these methods if you simply can't get the realtime engine working properly on your system.

The MIDI File Renderer

When you open a MIDI file in Audio Composer, the object is not to edit it but to render it as a .WAV file, or as live output to your sound card (or both). The MIDI file renderer displays your MIDI tracks as horizontal bands. (The banding won't be apparent if the file contains only one track--Type 0 MIDI files for example.)

Three large buttons on the left edge of the screen control the MIDI file renderer:



Go

Starts the MIDI file renderer.



Stop

Stops the MIDI file renderer prematurely, closing the output file.



Reopen

Reloads the MIDI file from disk. This can be useful if you are editing the MIDI file simultaneously in a second application.

When you first open a MIDI file, the entire renderer window is shaded yellow, indicating that Audio Composer is set to render the entire file. If you don't want to render all of the tracks, you can select any combination of them by clicking and dragging in the window using more or less the usual Windows idioms: clicking with the SHIFT key down selects a range of tracks; clicking with the CTRL key down toggles individual tracks without disturbing the others.

In the case of large or dense scores it's sometimes advisable to process the file in several passes, rendering a different group of tracks on each pass. To accomplish this, click on a track with the *right* mouse button and check "Subdivide processing" on the menu that pops up. A thick horizontal line is drawn to show where the subdivision will be made. For example, if you draw three subdivisions, the file will be rendered in four passes. If memory is in short supply relative to the size of the samples involved, this may actually be faster than rendering in a single pass.

The renderer depends on a number of parameters defined in the control bar at the bottom of the window:

[Input Parameters](#)

[Output Parameters](#)

[View Parameters](#)

[Advanced Parameters](#)

For other topics related to MIDI file rendering see:

[Preparing the MIDI File](#)

[Running the Job](#)

[Mapping Failure Warnings](#)

[Performance Considerations](#)

Resampling Algorithms
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MIDI Renderer Input Settings

A.C. patch file

Select this option to use an Audio Compositor patch file (and by implication, your layer file and sample collection) to render the output. You must then specify a patch file (click "... " to open the file browser).

SoundFont file

Select this option to use a SoundFont (SF2) file to render the output. A SoundFont file is self-contained, and takes the place of Audio Compositor's patch, layer, and .WAV files.

You must specify an SF2 file (click "... " to open the file browser).

Reload

Reloads the current .PAT or .SF2 file from disk.

Clear cache

Pressing this button while a job is in progress causes the sample cache to be deallocated. This is not useful ordinarily, but may come in handy if you realize too late that you don't have sufficient pagefile available for the current job. See [Memory Management](#).

MIDI Renderer Output Settings

Audible

Asks Audio Compositor to send a copy of its output to your sound card as it works. If the file is too complex to render in real time, you'll hear gaps in the playback (a stuttering effect). This will not affect the quality of output to a .WAV file. This box can be checked and unchecked while the renderer is running.

To File

Checking this box causes output to be sent to a .WAV file, which you must name in the field just below it. File output and audible output are chosen independently of one another.

If this file exists, the new material will be mixed into it--a fact which can be exploited if you want to add new material to a file after the original rendering. (If you do this, be sure the output sample rate matches that of the existing file.)

You can't turn file output on and off while the renderer is running.

Filters

Checking this box changes Audio Compositor's resampling algorithm and enables its variable lowpass filters. Even if you don't use the dynamic filters, this option may improve the quality of output because of the default anti-aliasing filters that come into play. However, it will slow down processing significantly.

The default is to leave this box unchecked.

Stereo

This selects stereo output. If you direct output to a file which already exists, this setting is ignored in favor of the number of channels in the existing file.

This setting can't be changed while the renderer is running.

Sample Rate

The output sample rate. The available rates are 11025 and 22050 Hz, plus 44100 and 48000 Hz in registered copies. Note that most sound cards don't handle 48000 Hz output. Nonstandard rates can be set in the advanced settings.

Attenuation

This is the master gain control for MIDI file rendering--it is combined with the "level" settings found throughout the various editors, to determine the amplitude at which each sample will be mixed into the output file. If the level settings are zero at every stage (sample, layer, and patch), then setting **Attenuation** to zero as well would mean that samples are transferred to the output file with no change in amplitude.

