



# Sound


# Multimedia System Sounds


- All computers are equipped with basic sounds such as beeps, dings and random startup sounds. However, in order to hear more sophisticated sounds (music and speeches), a user will need a sound card and either speakers or headphones.
- To see if a sound card installed on the machine, click Start! Settings! Control Panel and look for an icon called “Multimedia” or “Sounds”. Double-click on the icon to view your sound card information. To place a speaker icon in the bottom right hand tray, check the “Show Volume Control on Taskbar” box. To adjust the volume, double-click on the speaker icon and slide the Balance bar either up (louder) or down (quieter). Some laptops also have an external volume control knob.



# Digital Audio

- ▶ The sound recorded on an audio tape through a microphone or from other sources is in an analogue (continuous) form. The analogue format must be converted to a digital format for storage in a computer. This process is called digitizing. The method used for digitizing sound is called sampling.

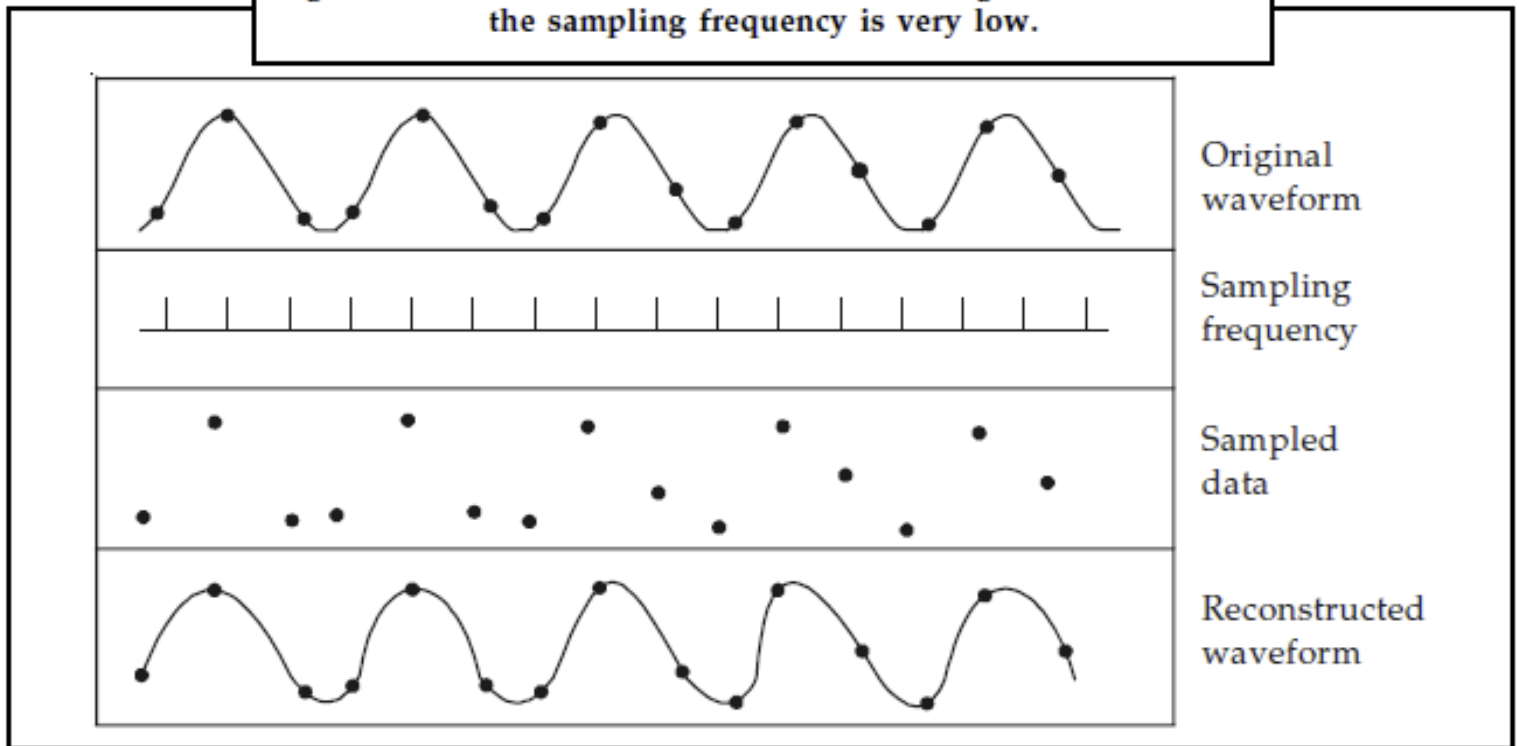
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- A dark grey arrow points to the right from the left edge of the slide. Several thin, light blue lines curve from the left side towards the text area.
- Digital audio represents a sound stored in thousands of numbers or samples. The quality of a digital recording depends upon how often the samples are taken. Digital data represents the loudness at discrete slices of time. It is not device dependent and should sound the same each time it is played. It is used for music CDs

