# PART III

# MULTIMEDIA AUDIO

## PART OVERVIEW

Multimedia digital audio gets the privilege of being the first media to be explored in detail, in our ongoing discussions on various aspects of multimedia technology.

Detailed chapters on various design / production and programming aspects of digital audio take you to the very depth of core multimedia professionalism.

The first chapter introduces the underlying principles of digital audio. Second and third chapters deal with the celebrated sound cards or audio cards in great detail. The last two chapters mercilessly crack down the myths surrounding MIDI (Musical instrument digital interface) technology and let you explore the joy of creating your own synthesized music.

# Chapter 5

# Digital Audio

## CHAPTER OVERVIEW

This chapter is a curtain raiser for our detailed discussions on digital audio technology.

Some of these discussions may sound a lot similar to our earlier discussions on digital data – but please note that digital audio is characterised by its own dimensions and attributes that distinguish them from general digitising process characteristics.

## 5.1 Introduction

Computers were making sounds (noises?) even before the advent of multimedia and digital audio technology.

Small beep sounds meant that the user was trying in vain, to accomplish a task that the computer has failed to recognize - a missing comma… perhaps, and when he pressed the 'Enter' key again and again in irritation, the computer beep was actually a warning to notify that the user was actively engaging himself in the process of impairing the innocent keyboard.

And sounds louder than a polite beep meant something much worse - may be, the paper deck had slipped out of the printer carriage again or your five year old was finally successful in pulling the draw containing all those backup floppies and files, out of the computer table!

Sigh!

Things are much better now and multimedia technology has taken us far away - You can listen to your favourite CD Music album, even during the office hours…if the earphone is small enough to skip the attention of your fellow employees and more so, your boss.

So, Welcome to the world of multimedia digital audio technology!

I can understand your urge to meet those sound blasters and the notorious MIDIs… But, wait!

The digital audio theory first, please.

## 5.2 Digital audio definition

Digital audio is the technology, by which sound signals are represented as a series of binary digital data - necessarily zeros and ones - which the computers can understand and comprehend.

Once digitised, the sounds can be processed like any other binary file, using the relevant audio editing and playback software. It is the responsibility of the audio software, to understand and interpret the sounds properly.

## 5.3 Why going digital?

Though the primary purpose of digitising sound is to make the multimedia computer understand and comprehend audio signals, we do get some additional benefits in this exercise.

There are certain unique features, which Digital audio enjoys over its predecessor – Analog audio. Let us see what they are and how they are beneficial to us.

- Digital audio is less susceptible to degradation or distortion, as it gets edited and processed. Or, we can also say that we have better control over the degree of quality degradation and distortion that is permissible.
- Editing audio with computers makes cutting, pasting and manipulating the parent signals, very easy. Also, lots of special effects like 'echo' or 'reverberation' can be artificially added, thus giving us more room for creativity.
- It is possible to record, edit and mix digital audio, without the need for expensive equipments. In a while from now, we will see how to set up an inexpensive multi track recorder and mixer – right inside your drawing room! On the other hand, Analog audio demands very expensive Hardware and a host of equipments to accomplish the same.
- Digital audio can be stored as digital data, in reliable mediums like the CD-ROMs - thus extending their shelf life to more than fifty years on an average without losses! Compare it with magnetic media like tapes – which are very difficult to maintain, over the years.

All these benefits are, however, subjected to the fact that the original data should have been digitised in high quality. It is rather difficult to improve the quality of a bad digital recording.

Today, digital audio has almost become the 'de-facto' standard for all professional recording / playback and distribution purposes.

## 5.4 Audio sampling

As with other forms of digitizing, Digital audio files are constructed by receiving Analog signals, taking out samples of the same at certain intervals and reconstructing the pattern using zeros and ones. This conversion process is often known as audio sampling.

Special processing chips and electronic circuits, technically called Analog to digital converters (ADC), are used to accomplish this.

Refer the following figure, for a schematic representation of this process.


*Figure 18: ADC and DAC Processes*

## 5.5 Audio sampling parameters

As with any other digitizing process, digital audio data quality is controlled by two main attributes: Audio sampling rate and Audio sampling size. Apart from these two, mono and stereo channels – which characterise all audio signals, also come into play.

### 5.5.1 Audio Sampling Rate

Audio sampling rate refers to the rate, at which sound samples are recorded, from the incoming Analog audio source. Sampling rates are measured per channel, along the X-axis - in terms of Hertz (Hz), which is the unit for cycles per second.

If your Audio card's manual says that the operating frequency is 11 kHz (Kilo Hertz - equal to one thousand Cycles per second), then, it means that around 11,000 audio samples are taken - in one second!

The higher the frequency, the better -because, if more number of samples are taken during a given time interval, then the wave reconstruction is relatively smooth and resembles the original signal – better.

The four sampling rates or sampling frequencies that are often used are: 11.025 kHz, 22.05 kHz, 44.1 kHz and 48 kHz. In order to appreciate the varying quality of audio these different sampling rates can produce, let us construct a table.

*Table 7: Sampling rates and quality*

| Sampling rate (in kHz) | Quality |
|---|---|
| 8 | Telephone voice |
| 18.9 | CD-ROM/ XA standard |
| 32 | Digital radio / Televisions |
| 44.1 | CD Audio |
| 48 | DAT (Digital Audio Tape) recording – used in recording studios. |

I mostly use **44.1 kHz** for all my music recordings.

### 5.5.2 Audio sampling size

Audio sampling size refers to the number of zeros and ones used to record the incoming signal. It is recorded along the Y-axis. This size depends upon the bits of data your digital sound systems can handle. Typically, an eight-bit sound system can record signals at around 256 quantum steps.

Larger audio sampling sizes can produce better quality sounds. A sixteen-bit recording is adequate for all normal usage, though professional recordings demand thirty-two bits or more.

### 5.5.3 Mono and stereo audio channels

In Mono audio recordings, same sound signals are recorded in both left and right channels. But in stereo, they're different.

Digital audio files record stereo sounds by incorporating necessary details in their header section. During playback, alternating byte samples will be sent to two speakers.

Once should be careful when to choose mono and when to go for stereo. Stereo recording results in larger audio file sizes – because data from both channels are to be recorded.

Stereo is required only *for* high quality music.

For other sounds and speeches, go in for mono recording without second thoughts.

## 5.6 Digital audio recording pitfalls

Normally, two types of problems can be anticipated in digital audio recording processes:

- **Quantization:** The value of each sound sample, during ADC process, is rounded off to the nearest integer value and it sometimes results in unwanted background noise.

- **Clipping:** During ADC, if the amplitude of the samples is greater than the intervals available, the wave is clipped in the top and the bottom and it sometimes results in severe distortions of fine music.

The sampling parameters have to be carefully assigned, in order to avoid quantization and clipping.

## 5.7 Digital audio file sizes

Uncompressed digital audio files consume huge disk space. An approximate idea of the disk space requirements - for a specific digital audio recording, can be calculated using the following formula:

Disk space required per second of recording (in Bits) = Sampling size (in Hz) × Sampling rate (in Bits) × Channel Multiplication Factor.

**Note:** Channel Multiplication factor is 1 for Mono recordings; 2 for stereo.

For example, a 16 bit Sound System, recording signals at 44 kHz in stereo will take up 16 × 44000 × 2 = 176000 bytes or roughly 172 KB per second!

## 5.8 Digital audio playback

During playback, the digital audio files are converted back to their parent form viz. analog signals and are fed into the speakers .The process involved is necessarily a reversal of A D C i.e. Digital to Analog conversion (DAC).

Refer the following figure for a schematic representation of the same.

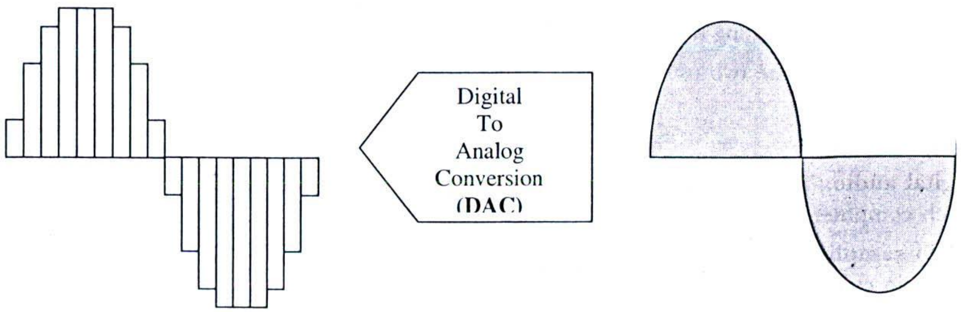


*Figure 19: DAC Processes*

The basic digital sound manipulation systems in almost all of the current multimedia computers are usually built up with their own amplifier circuits and can handle around 4 W at 4 Ohms on each channels; but, if you want to have a really 'surround' stereo effects-drop it at the earliest opportunity and go in for better amplifications using external amplification circuits, accompanied by higher capacity speakers.

It is not necessary that the digital sound should always be converted to analog signals for output. Some of the latest digital audio devices and digital speaker systems can accept digital signals directly from the sound card.

In fact, many professional multimedia audio cards today, come with dedicated jacks for analog output and digital output!

## 5.9 Digital audio file formats

The table lists down all the popular digital audio file formats in use:

*Table 8: Popular audio file formats*

| File format | Remarks |
|---|---|
| Wav | Microsoft's wave file format. Most famous in windows. Uncompressed. |
| Aiff | Apple's Audio interchange file format. Used in Silicon graphics machines as well. |
| Mp3 | MPEG Audio – Layer 3. Most famous in the Internet. Compressed. |
| Ra | Real audio. Well known for streaming audio with no footprints on the local computer. Compressed. |
| Voc | Creative's sound blaster format |
| au | Sun's audio format. |

## 5.10 Summary

Digitising is the processes of representing audio signals as a series of zeros and ones.

Apart from being comprehensible to computers, digital audio has several advantages over analog signals.

Audio sampling rate, audio sampling size and mono / stereo channels - are the three main parameters that ultimately decide the quality of the resulting digital audio signals.

Quantization and clipping are two usual pitfalls that occur during audio digitising process.

## 5.11 Keywords

- **Digital audio**: The technology, by which sound signals are represented as a series of binary digital data, which computers can understand and comprehend.
- **Audio sampling rate**: The rate at which sound samples are recorded from the incoming analog audio source, in cycles per second (Hertz)
- **Audio sampling size**: The number of zeros and ones used to record the incoming analog signal, in bits of data.

# Chapter 6

# The Sound Cards

## CHAPTER OVERVIEW

This chapter features in depth discussions on multimedia sound cards –the physical hardware devices that enable computers to make noise...sorry, music.

It starts with the basic composition of a sound card, followed by various analog devices that can connect to the sound card using various jacks. This is followed by procedures on how to rightly configure the sound card in a windows 2000 environment. The chapter concludes with discussions on audio compression-decompression schemes and other miscellaneous topics.

Schematic and diagrammatic representations of the concepts explained have been presented throughout the chapter.

## 6.1 Introduction

We saw, in the previous chapter, that the digital recording and playback is achieved by means of ADC and DAC Converters. All these conversion circuitry and a host of other electronic circuits that perform various other audio manipulation functions are bundled into the multimedia computers, in the form of electronic Printed circuit Boards. These boards are usually known as the *sound cards* or *audio cards.*

Though the basic function of sound cards is simple digital audio recording and playback, over a period of time, they have grown to encompass complex functionality like MIDI recording / playback, Wavetable synthesis, Environmental audio etc.

## 6.2 Basic composition of a sound card

Almost all sound cards available in the market today, consist of the following devices:

- An ADC Converter for capturing the incoming analog audio signals
- A DAC Converter for converting digital audio back to analog signals
- A Digital signal processor (DSP) for doing various audio computations
- Direct memory access (DMA) Channels for reading and writing audio data.
- A RAM memory chip for dedicated audio memory
- MIDI Interface for connecting MIDI compatible music instruments like Keyboards
- IN and OUT Jacks for connecting microphones, headphones, speakers and many other devices
- A Gaming port for joystick
- A CD ROM Interface

Most of the new generation cards fit into what is called a PCI (Peripheral Component Interconnect) Slot.

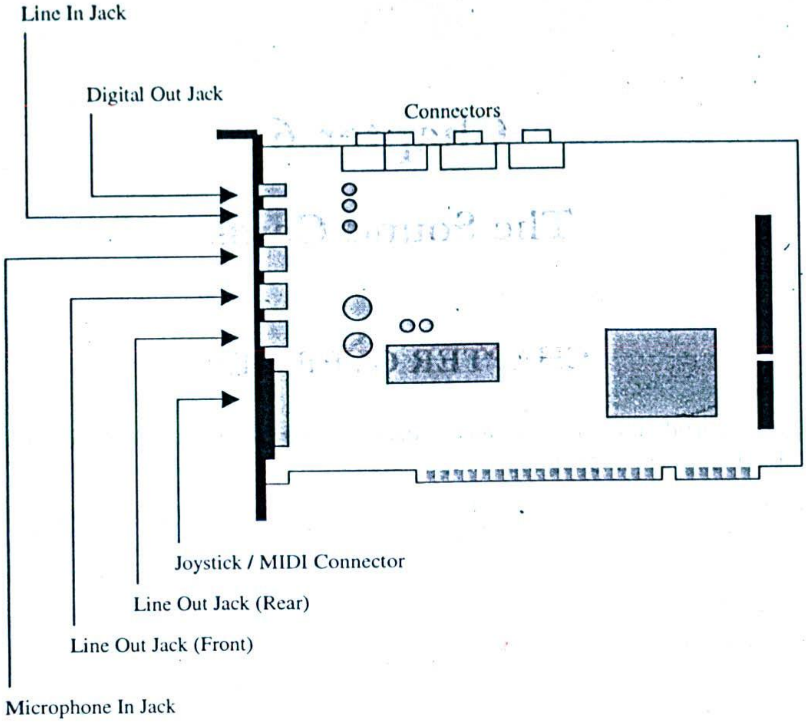


*Figure 20: Basic composition of a sound card*

Please do not buy the relatively older ISA Slot cards – even though they may seem to be somewhat cheaper than PCI! Because ISA audio cards are things of a bygone era and you will soon find that you have to replace it with a PCI.

## 6.3 Sound card connectivity

A sound card may be connected to many input sources and output devices - through relevant external jacks. The following table attempts to provide an overview of these connectivity options available:

*Table 9: Sound card connectivity options*

| Source | Input / Output device | Jack /Connector |
|---|---|---|
| Analog input sources | Microphone | Mic In |
| | Radio / Tape recorder | Line In |
| | CD Player | CD Audio Connector |
| Digital input sources | DAT (Digital Audio Tape) advanced models only) | RCA SPDIF In(Available in |
| | CD-ROM Drive | |

| | | |
|---|---|---|
| Analog output devices | Headphones / small speakers | Line Out |
| | External amplifiers and surround sound speakers | Line Out (Front / Rear) |
| Digital output device | DAT (Digital Audio Tape) advanced models only) | RCA SPDIF Out(Available in |
| | CD-Recordable | |
| | Digital speaker systems | Digital out |
| Others | Joystick / Gaming devices | Joystick / MIDI Port |
| | MIDI compatible music keyboard | Joystick / MIDI Port |

Kindly note that all these different devices require appropriate cables and wires, to get connected to the jacks mea for them. For example, a tape recorder needs a male-to-male stereo pin – to hook on to the sound card.

These cables / pins and wires adopt different sets of international standards. You should choose the devices an sound card in such a way that you could ultimately hook on the devices without much trouble.

The diagram illustrates some of the input / output devices connecting to the sound card.