Recommended practice is to keep the levels of the various components centered around an average of about 0 dB, then use the **Attenuation** setting to avoid clipping in the final output. You will need to experiment to find the right value. Very simple scores may need just a few dB of attenuation; dense ones as much as 24 or more.

SoundFont files found on the network often seem to have a good deal of "headroom" built in, and may require no attenuation at all even when run against large MIDI files.

Note that this parameter works oppositely from the level settings--a positive attenuation value *reduces* the amplitude, so it will normally be set to a positive value.

MIDI Renderer View Settings

Progress

Selects the normal MIDI track display and progress indicator.

Cache

Displays the contents of the sample cache. This lists each sample currently in memory, its reference count (the number of voices using the sample at a given instant), and its size in bytes.

The screen will be blank in this mode unless the renderer is running.

MIDI Renderer Advanced Settings

Press the **Advanced** button to set some of the renderer's more obscure parameters. They're organized into a tabbed dialog as follows:

Range

Constants

MIDI

Range

By default, Audio Compositor will render the entire length of the MIDI file; this page lets you restrict it to a certain section. The beginning and ending points are specified in the Measure/Beat/Tick format common to most sequencing programs. Note that measures and beats are counted from 1, but the first tick is numbered 0.

Note that you can also change the processing range by dragging the left and right edges of the yellow-shaded area of the MIDI renderer window. However, setting the range here gives you more precise control.

The range is also shown in "wall time." If you change the M/B/T values, you can press the "Recalc" button to refresh this part of the display.

The **End of Track Padding** setting tells Audio Compositor to continue for so many milliseconds after the end point is reached. This is necessary when you are processing the whole length of a score, because the end-of-track markers in the MIDI file may not give time for the release segments of your envelopes to complete. (If you plan to run the output through some other audio editor in order to add reverb, add a few more seconds of padding to make room for the final decay.)

If you are sending the output of several separate runs to the same file, it is up to you to ensure synchronization by setting the same beginning point each time. (Unless you exit the program, Audio Compositor will remember these settings from one run to the next, so this is not as inconvenient as it sounds.)

Constants

The values on this page govern either the characteristics of the output file, or Audio Compositor's internal behavior:

Output Sample Rate

The sample rate of the output file. If you are adding material to an existing file, be sure to set the same rate as was used to create the original file. The maximum output rate is 48000 (or 22050 in unregistered copies of Audio Compositor).

This entry duplicates the **Sample Rate** field in the MIDI renderer's main window, with the difference that it allows you to enter nonstandard rates--i.e. rates other than 11025, 22050, 44100, and 48000 Hz. Sound cards that handle nonstandard rates gracefully are relatively uncommon (actually 48000 is uncommon as well).

Output Buffer Size

This sets the size of Audio Compositor's output buffers. It is measured not in bytes but in samples, and there is more than one buffer, so the actual amount of memory allocated is several times the number you enter here. 32768 is a good value. Small values (a few thousand) may improve efficiency by conserving physical memory, while larger values may improve real-time rendering if you are memory-rich. The effects will vary between computer systems.

Amplitude Control Rate

This parameter tells Audio Compositor how often to recalculate a sound's amplitude when it is changing over time in response to an amplitude envelope. It is expressed in calculations per second.

Ideally this parameter should be equal to the sampling rate, making amplitude envelopes completely smooth. You will probably want to reserve this degree of perfectionism for final takes, since it can slow down processing severely. Values around 1000 are more practical. At this rate, the segments of your amplitude envelopes become step functions rather than straight lines, and the difference is heard as distortion at fast rates or "zipper effect" at slower ones. The effect is dramatic with a pure sine wave as source material, but often becomes negligible with real instrument sounds.

Pitch Control Rate

Similar to Amplitude Control Rate, this is the rate at which pitch is recalculated against modulation by the pitch wheel, or by LFOs or the auxiliary envelope. Values around 300 work well; higher values slow down the program and have very little benefit.

Attenuation

This duplicates the **Attenuation** value in the MIDI renderer's main window.

Pitch Bend Range

This sets the pitch bend range globally. A value of 4 means that the range is two half steps on either side of center. This version of Audio Compositor does not provide pitch bend range control for individual channels.

MIDI

This crude view of the input file's raw MIDI data is included simply as a convenience. The format is rather cryptic, but the MIDI-literate will be able to make out most of it. The data cannot be modified. MIDI data that Audio Compositor ignores, such as Poly Aftertouch and Sysex blocks, will not be visible because Audio Compositor has already discarded it from memory.

Mapping Failure Warnings

Organizing program-change messages in the MIDI file can be error-prone with Audio Composer. Most normal synthesizers come out of the box with a patch ready to play at each of the 128 program numbers, and bad changes are often easy to spot--a cymbal crash where a flute was expected, for example. Audio Composer is different: particularly in the hands of new users, the patch database is likely to be "sparsely populated," with most of the patch numbers vacant. Patch-change miscalculations thus result only in silence, and may be hard to diagnose.

Activating the **Warnings** option on the **View** menu can be helpful in diagnosing missing output; it is also a good way to scan for subtle errors in a large file that is substantially correct. When this option is active, the MIDI file renderer stops and issues a warning when it finds a MIDI note that doesn't map to any sound. A note can fail to map for several reasons:

- No program change was ever specified on the note's MIDI channel, and no patch is defined in the patch file for program 0 (which is the default).

- A program change was specified, but it was for a number that is not defined in the patch file.

- A valid patch was found for the current program number, but none of its layers includes a mapping for this particular note. (For example, you played a C7 in your tuba patch.)

In the warning box, the current patch may be reported as "0 or vacant". This can mean either that the patch was set (or defaulted) to zero, or that a program change was issued for some other number that was not defined in the patch file.

Note that Audio Composer numbers patches from 0 to 127, so if you are working with a sequencer that counts from 1 to 128 you will need to compensate.

Preparing the MIDI File

Audio Composer reads Standard MIDI Files in the usual Type 1 format. The number of tracks is currently limited to 64, but raising this number is trivial and I'll do it if anyone requests it.

There are certain types of MIDI data that Audio Composer ignores. One of these, Polyphonic Aftertouch, represents a real gap in Audio Composer's MIDI implementation. Sysex data is also ignored.

There is one serious defect in Audio Composer's approach to MIDI files, but you may find it either annoying or useful, depending on your habits. While Audio Composer is processing a given track in your MIDI file, it is oblivious to all the other tracks (except the tempo track). This means that a program change or MIDI controller on one track never affects the playback of notes on another track, even if they share the same MIDI channel. Many musicians deliberately separate same-channel data onto more than one track for a variety of special purposes, knowing that at performance time it will all be streamed together down a single MIDI cable. If you use this technique, you will need to merge such tracks before rendering your MIDI file with Audio Composer.

On the other hand, many musicians use multiple MIDI interfaces to overcome the 16-channel limitation of the MIDI specification. If you are placing multiple tracks on the same MIDI channel and then routing them to multiple MIDI interfaces to gain "virtual" channels, Audio Composer plays right into this approach--every track is effectively on its own channel(s).

Note that the converse of the situation just described is not a problem: if you mix several MIDI channels on a single track, Audio Composer will keep them properly separated.

Running the Job

Click the **Go** button to start the renderer. The shaded parts of the window change color as the program makes progress. When the job is finished a box pops up which gives some statistics about the amount of work that was done; the most interesting number here is probably the **Ratio**--this is the ratio between the time spent processing and the duration of the output file. It's a rough measure of the file's "difficulty."

When you have dismissed the little statistics box, your MIDI file is still loaded and you can immediately set up another run if you wish. For example, you might process a track that wasn't selected the first time around, adding it to the same output file.

To listen to the output after rendering, you must open the output WAV file using another application such as the Windows Media Player or your favorite audio editor. This version of Audio Compositor does not know how to play back the sound from your output file after it has been processed. Audio Compositor's own WAV editor is not designed for files larger than a few megs, so it won't normally be useful for playing your finished productions.

Performance Considerations

Since Audio Compositor makes relatively heavy demands on any computer system, it is natural to ask what can be done to configure it for best performance. The stubborn fact is that in most applications, the program spends the great majority of its time doing sample-rate conversion (i.e., pitch-shifting) and fine-tuning its other operations is not particularly rewarding.