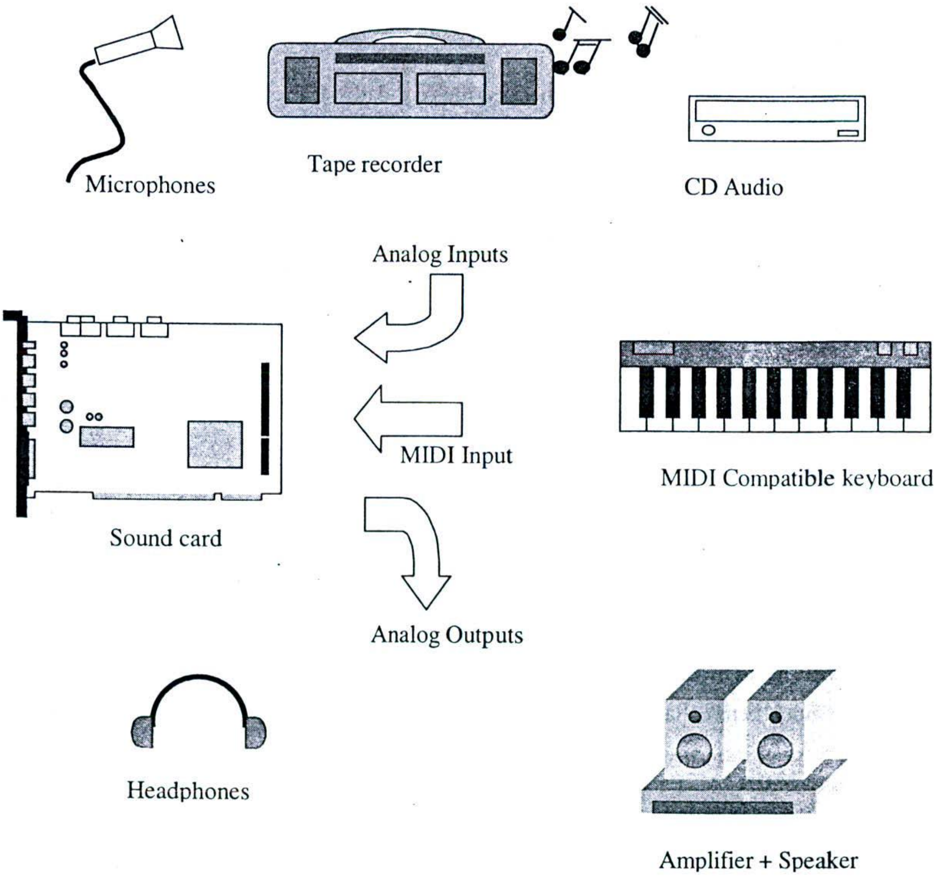


*Figure 21: Sound card connectivity*

## 6.4 Sound cards as CD drive interfaces

CD ROM drives with IDE interface need an IDE card for working with the motherboard. Almost all Sound cards today come bundled with this IDE interface for CD ROM Drives. Hence, if you are buying a Sound card, you may as well save a slot in your Motherboard by not fixing an IDE interface card.

A small cable links this Sound card IDE interface with CD ROM Drive. Your sound card kit should come bundled with this cable.

## 6.5 Music synthesis

The process by which a sound card "creates" Music is called *the Sound synthesis or the Audio synthesis technology*. In fact, many of the music keyboards we know - like Casio and Yamaha, make use of similar technology to reproduce the notes of various tunes and percussion instruments.

There are two fundamental types of Audio synthesis: *The FM Synthesis* and *The Wavetable Synthesis.*

### 6.5.1 The FM synthesis

FM synthesis is a relatively older technology.

By mixing two different sine waves, called carriers and modulators – FM Synthesisers produce complex waveforms that resemble the tunes from various music instruments. And by mixing various types of notes thus produced, complex music can be generated.

Yamaha, one of the pioneering companies in the synthesised music today, was the first to capitalize on this technology and almost all of the early Music synthesisers from this company were actually FM synthesisers.

As technology matured, a wide range of careers and modulators were combined in various permutations and combinations to produce almost all conceivable music notes from various instruments. Thus music synthesisers and keyboards got richer and were adopted widely in the music industry.

But one thing was a handicap – though.

FM synthesis sounds were still machine generated and subtle differences between them and the actual music instruments were, very much, noticeable. This led to the development of another technology – called wave table synthesis.

As on date, FM synthesisers are very much in use – but are usually at their best, when you want to generate artificial sounds – like those from robots / machines etc.

### 6.5.2 The wavetable synthesis

Instead of blending around with modulators and carriers, why not record the digital samples from the actual instruments themselves and play them back at the relevant keystroke? Well, that is what wave synthesis is all about!

Wave synthesis cards store digital samples of various instruments in RAM memory. Each sample is a digital representation of the actual waveform recorded directly from the original instrument. When a note from a particular instrument is played, the card actually looks up for an equivalent in its wide collection of digital audio samples and reconstructs the sound with those models.

The resulting sounds are much better than their FM counterparts.

By having a wide variety of samples – often called Music banks – Sound cards are able to provide realistic representations of complex notes like Beethoven's 7$^{th}$ symphony. These sound banks are naturally extensible – as they are nothing but a database of digital samples - and hence, it is possible to expand the number of instruments that are available at one's disposal.

How real, can Wavetable synthesis, bring Computer generated music close to the original ones? Well, it depends on the following factors:

- ❑ The overall quality of the sample recordings. Better the quality closer will be the reproduction.

- ❑ The frequency used for recording samples. Suitable frequencies should be adopted for playing back various ranges of notes.
- ❑ The number of samples recorded per instruments. Wider variety of samples ensures better range of notes. But more samples obviously result in more memory.
- ❑ The audio compression algorithms used to store and retrieve samples. Compressions that do not deter the quality of original recordings will result in high quality reproductions.

## 6.6 Motherboard integrated audio peripherals

Instead of having the pain of installing a new additional peripheral for handling audio, many motherboard manufacturers, these days – bundle sound card capabilities right into the motherboard itself, in the form of a dedicated audio chipset.

While this option certainly saves some bucks from leaving your purse, it is generally noticed that the audio capabilities they offer are very minimal and are unlikely to satisfy a multimedia professional.

Hence we will not be discussing much about them.

## 6.7 Configuring sound cards under Windows 2000

When a new Sound card is inserted into the PCI Slot and the computer is rebooted, it is highly likely that Windows 2000 will find a suitable driver for the card from its database and may install the same, if you give the go ahead. It may ask for the Win 2000 CDs, if the driver is not available in the hard disk.

However, as we had observed in one of the earlier chapters, it is always better to make use of the manufacturer's device driver, in order to obtain better performances and have greater control over the hardware.

**The lesson is:** *Never ever forget the device driver CD ROM – when you purchase a sound card!*

This should particularly be kept in mind, when you are making OEM purchases – without nicely packaged boxes and installation tool books.

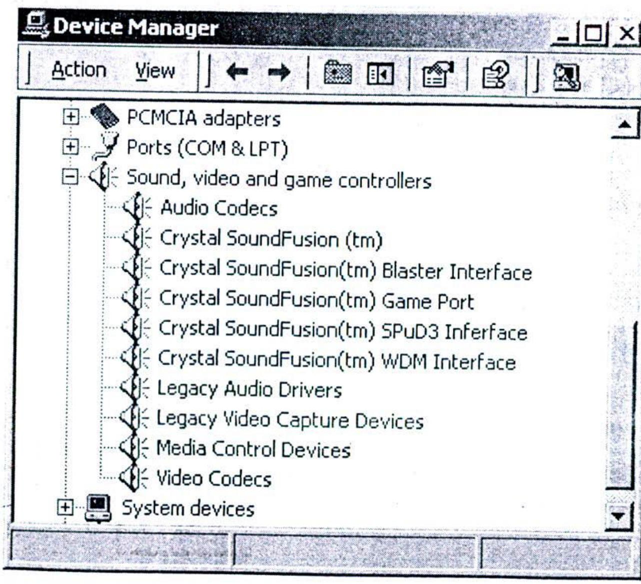


*Figure 22: Device manager panel, showing all the audio devices installed.*

Once a sound card is properly installed on the system, the device manager should list out all the corresponding audio devices – as shown below:

## 6.8 Audio codecs

Apart from device drivers, Windows need one more set of software for playing back a wide variety of sound formats. These are called audio codecs (acronym for compression – decompression algorithms).


*Figure 23: Reaching audio codecs properties*

You can take a quick look at the various audio codecs installed on your system, by right clicking the audio codecs icon under device manager and choosing properties.

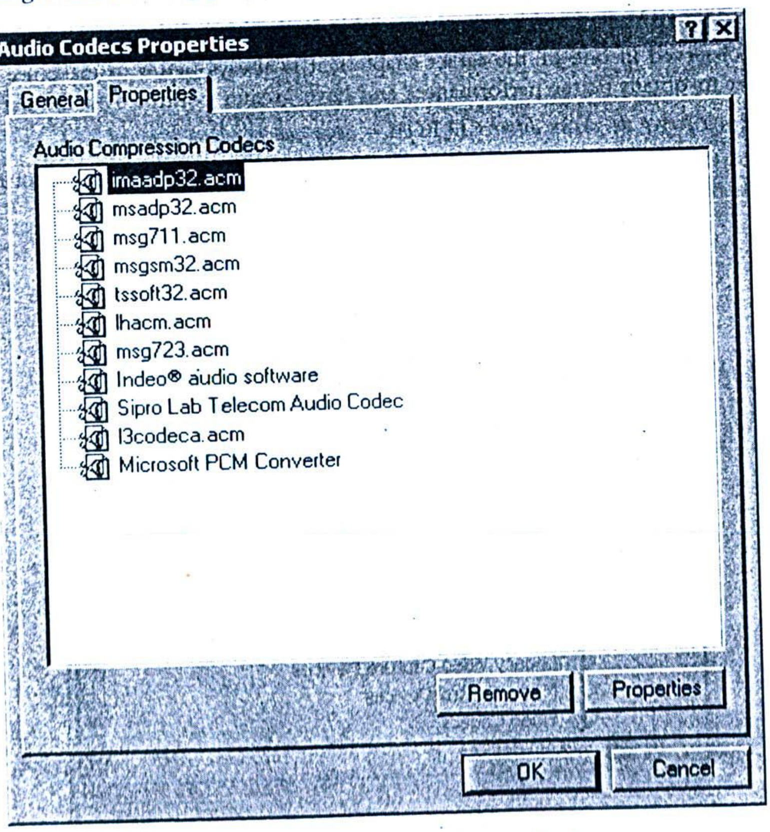


*Figure 24: Audio codecs installed*

Windows does not store audio in its natural form – because of stupendous memory requirements. It makes use of some compression algorithm while storing audio data and applies the decompression algorithms during playback. Mp3 technology – which has revolutionised online music distribution via the Internet – is nothing but an advanced audio compression- decompression algorithm!

Windows 2000 comes bundled with a host of audio codecs, and Windows media player is capable of downloading the relevant codecs via the Internet, on demand.

## 6.9 Capabilities of sound card

Almost all sound cards available in the market today, are capable of performing many functions, as detailed in the table below.

***Table 10: Functionalities of a Sound card***

| Core function | Remarks |
|---|---|
| Digital audio playback | Play back various kinds of digital audio files |
| CD Playback | Play Audio CDs from CD-ROM |
| Digital audio recording | Record music from microphone and various other analog sound sources. |
| Digital audio editing | Manipulate the audio signals recorded as digital data |
| MIDI Playback / Recording | Synthesized music recording / creation and playback |
| Joystick | Providing an interface for Joystick / Gaming devices. |

We will be looking at many of these functions in great detail, in the chapters that follow.

## 6.10 Summary

Audio cards are electronic PCB boards that basically perform analog to digital conversions and digital to analog conversions, apart from a host of other functions. Three main types of audio cards are the sound cards, the midi interface cards and the wave blaster cards. The process by which a sound card creates music is called the sound synthesis. Currently wave synthesis technology is widely in use.

Sound cards perform a variety of complex audio handling functions including digital audio playback and recording, CD audio playback, and MIDI audio playback and recording.

The sound cards have to be properly configured under the host operating system, in order to perform to its fullest potential.

## 6.11 Keywords

- **FM synthesis:** The technique of blending around with a range of frequency modulations to produce audio that resembles the original analog sound data.
- **Wavetable synthesis:** The technique of bundling a wide variety of built in sound samples (i.e. actual recordings) of various music instruments and reconstructing the sounds by looking up for samples.

# Chapter 7

# Audio Recording and Editing Techniques

## CHAPTER OVERVIEW

Having seen so many concepts and theories related to digital audio, are you in a dire urge to "do something"? This chapter shows you the way...

First learn how to play different audio files using different multimedia content players. Understand what is the difference between different kinds of players and wonder why there are so many varieties of players in the market.

A more exciting exercise would be to do a piece of digital recording on your own! You may want to dust out some of your old Hindi song collections in cassettes and convert them as mp3s or sing a karaoke number and distribute it to friends.

Interesting discussions on one of the famous audio editing tools available in the market – Syntrillium's Cool Edit 2000 – makes this chapter even more compelling.

## 7.1 Introduction

Having seen the configuration and installation of a sound card, it is time to make use of them to jazz up our drawing rooms or create some captivating karaoke numbers from the oldies!

In this chapter, we will be learning all the tips and tricks to do professional playback and recordings.

## 7.2 Capabilities of a sound card

*Table 11: Functionalities of a Sound card*

| Core function | Remarks |
|---|---|
| Digital audio playback | Play back various kinds of digital audio files |
| CD Playback | Play Audio CDs from CD-ROM |
| Digital audio recording | Record music from microphone and various other analog sound sources. |
| Digital audio editing | Manipulate the audio signals recorded as digital data |
| Audio file manipulation | Audio file format compression / transformations |
| MIDI Playback / Recording | Synthesized music recording / creation and playback |
| Joystick | Providing an interface for Joystick / Gaming devices. |

Almost all sound cards available in the market today, are capable of performing many kinds of audio editing, recordin and manipulation functions. The following table details some of the functions that can be expected in most card: The rest of our chapter is devoted to mastering each of these functionalities.

## 7.3 Digital audio playback

Today, quite a number of free and shareware software are available in the Internet, for playing a very wide variet of digital audio file formats. The file formats that are frequently used, include .wav, .mp3, .aif, .aiff, .mid, .snd .au,.mpa, .wma etc.

Let us first list out some of the most famous digital audio playback software available today. It should be note that many of the players mentioned below, are not just restricted to digital audio alone; they handle a wide variet of other multimedia file formats as well.

***Table 12: Some of the famous digital audio playback software***

| Software | Remarks |
|---|---|
| Windows media player | Play back various kinds of digital audio files |
| Apple's QuickTime media player | Play Audio CDs from CD-ROM |
| Media one | Play back various kinds of digital audio files |
| Creative media player | Play back various kinds of digital audio files |

## 7.4 Windows media player

Windows media player, from Microsoft – is perhaps, the de facto multimedia player, on the Windows platforn Since its inception in Windows 95 days, this software has been growing from strength to strength. Today, it is a ver powerful application that supports many advanced features in digital media playback and recording.

Windows media player is much more than a multimedia player – it is a Jukebox, CD Recorder and surround soun creator – all combined into one. The only hitch is its massive size: the latest version is around 13 MB!

First, let us see the different versions of Windows media player and the Operating systems they support.

***Table 13: Different versions of windows media player***

| Windows Media Player version | Supported platforms |
|---|---|
| 9 series | Best with Windows XP – but is also available for Win 98 SE, Millennium Edition (ME) and Win 2000. |
| 7.1 | Win 98, Win 2000 and Millennium Edition (ME), Mac OS 8.1 and above |
| 6.4 | Windows 95, Win NT 4 |

The 9 series is understandably a very complex beast with multitudes of functions and capabilities.

In this book, we will limit our discussions to Version 9 under windows 2000; and in the present chapter, ou concentration is on the audio capabilities of the same.

### 7.4.1 Downloading and installing media player

The home of Windows media player is located at the following URL:

*http://www.microsoft.com/windows/windowsmedia/players.aspx*

This page should give you a glimpse of all the latest versions of Media players, available online. Almost all of them should be free for usage. At the time of writing this, Version 9 series was the latest.

Links for appropriate downloads should be available from the above URL – on the left-hand side navigation bar. I downloaded MPSetup.exe and installed the same without a hitch – on my Windows 2000 PC. At the end of installation, you should be careful while associating media player with all the multimedia file formats. Because, Windows media player will then become a default player for the entire range of selected file formats; and you will not be able to invoke any of your favourite players by double clicking on the file names.

Once installed, the default screen pops up and you will be able to access the program from

Start → Programs → Accessories → Entertainment → Windows media player

### 7.4.2 Using media player for digital audio

Media player program has the following menu items: File, View, Play, Tools and Help.

It also has the following button controls: Play, Stop, Fast-forward, Rewind, Previous and Next.

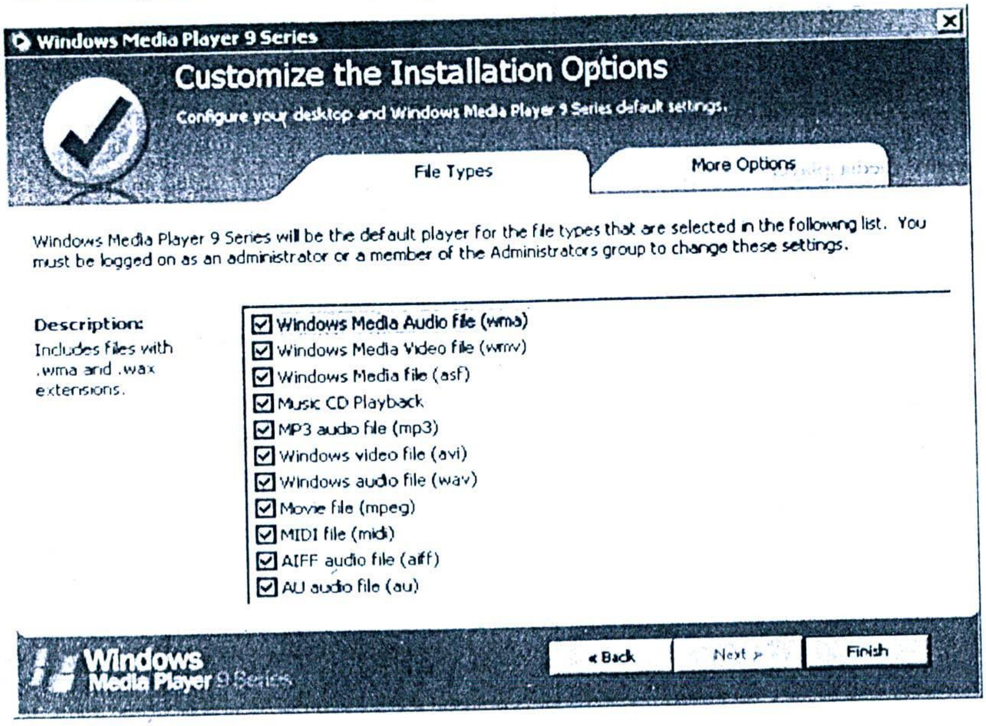


*Figure 25: File format association with media player*

File menu items allow you to open and close all multimedia files - including digital audio files like *.wav, *.snd, *.au, *.aif and *. aiff, MIDI audio files like *.mid, *.midi and *.rmi, Windows media audio file (*.wma.) and MPEG Audio files (.mp3)

Once a valid multimedia file is opened, all the buttons of the media player get activated. To play the file, Click on the triangular 'Play' Button. Those who don't have any other audio files to try can browse the Windows media directory for a sample. For example, if you have installed Windows under C:\Windows, then the media directory is at C:\Windows\Media.

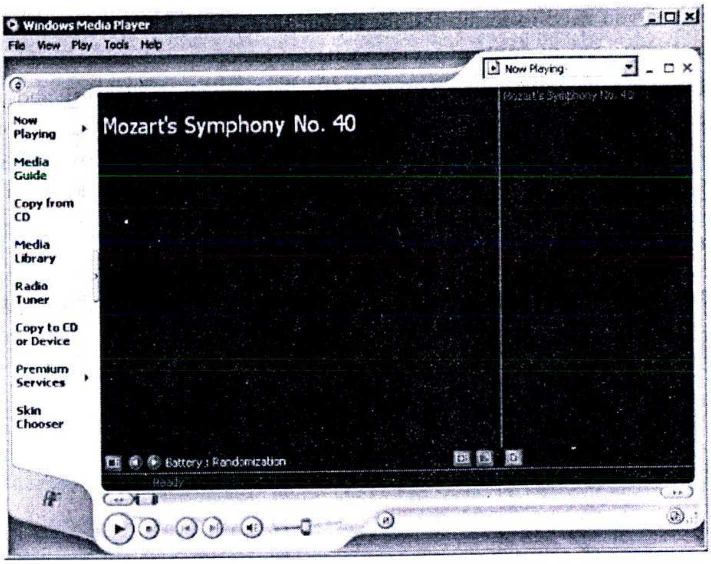


*Figure 26: Media player playing a MIDI file*

## 7.5 Apple's QuickTime

Apple computer's QuickTime is another popular multimedia player software for Mac and Windows. The latest version can be downloaded at the following URL:

*http://www.apple.com/quicktime/download/standalone/*

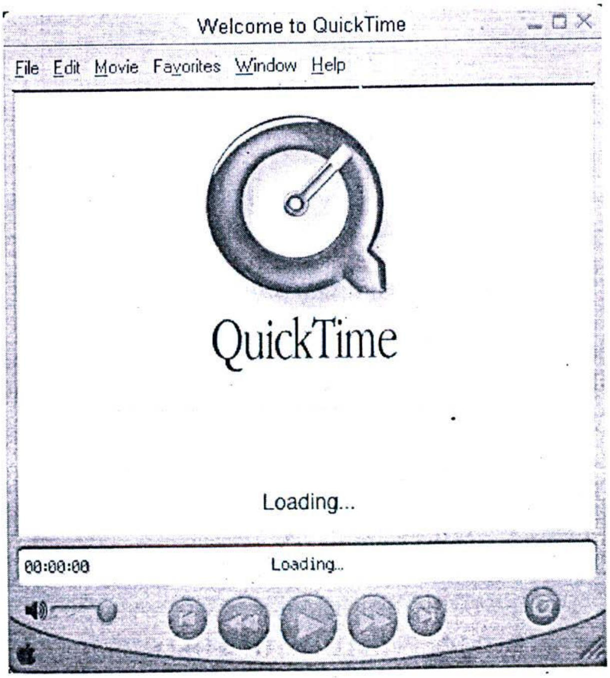


*Figure 27: QuickTime media player*

89100

Like Windows media player, you need to set up the file associations during installation. Once installed, the program can be invoked from desktop icon or from program manager group.

QuickTime plays all the digital audio file formats of the Macintosh platform like .aiff and .au. One hitch, however, is that it needs to convert some of the Windows file formats to .mov before playing... for example, if you try to open a simple MIDI file, QuickTime will say that it needs to convert this to a .mov file before playing – which is really an unwanted trouble. In such cases, you may as well play the file straight away from Windows media player!

## 7.6 Media One+ player

Media One is a popular shareware for playing a very wide variety of multimedia file formats.

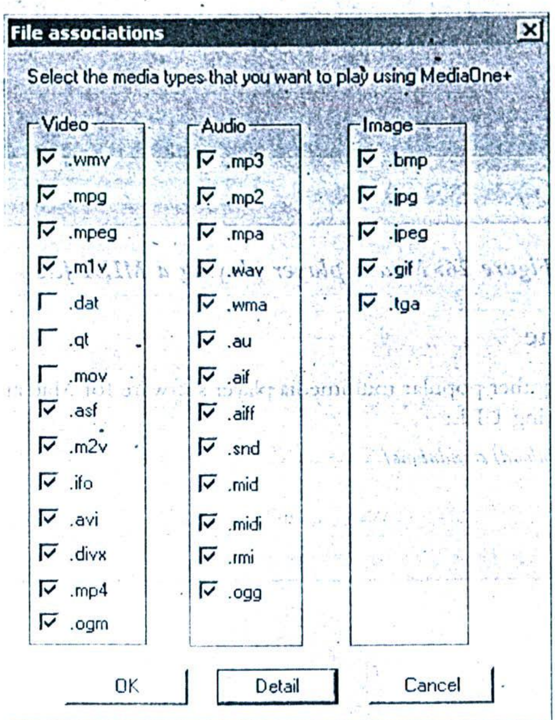


*Figure 28: Media One supported file formats*

It is perhaps the only player with very small memory footprint and size (around 500 KB!) and handles most of the digital audio file formats with great efficiency. Though the sophistication of windows media player cannot be expected in this cute little program, it is well worth a try!

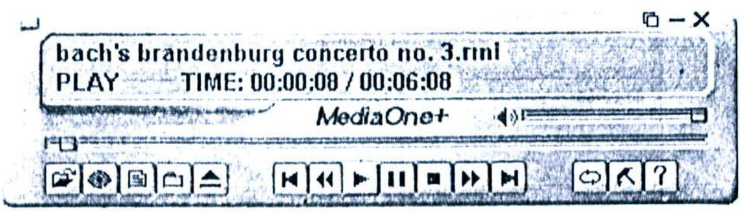


*Figure 29: Media One+ Player*

The program can be downloaded from http://www.media1moon.com

## 7.7 Digital audio recording techniques

Sound cards can be used to record audio signals from various sources like tape recorders, microphones, CD players etc. The recorded signals will be stored in the hard disk (or any other storage mediums), in one of the preferred file formats – usually .wav in Windows.

Why do we want to record digital signals at all? Here are some of the reasons:

- You may want to convert some of your old favourites from audiotapes to mp3 or Audio CDs.
- You may want to convert your Audio CDs to mp3 collections
- You may want to sing and record your own voice or music for your personal project or simply for fun!
- You may want to do professional recordings for your multimedia project or animation video

Many of the strategies and techniques we discuss here, are meant for personal recording environments at small office / home office only. However, before concluding the topic, we will briefly discuss how professional recordings are done in recording studios and how they can be made use of – in our multimedia projects.

## 7.8 Audio recording software

We first need to download and install an audio capture or recording software – before even we can begin the audio recording process.

Microsoft windows operating system comes with a cute little recorder program, suitable for simple and short recordings. It is available under start à Programs à Accessories à Windows recorder. It even has some nice editing options! But, since it cannot record anything beyond a minute, it is not suitable for long recordings. At the best, you can use it to test the quality of microphones etc.

If you have purchased a sound card along with the CD-ROM that comes bundled with it, then it usually has a sophisticated recording program and it is best to make use of the same for all recordings. For example, all Creative's audio cards come bundled with their own audio recording software – with extensive options for doing various kinds of recordings.

If – by some reason or other – you do not have any recording software at hand, then you can download one of the shareware programs from the Internet. I use audio grabber software available from http://www.audiograbber.com-us.net, to do my recordings.

You can also check http://www.download.com - with the search string "sound recorder". Quite a number of shareware and freeware popped up, when I checked.

The recording software usually does not matter much – as long as it has some of the basic capabilities and is able to provide interface to line inputs and microphone inputs. But what does matter – is the recording options you set, as they have a profound impact on the ultimate quality of your recordings.

## 7.9 Recording options

Before we even start the recording process, we need to set-up our environment and decide upon some environment options, like whether to record the signals in mono or stereo mode, at what frequency and sampling rate to record etc.

### 7.9.1 Mono vs. stereo recording

Mono recording usually gives decent results with microphone based voice recordings – without any music. It can also be used for solo instrument performances or for music with minimal set of instruments. However, if you are capturing songs or music from a tape recorder or a CD audio player, better use stereo options.

Stereo recordings usually require much more space than mono recordings – since two different sound signals have to be captured and stored.

### 7.9.2 Sampling size

We have already discussed in one of the earlier chapters, what sampling size was and how it affected the quality of recordings. Here, we will just discuss various sampling size options available and when to use what etc.

The sampling size options, usually available are 8 Bits/sec, 16 Bits/sec, and 32 Bits/sec. The increment in the resulting file size – from 8 bits to 16 bits can be astonishing and hence should be taken note of, before hand. Otherwise, halfway through your recording – you may get disk full error from the operating system.

It is usually better to go for 16 bps recordings – unless you want to compress the ultimate file size very substantially. Even then, you can actually do the recording in 16 bps and then convert the file to 8 bps – using one of the audio editing software.

32 bits are advisable – only for very professional recordings with a wide range of notes.

### 7.9.3 Sampling rate

The usual sampling rates are 11.025 kHz, 22.05 kHz, 44.1 kHz and 48 kHz.

Use anything less than 22 kHz for personal recordings and 44 or 48 for professional recordings. I normally use 44.1 kHz for most of my recordings.

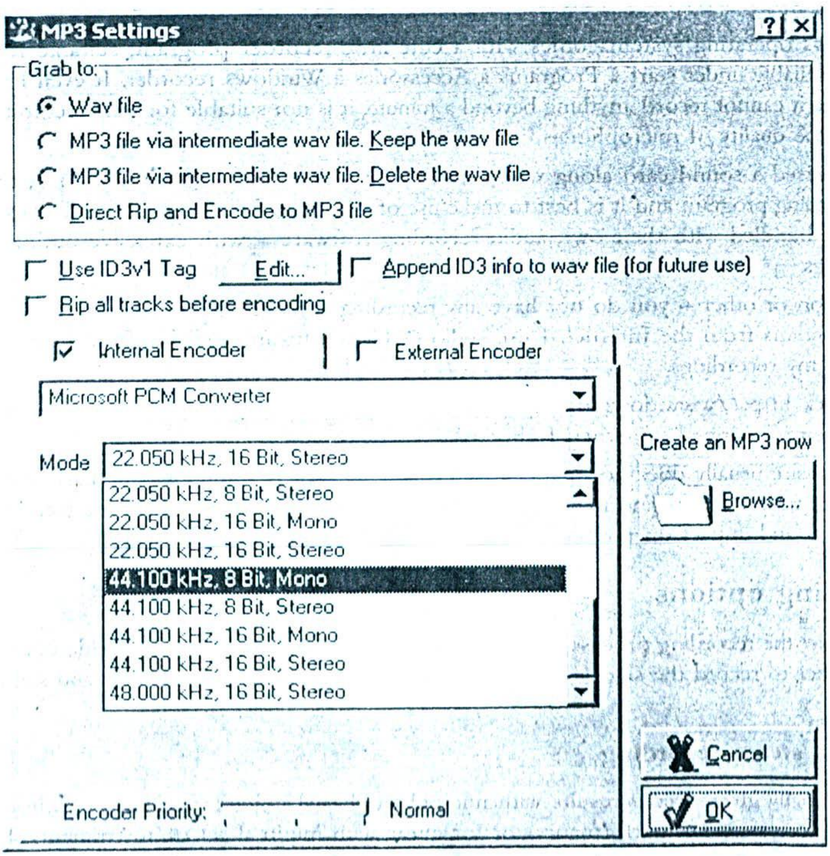


*Figure 30: Setting the recording options in audio grabber software*

Once you have decided upon the recording options, set those options in the recording software you are using. I've provided the illustration – with audio grabber.

### 7.9.4 File size requirements

The following table gives an approximate idea of the file size requirements for a one-minute recording, under various sampling rates and sizes.