The most important performance factors, then, are the speed of your processor, and the resampling algorithm you choose on the **Constants** page of the job setup dialog. For details about the latter see [Resampling Algorithms](#).

Memory issues are of secondary importance, but you will need to know more about them if you are working with large projects. See [Memory Management](#).

Resampling Algorithms

Audio Compositor can use either of two different resampling algorithms to transform samples when they must be pitch-shifted, or when their own sample rates do not match that of the output file you are creating. Choosing the algorithm means trading off quality against speed of execution.

In the MIDI file renderer, the resampling method is chosen on the "Constants" page of the Job Setup dialog. For realtime MIDI, linear interpolation is used unless the "Filters" option is checked in the Realtime Settings window. Filtering in real time is often not practical even on fast systems.

Linear Interpolation

If you don't have a background in digital signal processing (and I don't), linear interpolation is the method you might devise on the back of a napkin if asked to change the sample rate of a signal. The original signal is treated as a series of straight lines connecting adjacent samples, and the new signal is created by sampling this fictitious shape at the new rate.

While this method has no mathematical respectability at all, it works surprisingly well for smaller ratios--a few half steps--so long as the sound has no very prominent high-frequency components. It is also fast (though this is of course a relative term). Using linear interpolation on a high-pitched glockenspiel sample is a good way to reveal its limitations: the upper harmonics may alias at all sorts of unpleasant frequencies and the fundamental may even seem to disappear altogether. But with most kinds of sounds, you will probably find the method good enough for serious work.

Band-limited interpolation

This is far slower than the linear method. A lowpass filter is applied to remove frequencies that might alias as a result of the rate conversion. It will take some experimentation to decide whether you want to use this method. You are likely to find it is only needed for particular sounds, and there may even be cases where it can make things worse (because the filter is not perfect). A nice enhancement to Audio Compositor would be the ability to assign a resampling method to an individual sample, rather than choosing it globally for an entire rendering job.

Memory Management

When Audio Compositor is running, it allocates memory on the fly each time it needs to load a new sample file. When it is finished with a particular sample, this memory is not immediately freed, since the sample is as likely as not to be needed again before too long. The collection of allocated samples is referred to here as the *sample cache*, and it can grow quite large. During MIDI file rendering, the **Maximum Sample Cache** parameter (specified in the Preferences dialog) limits the cache size; when the maximum is reached Audio Compositor will try to free less-recently-used samples so as to stay within the limit. This effort can fail, if all of the currently cached samples are in use.

The MIDI renderer and any open instances of the Patch Editor all share a single sample cache. The Layer Editor has its own cache. If you have too many things open and find that your system is bogging down, you will find an option to clear the current window's cache on the **Cache** menu. Wave Editor windows don't participate in the cache, but of course each is associated with one memory-resident sample; this memory can only be recovered by closing the window.

The term "cache" is used here somewhat loosely. In reality, some of what is cached will be in physical memory at any given moment, while the rest sits in your computer's pagefile. Allowing samples to be paged in this way may sound pointless--after all, once a sample has been bumped from physical memory, it is as easy to read it again from its original file as from the pagefile. But in practice this is not quite true: your virtual memory subsystem can quickly retrieve the small portion of the sample that is needed at a particular moment, while reloading the entire WAV file would be much slower. Thus, where MIDI file rendering is concerned, raising the cache limit may allow Audio Compositor to run significantly faster. However, there is no need to raise it above the total size of your sample collection.

If the caching limit is set too high, the MIDI file renderer may try to allocate memory that is not available even in your pagefile, and issue an out-of-memory error. Under Windows 95 with a dynamically allocated pagefile, the caching limit should be set safely under the amount of free space available on the disk where your pagefile is located. If Audio Compositor is writing to this same disk, take that into consideration too, or your output file may butt heads with the pagefile as they both grow.

The size of your cache, and the pagefile space available, are displayed continuously in the MIDI file renderer's status bar when it is running.

If you are very short on pagefile space, you may find the the only way to manage a large score is to process it one or two tracks at a time. See the [Tracks](#) section of the Job Setup dialog.