*Table 14: Different recording options and corresponding file sizes*

| Channels – Sampling size – Sampling rate | Approx. file size requirements for one-minute recording (uncompressed) |
|---|---|
| Mono – 8 bps – 11 kHz | 5.03 MB |
| Stereo – 16 bps – 22 kHz | 40.28 MB |
| Stereo – 16 bps – 44 kHz | 80.56 MB |
| Stereo – 32 bps – 22 kHz | 80.56 MB |
| Stereo – 32 bps – 44 kHz | 161.13 MB |

Do not worry about storing such huge files in your hard disk!

Usually, once the recording is done, we will be able to edit and compress the file using one of the codecs available and the size will become manageable, by then.

## 7.10 Setting up the recording environment

Doing a microphone based recording is much more difficult than a tape or CD recording – in a SOHO atmosphere.

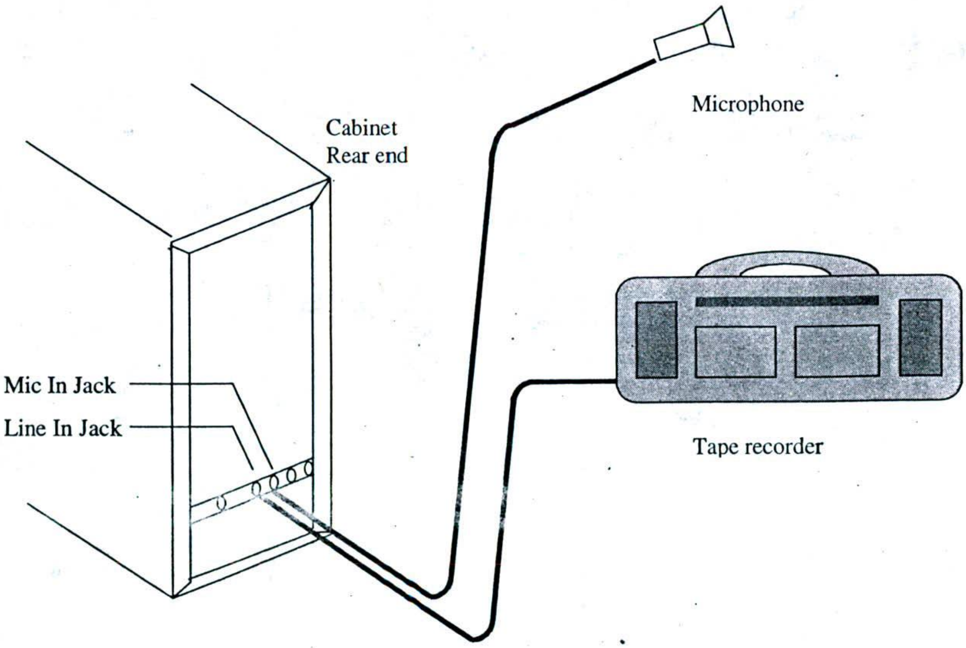


*Figure 31: Setting up the audio recording equipments*

If you are planning a microphone-based recording – select a place with least external noise and disturbances and move your computer there. Connect the microphone wire to the mic-in jack of the sound card, on the rear end of the cabinet – as shown in the figure.

Close all the doors and if possible, have sound absorbing materials like coir board or thermo coal around – so that the echo within the room is minimised. Set up the audio recording software with desired settings and go ahead with the recording as desired.

If you are recording something from Tape or a CD Player, then it becomes a pin-to-pin transmission of audio signal, and best results can be expected – without disturbances. All you need to do is to buy an appropriate cable with male-to-male stereo pins on either end. Then you can insert one end of the cable into sound card's line-in jack and the other into headphone jack and go ahead with the recording.

## 7.11 Various steps in audio recording process

1. Keep the microphone / line-in socket plugged into the sound card. If the audio source is a tape recorder, hook up the 'headphone out' or 'speaker out' jack to 'line-in ' jack of the sound card.
2. Activate the sound recorder or any other recording software from the accessories program group.
3. Verify that the volume control is adjusted properly and that the program properly receives the incoming signals.

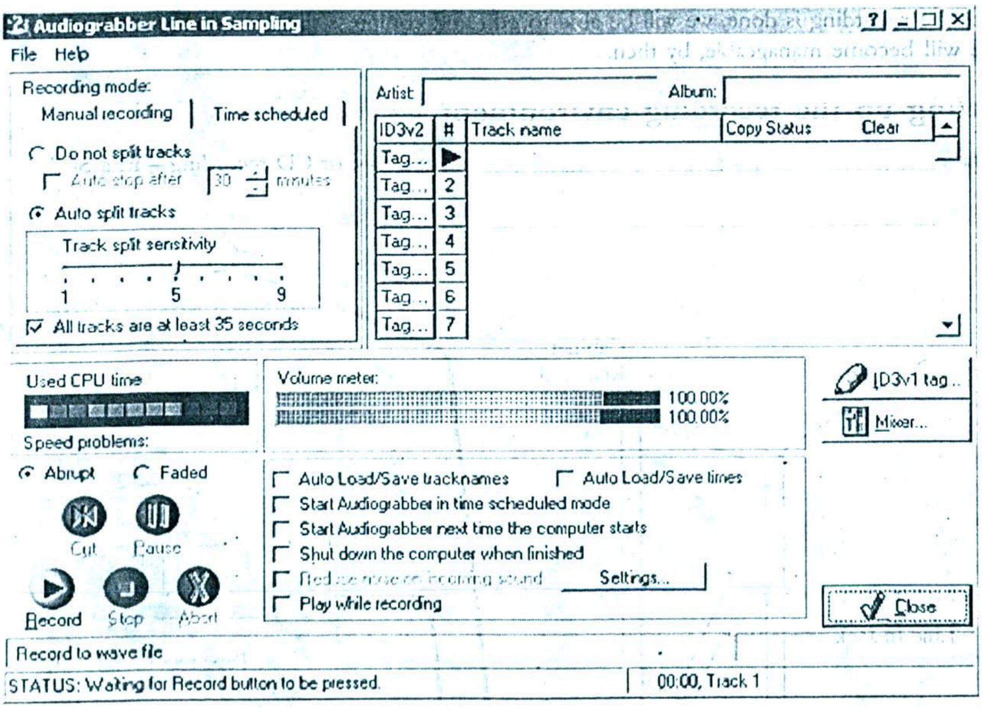


*Figure 32: Audio grabber – ready to record. Look at the volume meter showing incoming signals.*

4. Click on the record button and start recording.
5. Start speaking in front of the microphone. If the sound source is your tape recorder, switch on the play button.

6. If proper recording is happening, then you should see some movements in the volume meter. You may want to do a test piece and playback the same – before doing the actual recording.
7. To stop the recording process, click on the stop button.
8. Save the file and play back, using one of the media player programs.

## 7.12 Recording sounds with windows sound recorder

As mentioned earlier, windows operating system comes up with a neat utility program for recording sounds via a microphone – called the sound recorder. Apart from recording sounds, this utility can also open, edit and playback any .wav file.

To view the variety of recording options that are available, click on the audio properties in the sound recorder's edit menu. The audio properties window that pops up, shows both recording as well as playback environment settings.

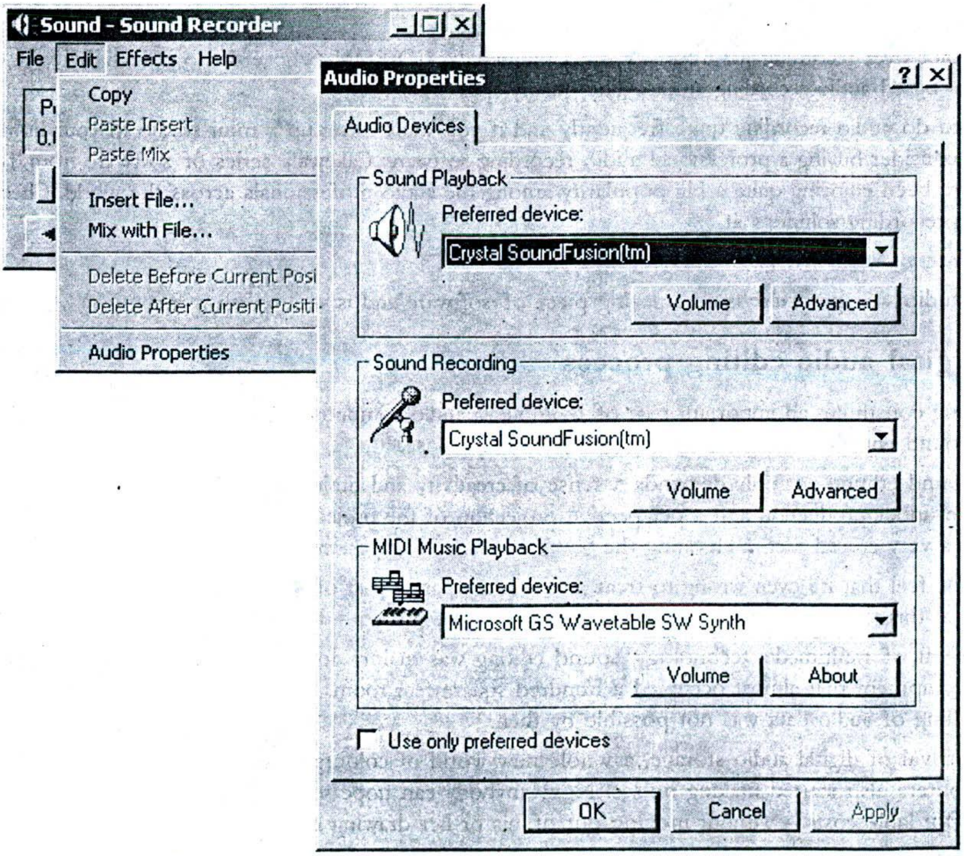


*Figure 33: Audio properties in windows sound recorder*

Once the options are selected, one can begin the recording process.

## 7.13 Professional multi-track recording

What we have been talking about – all along – is a simple home recording process with Mono and Stereo channels. But for slightly more serious business and professional purposes, we need more sophisticated recording applications.

For example, you may want to record Human voice in one track and mix it with some other pre-recorded music in another track. Or, while recording a music performance – you may want to record different instruments in different tracks and control them at the instrument level.

All these things and more - are possible, with professional recording software and studio-based recording.

If you are doing a one-time recording, then it is best to approach professional recording studios. These studios charge their clients on per-hour basis and the charges are nominal. Many Television and radio advertisements are recorded here – You will be able to find one in all cities of India wherein there is some base for entertainment industry.

Recording studios usually make use of DAT tape (Digital Audio Tape) for recording. Of late, CD Audio based recording is becoming more popular and the studio you approach is most likely to have this option.

When recordings are done in studios, highest quality of audio can be expected – with very little or no disturbances. You can also control the volume & treble/bass of each instrument to the best possible combination – and even edit the same, without disturbing other instrument notes.

I did the voice-over recording for Tourist's India multimedia CD-ROM (whose demo is included with this book), in a chennai based audio recording studio. The quality speaks for itself.

If you are to do audio recording quite frequently and if you want to set-up a mini studio in your home or office itself, then consider buying a professional audio recording software. Cakewalk series of software from Twelve Tone systems have been enjoying quite a big popularity among the audio professionals across the world. Check out their latest home recording solutions at

*http://www.cakewalk.com/Products/HomeRecording.asp.*

Cakewalk studio – in particular is an amazing piece of software and is well worth a try.

## 7.14 Digital audio editing process

Sound editing constitutes an important part of multimedia audio computing expertise. Literally, in every multimedia project, a sound engineer is exclusively engaged for this purpose.

Effective sound editing not only demands a sense of creativity and timing, but also a good understanding of the audio theme under production and a perspective projection of the project - as a whole. The background audio or music, play a very crucial part in elevating the '*combined multimedia experience*'.

In fact, some feel that it's even wrong to treat audio as a separate part of the project and that it should be treated as the central thread that maintains ambience and aura of the project – throughout its development.

Till the advent of multimedia technology, sound editing was mainly done at audio recording studios, with very expensive equipment that almost occupied a hundred square feet room. The scope of editing was also limited, as digital handling of audio data was not possible by then.

After the arrival of digital audio storage, a whole new world of colours opened up for sound enthusiasts - and soon, computers also started making noises. Now, anybody can hope to build up a decent digital sound effects studio at affordable costs – without moving out of his or her drawing rooms.

All thanks to multimedia audio technology!

A question that comes naturally raises at this point is:

Why do we need to edit sounds at all?

## 7.15 Need for audio editing

The dire need for editing audio, arises out of one or more of the following reasons, listed below:

1. We need to cut out unwanted bits and pieces of audio, from the original recording. Disturbances almost certainly creep in, during the beginning and at the end of recording sessions.

2. We may want to reduce the length of the audio recording, to suit the required time gap. Controlling the audio time duration is extremely difficult with the original sound sources. Often, the recording is either a bit short or a bit long! Seldom do we find a sound piece that perfectly matches the required timings. This factor becomes particularly important in presentation shows and animated movies.
3. We may have to enhance audio quality / volume or chillness in certain places.
4. We may have to take care of mistakes and errors that have been committed and not noticed during the original recording session.
5. We may wish to add some special sound effects, which could not be added during recording.

All these are, but a tip of the iceberg...a lot more problems surface during the actual recording process - making audio editing, almost an inevitable exercise.

## 7.16 Audio editing terminology

In a typical Audio Editing environment, the following operations are frequently performed:

- **Trimming:** Every recording is usually associated with a blank space for a few seconds before and / or after the required sound. The process of removing these blank spaces is called trimming. Removal of blank spaces, apart from reducing the audio file size considerably, enable perfect time matching and further editing operations easy. Trimming is usually done, by cutting out the recorded piece - using the 'Cut-File' feature in audio editing programs.
- **Splicing:** Splicing process refers to removal of unwanted sounds that have crept in, during the recordings. This is also accomplished by means of 'Cut-file' feature in editing programs.
- **Reassembling:** One may want to assemble several stray pieces of audio together, to make up a single file. This process, accomplished by 'cutting', 'copying', and 'pasting' multiple sound clips, is called reassembling.
- **Volume control:** Increasing or decreasing the volume, of either portions, or the whole of recorded audio clippings, is called volume control. This becomes particularly important during reassembling operation; as different sounds pieces may have been recorded at different volume levels, they all have to be neutralised, to the same level of volume - so that the resulting file sounds as an integral audio piece.
- **Time stretching:** We have already seen the importance of matching audio with time slice available. All good editing programs give us an option to either stretch or contract the audio clipping, to suit the particular time slot. This feature can be expected in some of the digital video editing software as well, wherein, a selected sound sample can be matched over a selected sequence of video or animation.
- **Fade ins and fade-outs:** This special feature has been so extensively bundled with almost all sound editing software, that it's no longer considered in the 'special audio effects' category! This is the process of smoothing out the beginning and the end of audio files, for gradual transition effects.
- **Re-sampling:** This is the process of reducing the sound quality from 32 bit recordings to 16 bit or from 16 bit to 8 bit - thus reducing the quality of the recorded audio – slightly and eventually reducing the ultimate file size - substantially.

The need for re-sampling usually arises, when the expected file sizes are much larger than that was originally expected. Re-sampling, in general, does not result in severe degradation of audio quality; in most cases, the changes are so thin that they are hardly noticeable. All speech recordings or voice over - can be conveniently re-sampled to 8 Bits, without sacrificing quality.

## 7.17 Basic audio editing with windows sound recorder

The windows sound recorder program introduced earlier can be used to perform some preliminary audio editing functions. All the functions are available under the effects menu.

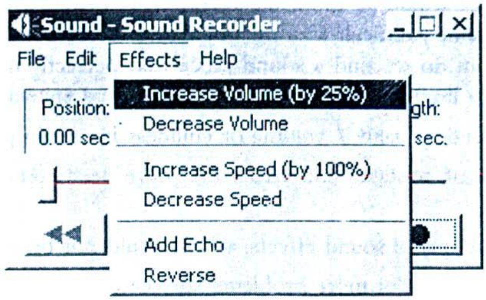


*Figure 34: Basic audio editing with windows sound recorder*

- **Volume control:** Increase / decrease volume - Makes the volume of the source file, to increase or decrease.
- **Playback speed:** Increase / decrease speed - Makes the playback speed of the source file, to increase or decrease. (If you have recorded some voice other than your own, just apply this effect and have some real fun!)
- **Special effects:** Add echo - Adds a mild echo effect to the audio signals. Reverse - Reverses the playback of the source file. Mix - Mix two sound files, one over the other.

You can play around with windows sound recorder's editing options to wet your hands with basic audio editing – before delving deep into more serious editing programs. And I can promise you that this experience can be quite a bit of fun!

## 7.18 Advanced audio editing with cool edit

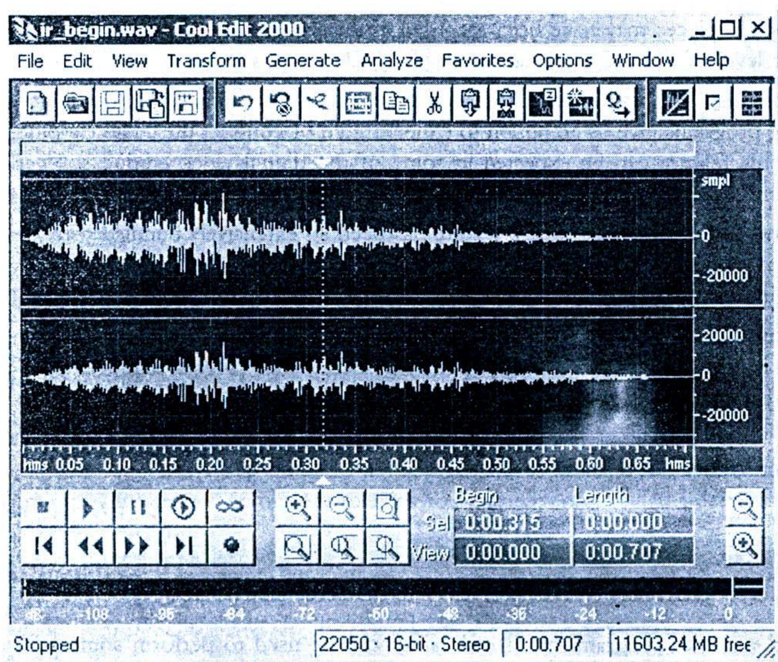


*Figure 35: A simple audio file waveform – as displayed in cool edit*

'CoolEdit' from Syntrillium Software Corporation (http://www.cooledit.com) is one of the most advanced audio editing software you can find in the market. By exploring Cool edit, we will get an exposure to the world of audio editing and recording professionals, techniques and tactics they use and possibly we may also apply some of these capabilities in our own projects.

I have tried to present the topic in such a way that we are not dealing with features that are very specific to this software. Most of the capabilities we will be discussing here should be available in most – if not all – professional audio editing software in the market today.

## 7.19 Cool edit - core features

Cool edit's audio features can be grouped under the following categories:

1. Audio playback – playback digital audio files of various file formats
2. Audio recording – record various types of audio.
3. Audio analysis – perform various kinds of analysis over the recorded audio
4. Audio editing – cut, paste, delete audio data
5. Audio transformation – add special effects to audio files
6. Audio file formats conversion – convert wav to mp3 etc.

Let us take a brief preview of each of this functionality.

## 7.20 Audio playback

Though cool edit can play a very wide range of file formats like .wav, .au, .mp3. .aif, .pcm, .snd etc., it is best not to make use of this software for playback.

Because, when you try to open an audio file with cool edit, it first tries to draw up the waveform of the audio file – which can take up a very long time. This is absolutely unnecessary for simple audio playback – which can be done without much delay by windows media player or quick time player.

Hence, it is best to use cool edit – only for professional recording and editing and not for simple tasks like playback.

You should keep this in mind – while installing cool edit, because there will be a dialog box to associate all the audio file formats with cool edit program. If you do not want to use cool editing for audio playback, remember to uncheck all common file formats like .wav and .mp3 from associating themselves with the program.

## 7.21 Audio recording

Audio recording is simple and straightforward.

Just play the desired audio file or bring in audio from external sound sources using line-in jack. Simply press the record button to begin the recording process.

## 7.22 Audio analysis

There is a menu item called analysis – within the cool edit main panel, under which the frequency analysis tool is located. When you click on the option, cool edit does a clean scan of the entire audio file, does a complex Fourier transformation calculations – and draws up a frequency analysis graph of the left and right channels that make up the audio file.

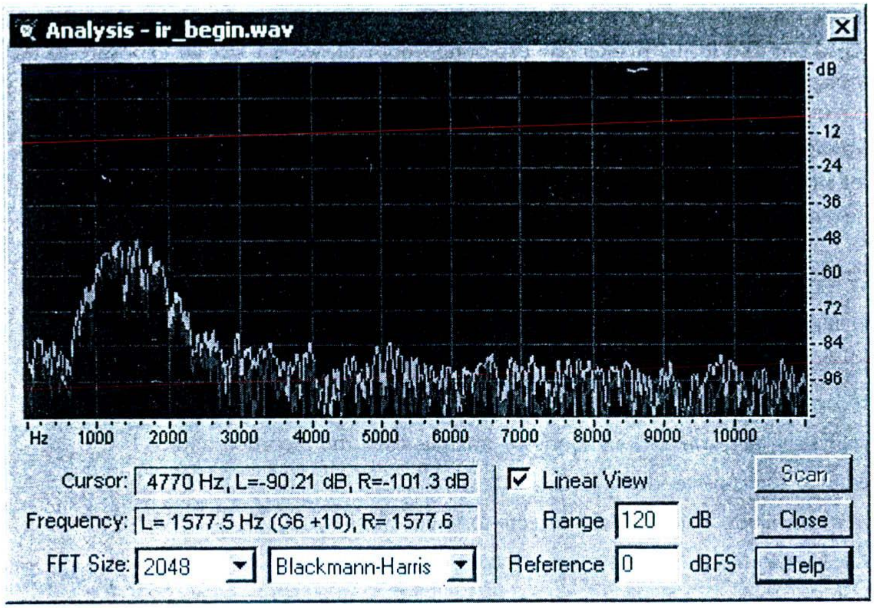


*Figure 36: Audio frequency analysis – as done by cool edit*

This graph is useful for audio professionals – to analyse the overall frequency modulations and decibel changes in the audio file and do the necessary transformations.

## 7.23 Audio editing

Audio editing capabilities constitute the core feature of cool edit.

With cool edit we can delete, cut, copy and paste audio files at our own will and wish – just like editing text in MS Word!

More than that, by assembling various audio clippings from different audio file sources, we can actually create entirely new audio files of our own (*hopefully without violating copyrights*)!

All the editing options are available under cool edit's Edit menu.

Let us see how to do some basic editing operations. The process goes like this:

1. Open the audio file you want to edit. The file opens up with a graphic display of the underlying wave pattern.
2. Playback the audio clips several times and decide the portions to be edited. It is better to note down the time period of the clips, which you want to cut or copy - like, from 11th second to 21st Second.
3. Select the portions you want to edit, with the mouse. The time period note helps you to select the exact portions you want to edit.
4. 'Cut' or 'Copy' the selected portions, with the 'Edit' menu options. The portions of the selected audio file are copied to the windows clipboard.
5. 'Paste' the audio clips in the right place in the right file.

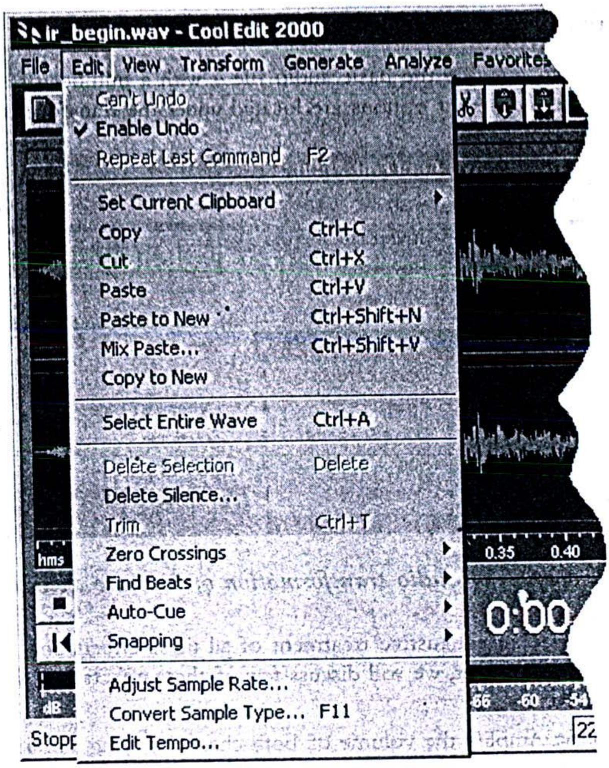


*Figure 37: Audio editing features available with cool edit*

6. There is an option called Mix paste. This is interesting – it helps us to blend 2 different wave files into one! For example, if you want to add light background music to your human voice over, you can copy the music clip and then mix paste the stuff over voice over. This mix paste options are illustrated in the figure. You can have complete control over the volume of the clipping that is mixed into the original audio; this is particularly useful for providing background music.

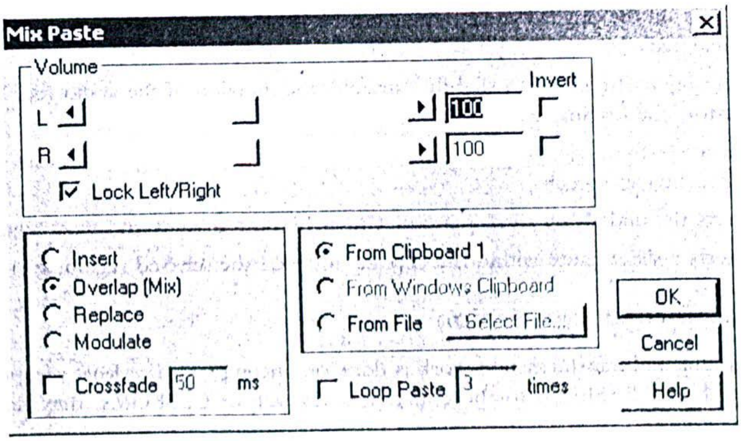


*Figure 38: Mix paste dialog box*

## 7.24 Audio transformation

Apart from editing, we can also *transform* the basic audio file – with a number of special effect options that are available with cool edit. All these different options are located under the transform menu.

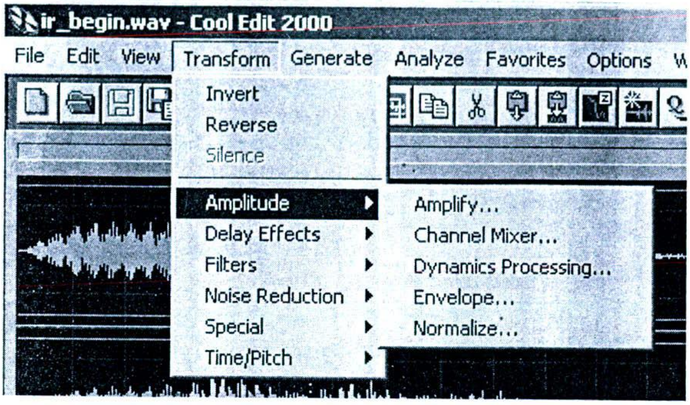


*Figure 39: Different types of audio transformation options are available with cool edit.*

It may not be possible to provide an exhaustive treatment of all these effects and how to make the best use of them – within this small chapter. However, we will discuss few of these effects below – for wetting our hands with audio transformation techniques.

- **Amplitude → Amplify:** Amplify the volume of both channels. Using amplify, we can also provide fade in and fade out effects i.e. increase or decrease the amplitude of the sound gradually. Time of fading in or out becomes important and it's better to apply this effect after making the selections and not to the entire file.
- **Amplitude → Channel mixer:** Control the volume of left and right channels.
- **Delay effects → Delay:** Introduce a small delay between the channels. Should be used with care
- **Delay → echo:** Adds an echo effect to the file. The results can be perceived only after applying the filter and cannot be imagined in advance. Depending upon the underlying wave pattern, this effect can be very interesting or even be disappointing, at times.
- Noise reduction → Remove the background noises and tape hissing sound. Sometimes it does have an unwanted effect on the original audio – may not be very successful, all the time. Should be used after trials on the original audio
- Stretch → A very useful effect to slightly "stretch" the duration of the audio file. Excessive stretching will, however, distort the original waveform!
- **Reverse:** Reverses the underlying wave pattern. Unusual and funny patterns can be obtained, particularly with respect to human speech.
- **Invert:** Inverts the underlying wave pattern...Crests become troughs and vice versa.
- **Silence:** Inserts a silent pause within the clip. i.e. it makes the selected region, silent

## 7.25 Audio file format conversions

Most of the audio editing and transformation work is done on uncompressed or least compressed audio file formats like wav. But, before being distributed to the consumers via web or CD-ROMs, they have to be compressed – because of space constraints.

After the advent of mp3 and its dramatic size reduction - for the given audio quality, conversion from various conventional file formats (like wav) to mp3 has become very common. Mp3 comes in several flavours and can be tuned for better quality or better size.

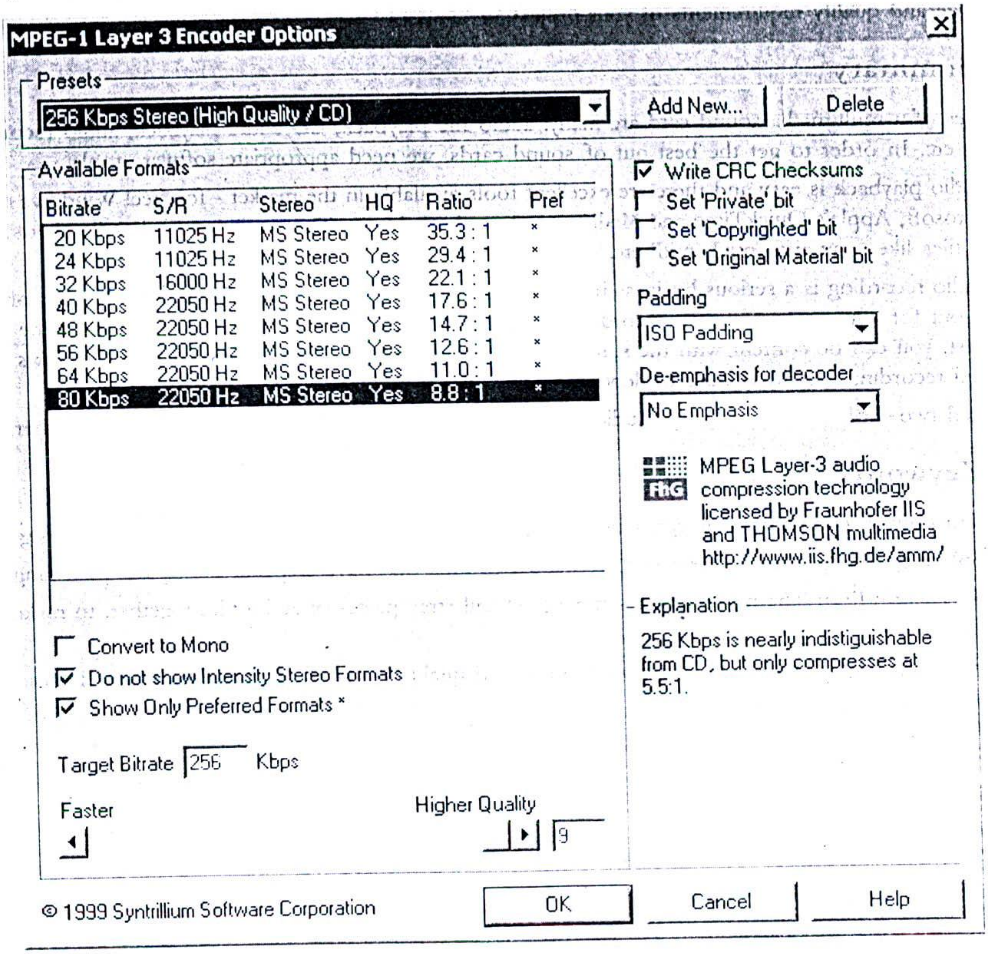


*Figure 40: mp3 encoding options with cool edit*

Cool edit provides options to convert between various file formats – including wav, au, aif, mp3, pcm, snd and what not?

Here, we will just focus on the mp3 file format conversion options of cool edit.

The format conversion to mp3 is sometimes called mp3 encoding – since it is not a simple file format conversion, but actually a complex compression and encoding process. There are quite a number of encoders available in the market today – and their quality also varies.

The mp3 encoding offered by cool edit is one of the best I have seen in the industry: it adopts the compression algorithm patented by Fraunhofer IIS and Thomson Multimedia (http://www.iis.fhg.de/amm/). But the hitch is that this is not bundled with cool edit core application – it has to be purchased separately.

Once you try to save a file as mp3, you will get various conversion options in the dialog box that pops up. You can choose from a wide range of pre-sets available. By pre-set, we mean a specific combination of bit rate – sample rate –channels – and quality of the resulting audio file.

Higher bit rates result in bigger file size and better quality. The pre-set you choose should be determined by the specific size and quality requirements of your project.

## 7.26 Summary

Capabilities of a multimedia sound card are many: audio file playback, CD audio playback, audio recording, MIDI recording etc. In order to get the best out of sound cards, we need appropriate software tools.

Digital audio playback is easy and there are excellent tools available in the market - for free! Windows media player from Microsoft, Apple's QuickTime and Media one player lead the lot. These utilities can handle almost all varieties of audio files like .wav,.aiff, .mp3, midi etc.

Digital audio recording is a serious business if you want to do a good job. There are quite a number of parameters to watch out for - like mono/ stereo recording, sampling rate of recording, sampling size etc. If the recording is a short test, you can be content with the sound recorder program that comes bundled with Windows OS, but for meaningful recordings you may have to depend on third party utilities.

We covered two tools in this category: Audio grabber on the lower end and Cool Edit on the higher end.

## 7.27 Keywords

- **Trimming:** The process of removing the blank spaces before and / or after the required signals.
- **Splicing:** The process of removing unwanted sounds that have crept in, during the recordings.
- **Re-assembling:** The process of assembling several stray pieces of audio files together, to make up a single file.
- **Re-sampling:** The process of reducing the sound quality from 32 Bit recordings to 16 Bit or from 16 Bit to 8 Bit.

# Chapter 8

# The MP3 Revolution

## CHAPTER OVERVIEW

This small chapter is exclusively meant to discuss MPEG Audio layer III technology – more popularly known as mp3.

We will be looking at the core concepts behind mp3 – how it achieves such a stupendous rate of compression, what is the sacrifice in quality we make, how mp3 will shape up in the future and so on.

One of the fundamental concepts behind mp3 is the acoustic capability and limitations of a human ear. The chapter also introduces, rather briefly, this field of study called psychoacoustics.

## 8.1 Introduction

In the previous chapter, we saw how sounds can be recorded as digital audio files using a sound card and relevant software. We also saw that audio files have to be compressed using some audio compression algorithms before being stored into the computer – because of massive memory requirements.

MPEG Audio layer III or simply mp3 is perhaps the most notable technology that has changed the landscape of digital music and the Internet music forever. The impact of mp3 is so significant that its influence is felt above and beyond the world of computers: In digital audio players, disc formats, recorders and a whole range of other electronic equipments.

This chapter covers some grounds related to mp3. We will begin with the invention of mp3 and its popularity with the growth of Internet.

## 8.2 The Birth of mp3

The Fraunhofer Institute, Denmark (http://www.iis.fraunhofer.de) invented mp3. A fruit of nearly twenty years of research work by Dr.Karlheinz Brandenburg and his associates, mp3 has grown steadily from being a research thesis to becoming almost a standard in the world of digital music.

Though the initial work had started in early 80s, mp3 started to look serious only in 1986 and efforts to make it an audio compression standard started in 1988. In 1994, the first mp3 decoder for windows had appeared. But mp3 had to wait for bit more time – almost till 1998-99, before becoming a global phenomenon.

The growth of Internet and the need for distributing songs and digital music over the web had a lot to do with the popularity of mp3. In April 1999, the string "mp3" became the string most widely searched for – in all search engines. From researcher's white papers, it had become an industry standard!

## 8.3 Need for audio encoding

Uncompressed audio files are very huge in size and are not easy to handle.

A three-minute song, recorded at a sampling rate of 44.1kHz and a sampling size of 16 bits with two stereo channels will consume around 30 MB of storage space. If we were to download this file from the Internet - at a comfortable access speed of around 64 kbps, it would still take nearly an hour to download! Even if the audio file were compressed using one of the conventional audio compression algorithms, its size would still be substantially large, making it impractical to distribute over the net.

Enter mp3!

Using mp3 compression algorithms, the same audio file at 44.1kHz, 16 bits and 2 channels can be compressed to roughly 3 MB! Or roughly one tenth of its original size... with little or no loss of quality!

A 3 MB file is not very difficult to download even if the speed of net access is pretty low.

So, how mp3 achieves such a high degree of compression rate with relatively no noticeable loss of quality? The secret lies somewhat in the acoustic capabilities of a human ear and its limitations. This specialised field of study is called psychoacoustics.

Mp3 achieves its very impressive audio compression by making the best use of results available from psycho acoustic study. The key concept is in finding out what are all the sounds that are audible to human ears at any point of time, in a given song. If a particular instrument sound is barely audible or not audible at all, why storing it in the file?

## 8.4 Psychoacoustics

The term psychoacoustics stands for the study on the characteristics of the human auditory system. Two concepts of psychoacoustics are particularly important in our context.

They are:

- ❑ Hearing perception
- ❑ Masking

How sensitive our ears are, to a particular range of frequency - is what is called the hearing perception. The sensitivity of the human auditory system is very good for frequencies between 2.5 kHz and 5 kHz. Our ears are less sensitive to anything above or below these frequencies. The threshold in quiet represents the audio sensitivity. Any tone, above or below this threshold will not be perceived.

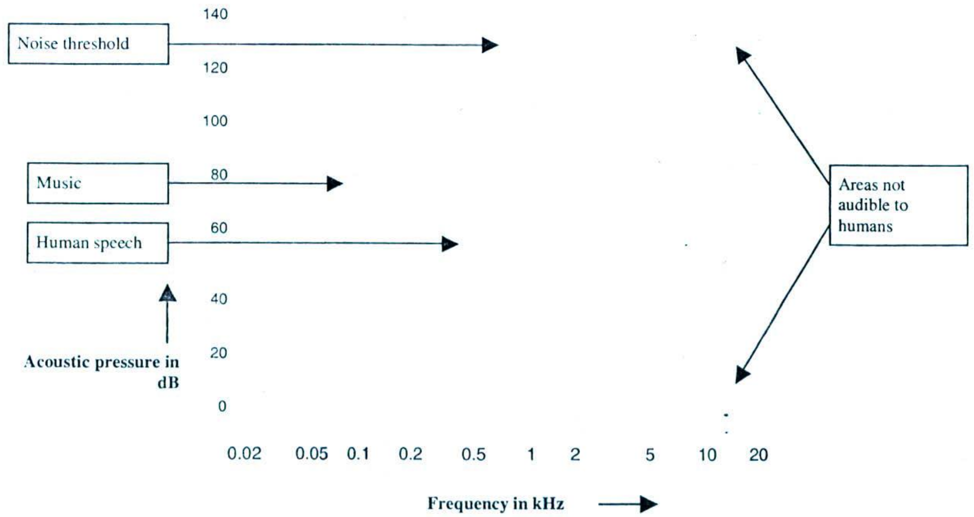


*Figure 41: Hearing perception of a human ear*

From the perspective of audio encoding, signals that are not audible to the human ears can be removed. This reduces the overall size of the file.

The second concept is masking. Masking is the phenomenon by which a short, loud sound covers or masks the quieter, longer sound of the strings for a moment. For every tone in the audio signal, a masking threshold can be calculated. If another tone lies below this masking threshold, it will be masked by the louder tone. Thus, it remains inaudible to the listener.

Again from the perspective of audio encoding, these inaudible signals can be removed.

There is another phenomenon associated with masking. It is called pre-masking and post-masking.

Human brain needs certain fraction of time to differentiate between two different sounds of varying loudness. For example, if a loud sound follows a quiet sound – it takes 20 milliseconds for the brain to perceive the loud sound. This is what is called pre-masking. Similarly, it takes nearly 200 milliseconds for the brain to identify a quiet sound that follows a loud sound!

During these very short periods, we may as well continue with the earlier sound or have no sound signal at all! The user will not perceive it.

The quality of mp3 encoding will very much depend upon the fineness with which it takes care of all these psychoacoustics models and phenomena. Not all mp3 encoders take care of all these factors and hence the difference in quality between two commercial mp3 encoders.

## 8.5 Mp3 encoders

An mp3 encoder is a piece of software that can convert digital audio files in other formats to mp3 files. Encoders make use of psychoacoustics phenomena to find our what are all the unnecessary portions that can be removed from the original file, without being noticed by the user – thus trimming down the overall size of the file. It then uses Huffmann encoding algorithm to further compress this data – cleansed from unwanted frequencies.

Not all encoders take care of all the different factors associated with psychoacoustics; hence the quality of mp3 encoding vastly differs from one encoder to another. This can be easily tested by encoding the same song with two different encoders – while maintaining the process parameters like bit rate, compression ratio and channels. Try to use a digital music file with human voice in it – to perceive the differences very clearly.

Mp3 encoders are available in two flavours: They can either be purchased as a stand-alone program or as a plug-in to one of the commercial audio editing programs available in the market. Go to http://www.download.com and search for 'mp3 encoders' – to see a whole list of free and shareware software available for download. Click on the 'use rating' hyperlink to see all the software that received the best user rating. Choose the one that appeals to you most!

One of the best encoders I have tried is from Fraunhofer Institute –the inventors of mp3. To have a look at their audio-video solutions, go to http://www.iis.fraunhofer.de/amm/app/index.html.

## 8.6 The mp3 encoding cycle

It will be interesting to see the process cycle through which a digital audio file such as .wav goes through, from its nascent form to encoded mp3. This cycle is detailed below.

The first step is the creation of signal spectrum – which consists of 576 sub-bands – by passing the audio signal through a filter bank. This process requires complex filters and makes use of an algorithm called discrete cosine transformation. In real time, it calculates all the unnecessary frequencies and eliminates them iteratively.

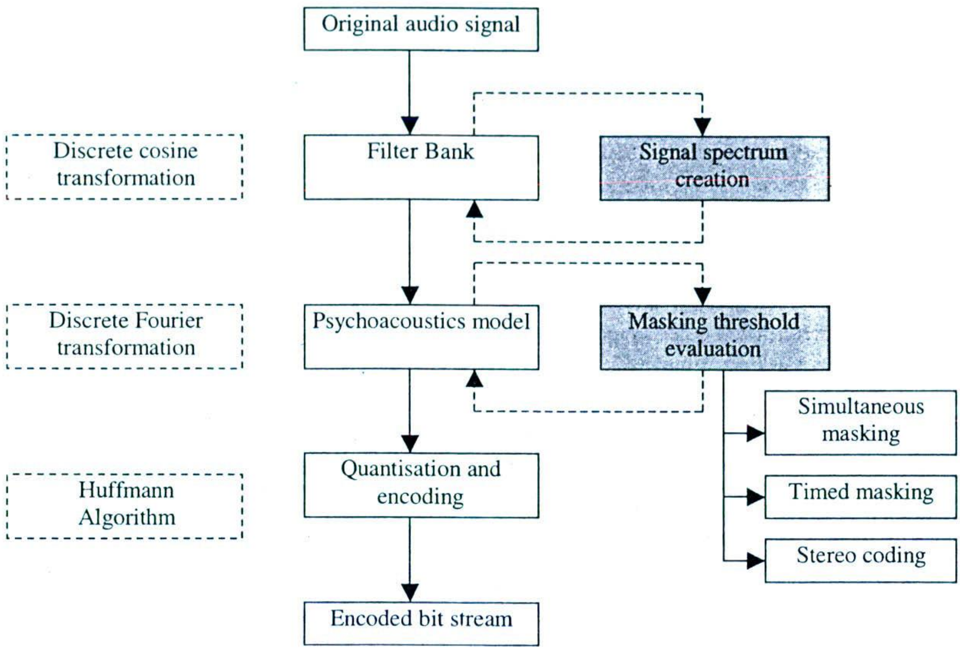


*Figure 42: Hearing perception of a human ear*

The second step is to pass the filtered signal spectrum through the psychoacoustics model built within the encoder. Here, the model calculates the masking threshold for each of the sub-bands in the entire spectrum and decides what are all the frequencies which are to be masked. The process involves another algorithm called discrete Fourier transformation. The stereo effects are also transformed by means of a process called joint stereo coding.

Quantization process results in further data reduction, followed by Huffmann encoding for compressing the resulting file. The latter can reduce the file size up to twenty percent or more without losses.

The last step is the formation of bit stream, which is recorded as an mp3 file.

## 8.7 Encoding parameters

There are three important parameters that you can control, during the mp3 encoding process. They are:

- Bit rate
- Sampling rate
- Compression ratio
- No of channels

Bit rate denotes the average number of bits that one second of audio data will consume. The usually units here are kbps, which is kilobits per second, or 1000 bits/s. Kbps is also useful to match the encoding quality with the downloading speed of the Internet. For example, if you want one-second duration of the song to be downloaded per second at the rate of 64 kbps Internet access – then you should use 64 kbps as your bit rate.

The following table provides you with an indicator to the combinations you could use and the resulting quality:

*Table 15: Different combination of parameters in mp3 encoding*

| Bit rate (kbps) | Bandwidth (kHz) | Channels | Reduction ratio | Quality |
|---|---|---|---|---|
| 8 | 2.5 | Mono | 96:1 | Telephone quality |
| 16 | 4.5 | Mono | 48:1 Short wave or better | Radio quality – |
| 32 | 7.5 | Mono | 24:1 | Radio quality – AM or better |
| 56 to 64 | 11 | Stereo | 26:1 or 24:1 | Radio quality – FM or better |
| 96 | 15 | Stereo | 16:1 | Near CD quality |
| 112 to 128 | >15 | Stereo | 14:1 or 12:1 | CD Quality |

The settings I mostly use for my recordings are 128 kbps bit rate at best quality (which means compression ratio is less), stereo and 22.05 kHz. This is denoted as the CD quality music in the above table.

## 8.8 Summary

Mp3 stands for MPEG Audio layer III. It is an advanced compression algorithm for compressing digital audio files. A compression ratio of 1:10 or even more can be achieved with mp3.

Psychoacoustics or the study of acoustic capabilities of human ear forms the basis for mp3 compression. Signals that cannot be heard or are suppressed by louder signals are identified and eliminated in mp3. Mp3 encoders are commercial software tools or libraries that perform mp3 compression over audio files. The mp3 encoding cycle is a complex process involving advanced mathematical algorithms.

Parameters involved with mp3 compression are bit rate, compression ratio, signalling frequency and channel information.

## 8.9 Keywords

- **Psychoacoustics:** Study on the characteristics of the human auditory system.
- **Mp3 encoding:** The process of compressing digital audio data using mpeg audio layer III algorithms.
- **Bit rate:** The average number of bits that one second of audio data will consume.

# Chapter 9

# MIDI Fundamemtals

## CHAPTER OVERVIEW

MIDI or Musical instrument digital interface is conceptually different from digital audio. It is a refined form of music storage and reproduction.

This chapter takes you through the world of MIDI, covering how MIDI stores music information and what differentiates MIDI from other forms of digital audio. The chapter then explores Microsoft's general MIDI standards, the de-facto standard for MIDI technology. It also touches the device capabilities of MIDI in a windows environment.

## 9.1 Introduction

Digital audio files, though are being extensively used in many multimedia audio applications, have their own limitations - the major one being their exorbitant file sizes.

A small dash sound that lasts for less than a second takes up around 25 KB and a sound file that runs just for around 12 seconds comfortably occupies more than 2.5 MB of precious hard disk space! For example, the audio file 'chord.wav' - located under windows media directory (usually c:\windows\media) lasts for one second or so, but its size is 95 KB!

Such massive file sizes pose serious issues, when music has to be played on for hours together, in a typical project. This necessitated the need for inventing alternative forms of music storage and distribution and audio research labs across the globe were investing their time and energy for a breakthrough.

In the meantime, the electronic musical keyboard giants like Casio and Yamaha were trying to evolve some standards for communicating the keyboard and synthesize signals electronically, across several disparate platforms.

The end result was the concept of MIDI.

Multimedia audio pioneers like the creative labs and Ad Lib wasted no time in managing to adopt these standards intelligently into their audio cards.

Today, computers can straight away talk to any MIDI compatible electronic keyboards – without a hitch!

Over the period of time, MIDI technology has never looked back and managed to carve out it's own enclave in the multimedia market segment. MIDI has the ability to reproduce fine / noise-free electronic music and enjoys astonishingly smaller file sizes.

## 9.2 The concept of MIDI

The concept of MIDI goes like this:

Instead of storing the actual sound samples in the form of analog signals, a typical MIDI file just records the description of the ongoing music. This description includes the start of a note, its pitch, length, volume and other musical attributes such as vibrato.

Details like the different musical notes that are being played at any point of time – by a particular instrument, their sequence, their time duration etc. are noted down in the form of numeric data. Capturing this information for all instruments that are being played at any point of time is nothing but digitally creating the comprehensive music notes of the composer.

During playback, these notations are read and understood by MIDI compatible devices such as sound cards and electronic keyboards, which then refer to a database of pre-recorded sound samples of various instruments and play the appropriate note.

These notations follow international standards, so that the music information can be exchanged between computers and other musical instruments such as synthesizers! The encoded instructions transmitted between such MIDI devices are called MIDI messages.

Depending upon the quality and range of presets available, this synthesized music can get very close to the original ones. For example, try to play Beethoven's 5th Symphony MIDI file that is installed under windows media directory – and you will be amazed at the level of precision and quality.

MIDI is capable of recording more than 16 tracks of different notes from various instruments, simultaneously. It is also possible to edit a particular track's volume, tempo, notes etc. without affecting the rest. This gives unprecedented flexibility in editing the recorded music.

MIDI data can also be understood by a host of several other electronic instruments and keyboards – apart from computers. This results in seamless flow of audio data across a wide variety of synthesizers, keyboards and multimedia devices, facilitating integration and synchronization.

In olden days, a concept called FM synthesis was used to generate the music notes from various instruments. But, of late, a better technology called Wavetable synthesis has replaced FM synthesis in almost all devices.

The biggest limitation of MIDI is that it can record only music data and not just any form of audio signals!

## 9.3 Comparing MIDI with digital audio

It could be recalled that digital audio files store all kinds of audio data in the form of numerical samples that are converted back to audio waves during playback. A whole lot of ADC and DAC conversions are involved in this transformation cycles.

On the other hand, MIDI stores just music data and records data as musical notes and instruments.

Both techniques are widely used in multimedia audio applications and both have their own inherent blessings and limitations.

*Table 16: Differentiating MIDI from digital audio*

| Factor | Digital audio | MIDI |
|---|---|---|
| File size | Huge | Very less |
| Quality of music and disturbances | Varies. Susceptible to noises noises or disturbances | Mostly stable. No |
| Editing and manipulation | Inflexible | Highly flexible |
| RAM and processing power | More | Less required |
| Playback | Highly consistent | Inconsistent |
| Audio | All kinds of sounds | Only music |
| Knowledge of music | Required, but not compulsory | Compulsory |

Let us do a comparative analysis of these two forms of audio data storage, in the light of various factors like file size, processing power require etc. This will help you to choose the right form of audio, for a given purpose at hand.

### 9.3.1 File Sizes

The main attraction of MIDI is it's amazingly low file size. The difference is so significant that it is even unfair to compare it with digital audio.

In some cases, the compression ratio is as high as 1: 1000! This is obviously due to the basic difference in capturing audio data, which was explained earlier.

For example, with a 100 KB digital audio file, you cannot hold more than a few seconds of audio data whereas a MIDI file of the same size may run well over 6 to 10 minutes!

Windows users can check out all the MIDI files located under the windows media directory and make a comparison of various file sizes and their duration.

More often than not, file size becomes a very important factor in most multimedia projects. The limitation is set by the storage capacity of the CD-ROM media, which is around 640 to 700 MB today.

Thus, it is important to remain conscious about the size of various media files involved in the project. It is best to make use of MIDI for all background music, which may have to last for long duration and depend on digital audio for the rest.

Also, for applications involving Internet multimedia, file size becomes a critical constraint because of the downloading time associated - and MIDI offers a reliable solace.

### 9.3.2 Quality of music

Unlike digital audio, MIDI recordings are not susceptible to external disturbances or additional 'hisses' or 'noises'. This is because, all recordings are done electronically and there is absolutely no intervention of external mediums like air - which degrade the quality of the resulting audio files.

Thus, in most of the cases, MIDI music is usually far more superior than it's counterpart.

However, for decent MIDI playback, at least a 16-bit soundboard with wave table synthesis technology is the minimum requirement.

### 9.3.3 Editing and manipulation options

As far as editing operations are concerned, MIDI is far more flexible than digital audio files.

Literally, each and every note of the instruments that are being played back can be controlled. The very pitch of individual instruments- or the music piece as a whole - can be manipulated. The tempo of playback can be altered to suit the required time frame. The volume of every individual instrument can also be altered to one's will and wish.

All these fancy capabilities are not available in digital audio, wherein various music notes often blend and mix with each other as a single stream – unless recorded on separate tracks.

Cutting, copying and pasting the notes of individual instrument notes is also very much possible!

Thus, it is possible to mix up three different instruments and notes from three different recordings, and reconstruct a fourth music file with them! I have done it several times in my projects, with reasonable success rate!

The time length of the recording can be stretched or contracted without much sacrifice in quality.

### 9.3.4 RAM / Processing power required

MIDI file playbacks demand much lesser RAM and processing power, than digital audio files.

This becomes a constraint to be considered in multimedia projects, wherein several things have to be handled simultaneously. For example, you may have to throw the appropriate piece of information, fetch the relevant background music and play it in the background and in the meanwhile - keep yourself ready for any mouse clicks or other interactions from the user.

If relevant RAM or processing power is not available at any point of time, the time taken to respond for an action might be delayed and there is always the fear of machine suspending – which is the last thing you want to happen on a client's machine!

With all these positive aspects in its defence, MIDI does have its own share of limitations too.

The following paragraphs detail the same.

### 9.3.5 MIDI Playback

The main disadvantage of MIDI, which amounts to lot of negative credit - is the high degree of inconsistency associated with its playback.

During playback on a client's machine, the operating system tries to look out for the appropriate instruments or their equivalents and tries to load them on demand. It then reconstructs the MIDI music and plays back. Thus, the quality of MIDI music directly depends upon the underlying sound cards and sound samples available in them.

Unless the user's playback audio platform is similar to or at least comparable to that of the developer's card, the results shall be quite ambiguous.

This problem degenerates further, because of the variations in the 'Instrument tables' across different audio boards that supply the required sample notes to MIDI, during playback.

The end result is that MIDI files do not sound the same across different audio cards.

In case of digital audio, the story is quite different.

As these audio files are simply large chunks of numbers and do not depend upon any Instruments library during playback, the playbacks under a variety of audio platforms are quite consistent.

Even under totally different hardware environments and between audio boards from different manufacturers, digital playback manages to remain the same. Because of this fact, many multimedia CD-ROM producers prefer to use digital audio files instead of MIDI.

In cases where the playback environment is known beforehand, MIDI will obviously stand out as a better choice. Also, with simpler and limited usage of popular Instruments - like piano - during recording, MIDI playbacks maintain tolerable consistency.

### 9.3.6 An experiment to try

Try to play a good MIDI music file in a 64-bit audio card.

Then, try to play the same file using a 32-bit card or even less.

The difference could be minor or very significant – depending upon the number and range of instruments used.

### 9.3.7 Limitations of MIDI audio

Throughout our discussions on MIDI, I've only used the phrase 'MIDI Music' and not MIDI audio.

Because, unlike digital audio files, MIDI can hold only music and instrument data and not all kinds of sounds!

In particular, MIDI is incapable of recording human voice - an element usually associated with most of the multimedia projects. This is because of the fact that human voice is basically transmitted over air and can only be recorded as digital data.

Also, one cannot hope to develop a library of human sound samples and create all possible combinations of voices that can be made use of by MIDI! Only some expensive MIDI boards meant for high end applications are capable of 'emulating' human voice and many times, the results are far from satisfactory.

### 9.3.8 Handling MIDI

Last but not the least, intelligent MIDI manipulations demand some decent understanding of the western or eastern music theory. There are software which even print out MIDI files as pure western music notes...such is the integrity of MIDI files!

The best option available for those who don't have proper music knowledge is to lookout for royalty-free MIDI music banks available in the Internet.

The best practise is to make use of both digital audio as well as MIDI files, in a given project and try to strike a balance between the two.

## 9.4 The general MIDI standards

Microsoft introduced a set of MIDI standards with its windows 3.1 operating system that have ultimately become the industry norms and are now known as 'The General MIDI Standards'. Almost all MIDI hardware and software developers now confine to this, or make their systems compatible with these standards.

MIDI files that confine to Microsoft's standards are called device independent MIDI files i.e. these files have best chances of producing the same kind of music, even under a varying hardware and software configurations that are compatible with general MIDI standards.

Microsoft divides all MIDI compatible devices into two streams:

- ❑ Base level MIDI devices
- ❑ Extended level MIDI devices

Let us explore these two categories, further.

### 9.4.1 Base level MIDI devices

Base level MIDI devices are those that are capable of playing back at least three melodic instrument tracks - with at least six notes being played simultaneously and a percussion track - with at least three notes being played simultaneously.

Melodic instrument tracks are those that contain musical notes - like the piano sounds and the percussion tracks are those that carry the accompanying rhythms, like the drumbeats.

The earliest family of sound cards - particularly those using FM synthesis technology, confined to base level MIDI devices. The ultimate music resulting from these devices were preliminary in nature and are not comparable to that from extended level MIDI devices

### 9.4.2 Extended level MIDI devices

Extended level MIDI devices are those that are capable of playing back at least nine melodic instrument tracks - with at least sixteen notes being played simultaneously, along with a percussion track - with at least sixteen notes being played simultaneously.

Almost all-decent sound cards, these days, can be called as extended level MIDI devices.

## 9.5 Polyphony

Before focussing our attention on the actual channel assignments in general MIDI standards, it's important to understand a concept called polyphony.

Polyphony is the technique of accompanying more than one musical note to be played simultaneously, during recording. For example, the piano alone may produce more than three different notes, simultaneously. In western music school, this is what is called a chord.

So, based on our earlier notations, we can conclude that base level MIDI devices allow 6-note polyphony for melodic instruments and that the extended level MIDI devices allow a lot more – 16 to be precise.

## 9.6 General MIDI channel assignments

We saw earlier that those MIDI files that confine to Microsoft's general MIDI standards are called device independent MIDI files.

These files store two different streams of instruments, for each music piece that's recorded - one for base level MIDI and another for extended level MIDI.

Out of the 16 MIDI channels available, the general MIDI mapping assignments are as follows:

*Table 17: General MIDI mapping assignments*

| Channels | Purpose |
|---|---|
| 1 to 9 | Extended level melodic instruments with 16 note polyphony |
| 10 | Extended level percussion instruments with 16 note polyphony |
| 13 to 15 | Base level melodic instruments with 3 note polyphony |
| 16 | Base level percussion instruments with 3 note polyphony |

When a new piece of MIDI hardware is installed, all these mappings are automatically taken care of.

If the additional hardware is an extended level MIDI device, the base level channels are ignored and if the hardware is base level device, the extended channels are ignored. Usually, pre-developed MIDI maps are supplied along with the installation driver, which informs the operating system regarding the mapping preferences to be looked after, during MIDI playback.

## 9.7 General MIDI instrument assignments

The General MIDI standards also dictate a standardized program change mapping system and a standardized set of Instrument samples.

For example, all general MIDI compatible devices have acoustic grand piano instrument sample matched to number 0, violin matched to number 40 and so on. This also means that the same instrument sample issues out of every other MIDI compatible device, when a call to specific Instrument is made.

The general MIDI Instrument assignments define 128 such Instruments, covering all types of melodic and percussion instruments imaginable.

In order not to make things more complicated, most of the MIDI device vendors adopt the same set of 128 instrument samples and sound effects with their product.

With so much pre arrangement, you should be wondering why we observed earlier that MIDI playback is ambiguous. Remember! The MIDI music is reconstructed from the back of sound samples held by the device. These sound samples will certainly vary from one manufacturer to the other and hence the difference in the ultimate MIDI music.

## 9.8 The preferred MIDI device on your system

To see what is the preferred MIDI device that plays the MIDI music on your system, you can refer to the same sounds and multimedia window in the control panel.

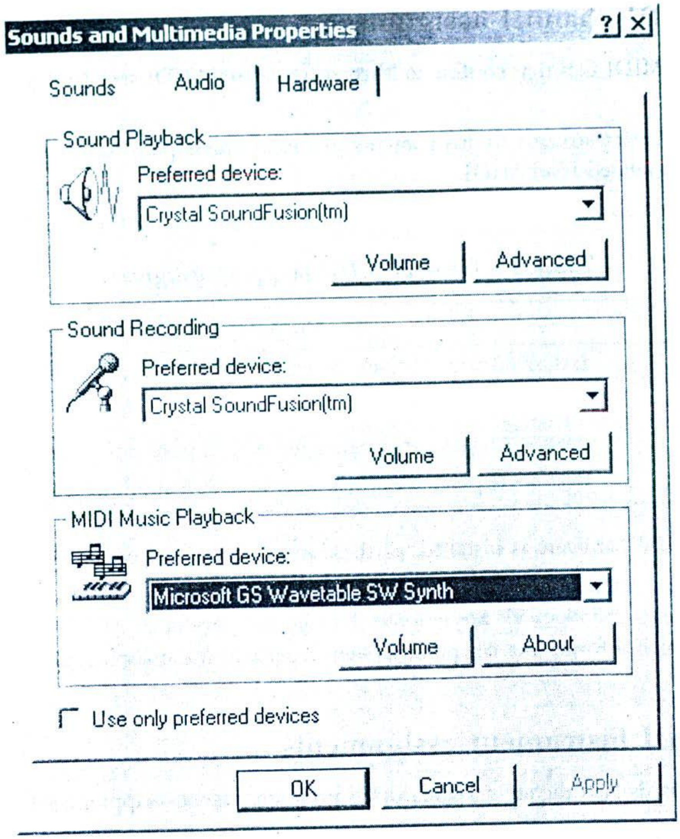


*Figure 43: Preferred MIDI device on your system*

## 9.9 Summary

MIDI files store audio data as environment variables - and not as audio signals issued out of the source.

MIDI and digital audio are characterised by inherent advantages as well as limitations. Both types of audio are generally employed in multimedia projects – MIDI is best suited for background music while digital audio is for voice over and other miscellaneous sounds.

Microsoft's general MIDI standards, defining all instruments, mapping and channels associated with MIDI devices have become industry standards. MIDI files that adhere to these standards are conceptually known as device independent MIDI files.

## 9.10 Keywords

- **Base level MIDI devices:** MIDI devices capable of playing back at least three melodic instrument tracks - with at least six note polyphony and a percussion track - with at least three note polyphony.
- **Extended level MIDI devices:** MIDI devices capable of playing back at least nine melodic instrument tracks - with at least sixteen-note polyphony and a percussion track with at least sixteen-note polyphony.

# Chapter 10

# Working with MIDI

## CHAPTER OVERVIEW

This second chapter on MIDI as well as the last chapter on multimedia audio details how to set up a professional MIDI music studio at home and start recording your own MIDI hit numbers.

MIDI recording demands equipment and expertise. Hence, it makes sense to check out the royalty free MIDI music banks available in the market before getting your hands busy with the MIDI keyboard.

This chapter also covers MIDI editing techniques, using a piece of software utility called the MIDI sequencer.

## 10.1 Introduction

In the previous chapter, we had developed a conceptual understanding about MIDI and saw how MIDI files a different from nsormal digital audio files. In this chapter, we will focus our views on setting up a MIDI recordi environment and editing the same using sequencer software.

While playing back MIDI files can be easily achieved with the help of windows media player or any other multimed player program, recording a MIDI piece is reasonably complex. You need to buy a MIDI compatible music keyboar cables and sequencer software.

That is why it makes sense to make use of ready-made MIDI music against creating one on our own.

## 10.2 MIDI recording vs. MIDI gallery files

Before you get down to record a specific piece of MIDI music for your project, it is worthwhile to check up t royalty free MIDI music libraries that are available in the market and on the Internet. These libraries conta hundreds and thousands of pre recorded MIDI music files, for various themes and occasions. Chances are high t you may find a specific piece that will actually suit your requirements. This will save you lot of time and effort a in most of the cases, the price you pay will be worth it.

Check out the following websites for such free and paid MIDI music banks:

*http://www.musicrobot.com*

*http://www.mididb.com*

*http://msod.formata.org/topdowns.html*

*http://www.ifni.com*

## 10.3 Setting up a MIDI music studio

We shall now discuss how to set up a personal MIDI music studio environment for our own use.

We need to have the following hardware, peripherals and software:

- ❑ A MIDI compatible synthesizer electronic keyboard
- ❑ MIDI plug in cables
- ❑ MIDI sequencer software

The following paragraphs depict what they are and why we need them.

### 10.3.1 A MIDI compatible synthesizer electronic keyboard

Synthesized music is something that's not natural, but emulated. But when you hear a piece from one of the latest models, you feel as though their notes are more natural than the original instruments themselves.

Such is the level of fineness and sophistication modern music keyboards are endowed with.

To record your MIDI music, you need to purchase a MIDI compatible music keyboard. Yamaha, Casio and Roland are some of the names well known in this arena.

Please note that most of the low-end keyboard models may not have MIDI compatibility. Only those at the higher end – which may cost a fortune – are blessed with such advanced capabilities.

One quick check to find out whether a keyboard is compatible with MIDI or not – is to lookout for MIDI output sockets. If there is one, then it is most likely that the keyboard is MIDI compatible. Also, look out for the 'General MIDI compatible' wordings in the information booklet or product brochure.

If you are not a full time musician or if it is a one-off exercise, then it is better to hire a MIDI compatible keyboard from relevant sources like the light music troupes.

High-end keyboards are capable of supporting hundreds of instrument banks – apart from a host of other interesting features.

There are few audio cards that can send MIDI data from sources other than synthesizer keyboards – like the electric guitars. However, this may not be necessary in most of the cases, as you have various lead and base guitar options in the synthesizer keyboard itself.

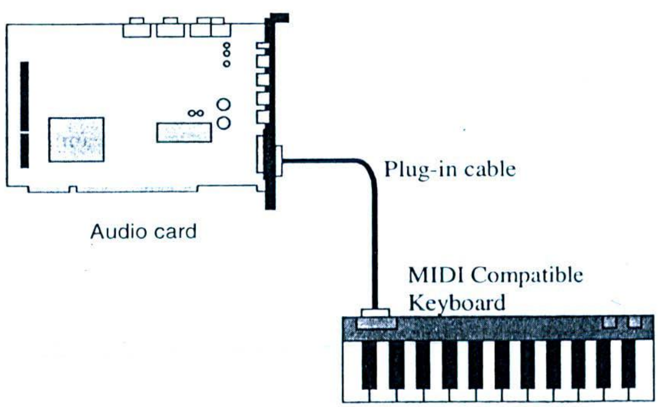


*Figure 44: Connecting a MIDI keyboard to sound card.*

### 10.3.2 MIDI plug- in cables

These cables establish the actual electronic connection between the sound card and the MIDI compatible keyboard. In general, these are not readily bundled with sound card or multimedia kit and have to be purchased separately.

Keyboard's MIDI OUT port must be connected to sound card's MIDI IN socket.

When you restart the computer, windows must be able to identify this new MIDI device and configure the sam for recording.

### 10.3.3 MIDI sequencer software

MIDI sequencer is a piece of software that allows you to open the various instrument tracks in the MIDI file an edit them to the finest detail possible.

Various tracks available with the MIDI file are usually displayed in graphics form and you have the option of editin every single track without disturbing the other!

Some sequencing software even displays the music notes in western notation form. This facility is a very useful featu for professional composers, who can then compose notes for various instruments and tracks, print them out an issue them to instrumentalists directly.

One such MIDI sequencing software that is quite popular is the 'Music sculptor' from Aleph omega software. W will discuss this utility later, in an exclusive section.

## 10.4 Recording MIDI music

- ❑ Keep the electronic keyboard at a convenient place near the computer.
- ❑ Connect the 'MIDI Out' jack of the keyboard to the 'MIDI / JOYSTICK In' connector jack of yo sound card.
- ❑ Open the MIDI sequencer software and start with a 'New File' option in the File menu.
- ❑ Pre-set the synthesizer keyboard with the appropriate instrument selections and rhythm backgrounds.
- ❑ When you are ready for the recording, push the 'Record' button in the sequencer software.
- ❑ Start playing the notes you want to record.
- ❑ When you are done, click the 'Stop' button and stop the recording process.

If you are not fluent enough to play the music keyboard with ease, you may find it very difficult to maintain th right keystrokes, while keeping up the time and rhythm.

If you are a new bee to the world of music, be prepared to do some hectic editing!

All your keystrokes are recorded on the MIDI Music file. Some advanced sequencing software even allow you t record as many as sixteen different channels of data, each containing different instruments!

If you want to undertake such complicated recordings – involving multiple tracks and instruments, you may have t select the channels one after another and play the relevant notes. Alternatively, you can also connect several keyboarc simultaneously and place one musician at every board and start recording all the keystrokes simultaneously.

Once the recording is over, the resulting file can be saved with the 'Save as' Option in the sequencing software. Th files are usually stored with a .mid extension.

## 10.5 Editing MIDI music files

This is perhaps the most difficult part of the exercise!

Once the MIDI recording is over, we need to edit the resulting MIDI Data. Apart from simple editing operation like removing the silence before and after the notes, there are many other operations that can be applied upon th recorder file.

For example, the time duration required for a specific MIDI piece may be a bit lesser or greater than that of th actual recording. In such cases, the files may have to be edited to suit the required time intervals.

MIDI sequencer software allows us to cut, copy and paste relevant pieces of instrument notes – just like Microso Word for text! Since different instrument streams and notes can be cut and manipulated within the same MIDI fil the sequencer software gives full power and freedom to a creative music composer, to experiment with incredible ease and innovation.

Quite a number of MIDI sequencer software is currently available in the market today.

Some are bundled along with the sound cards and the rest are available as separate packages.

Fortunately, a good many number of simple sequencer software are available as shareware and freeware, across the Internet. For understanding the fundamental MIDI editing operations, we will take up one such shareware called "Music Sculptor ".

## 10.6 Music sculptor

Music sculptor has a very simple and easy to use interface.

You can open any valid file with a .mid extension. As the file opens out, a graphical representation of all the tracks available within the file are displayed.

Music sculptor can show the MIDI music files in two modes - Keyboard mode and Track mode. Track mode is usually preferable for all editing operations.

The opened file can be played back with the 'Play' button that protrudes out from the tools box.

As the file is being played from the beginning, a rectangular pointer moves along the time track, which has been divided into equal number of intervals. This pointer is very useful for pin pointing and selecting the required track / sequence along the right time interval, during the editing operation.

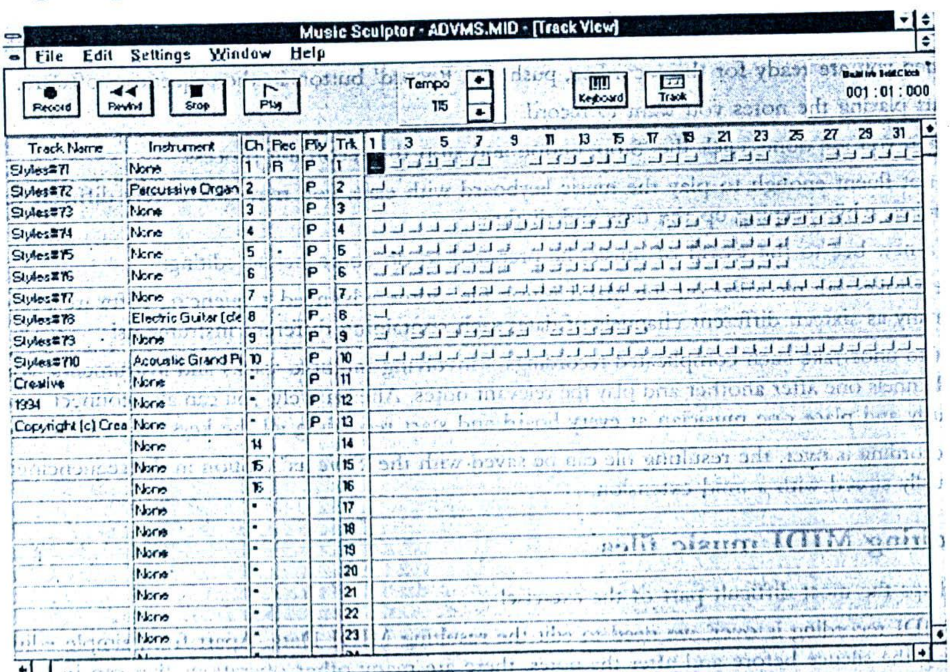


*Figure 45: Music sculptor interface*

After playing back the file several times, you can decide upon the tracks and notes to be edited. If need be, an individual track alone can be selected and played back, to hear the notes of that particular instrument. (In the figure that follows, I have selected three such tracks for editing).

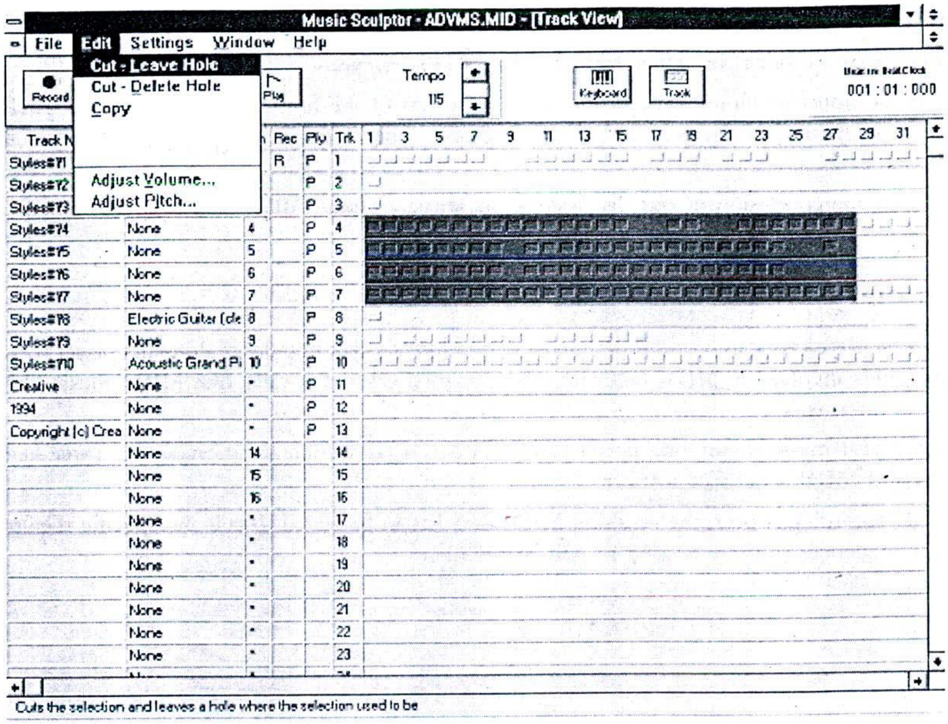


*Figure 46: Editing tracks with music sculptor*

Once the portions to be edited are decided accurately without ambiguity, the actual editing operation starts.

Unwanted portions can be selected and deleted, using the 'Cut' menu option.

In cut, there are two options available: To cut without leaving hole and to cut and leave the holes. Selecting the first option deletes the entire segment, leaving no time gap. The second option deletes the specific notes from the file, but leaves the time gap, as it is. The first choice is suitable, when all the tracks along a particular time interval are selected and removed and the second is meant for selecting and removing notes of specific instruments.

Selecting a portion of the file segment and copying it repeatedly can extend a small MIDI piece to a prolonged duration of time. This feature is particularly useful for creating background music, wherein, some specific notes are to be repeated over a period of time.

MIDI integrity is so powerful, that the edited pieces seamlessly blend with the parent piece and are hardly noticeable. Thus, even professional music performances can be created with the help of MIDI and with little effort.

## 10.7 Pitch correction

Apart from all the operations discussed above, Music sculptor also allows us to edit the pitch of a specific instrument - a feature that is very useful while different pieces of instrumental notes are cut and pasted - to evolve a new MIDI file.

When a Music score is conducted with several Instruments, the pitch or scale of each instrument is adjusted to be similar, to produce the synchronisation we enjoy with the blending of various kinds of music notes. Similarly, even with MIDI files, every instrument participating in a particular file is set to the same frequency, to produce a melodious synchronisation.

Now, when different Instrumental notes from different files are integrated with the editing options available, chances are that they may not maintain the same pitch, resulting in an asynchronous music.

It's exactly here, that the pitch correction feature comes to the rescue.

By selecting those notes which are asynchronous with the rest of the Music, and adjusting the pitch of the same (which can be incremented or decremented, depending upon one's requirement), the required synchronisation can be achieved with ease.

This brings us to the conclusion of our discussions on Music sculptor, MIDI as well as multimedia audio – as a whole.

## 10.8 Summary

MIDI files can be played back using windows media player and all other popular multimedia player software. Because of the complexities involved in MIDI recording, it pays to check out royalty free MIDI music libraries that are available on the Internet.

To build up a MIDI music studio, one needs to have a MIDI compatible synthesizer electronic keyboard, MIDI plug-in cables and MIDI sequencer software.

After the MIDI recording is over, the resulting MIDI data has to be edited to suit the specific requirements of the project. MIDI sequencer software is used to edit MIDI files.