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- ▶ The sampling rate determines the frequency at which samples will be drawn for the recording. The number of times the analogue sound is sampled during each period and transformed into digital information is called sampling rate. Sampling rates are calculated in Hertz (HZ or Kilo HZ). The most common sampling rates used in multimedia applications are 44.1 KHZ, 22.05 JHZ and 11.025 KHZ. Sampling at higher rates more accurately captures the high frequency content of the sound. Higher sampling rate means higher quality of sound. However, a higher sampling rate occupies greater storage capacity. Conversion from a higher sampling rate to a lower rate is possible.


# Sound Bit Depth

- ▶ Sampling rate and sound bit depth are the audio equivalent of resolution and colour depth of a graphic image. Bit depth depends on the amount of space in bytes used for storing a given piece of audio information. Higher the number of bytes higher is the quality of sound. Multimedia sound comes in 8-bit, 16-bit, 32-bit and 64-bit formats. An 8-bit has 28 or 256 possible values. A single bit rate and single sampling rate are recommended throughout the work. An audio file size can be calculated with the simple formula:
  - ▶ *File Size in Disk = (Length in seconds) × (sample rate) × (bit depth/8 bits per byte).*

Figure 3.1: It is difficult to reconstruct the original waveform if the sampling frequency is very low.



Source: Multimedia: Making It Work, By Tay Vaughan page 100

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- ▶ Bit Rate refers to the amount of data, specifically bits, transmitted or received per second. It is comparable to the sample rate but refers to the digital encoding of the sound. It refers specifically to how many digital 1s and 0s are used each second to represent the sound signal. This means the higher the bit rate, the higher the quality and size of your recording. For instance, an MP3 file might be described as having a bit rate of 320 kb/s or 320000 b/s. This indicates the amount of compressed data needed to store one second of music.

$$\text{Bit Rate} = (\text{Sample Rate}) \times (\text{Bit Depth}) \times (\text{Number of Channels})$$



# Mono or Stereo

- Mono sounds are flat and unrealistic compared to stereo sounds, which are much more dynamic and lifelike. However, stereo sound files require twice the storage capacity of mono sound files. Therefore, if storage and transfer are concerns, mono sound files may be the more appropriate choice.

Formula for determining the size of the digital audio is given below:

Monophonic = Sampling rate × duration of recording in seconds × (bit resolution/8) × 1

Stereo = Sampling rate × duration of recording in seconds × (bit resolution/8) × 2

# Analogue verses Digital

- There are two types of sound – analogue and digital. Analogue sound is a continuous stream of sound waves. To be understood by the computer, these sound waves must be converted to numbers. The process of converting analogue sounds into numbers is called digitizing or sound sampling. Analogue sounds that have been converted to numbers are digital sounds. When we are working with digital sound, we call it audio. Therefore, sound that has been converted from analogue to digital is often called digital audio sounds. Non-destructive sound processing methods maintain the original file. A copy of the original file can be manipulated by playing it louder or softer, combining it with other sounds on other tracks, or modifying it in other ways.
- Once a sound has been recorded, digitized, processed, and incorporated into a multimedia application, it is ready to be delivered. So that you can hear it through your speakers, the digital sound is sent through a digital-to-analogue converter (DAC).

# Delivery System

- ▶ The delivery system will vary as the intended audience of the multimedia application changes. When considering the delivery of sound, you should consider the number of different sounds that will be delivered, how they will be delivered, where they will be delivered, and to whom they will be delivered. In other words, the application may include one voice and some background music designed for one user in front of a desktop computer or it may be a presentation with various different types of music, narration, and special effects designed for a larger audience to be presented in an auditorium. Regardless, a high quality delivery system will help ensure that the sounds you've worked hard to create make the right impression.

# Types of Digital Audio File Formats

There are many different types of digital audio file formats that have resulted from working with different computer platforms and software. Some of the better known formats include:

# WAV

- ▶ WAV is the Waveform format. It is the most commonly used and supported format on the Windows platform. Developed by Microsoft, the Wave format is a subset of RIFE RIFF is capable of sampling rates of 8 and 16 bits. With Wave, there are several different encoding methods to choose from including Wave or PCM format. Therefore, when developing sound for the Internet, it is important to make sure you use the encoding method that the player you're recommending

# AU

- ▶ AU is the Sun Audio format. It was developed by Sun Microsystems to be used on UNIX, NeXT and Sun Sparc workstations. It is a 16-bit compressed audio format that is fairly prevalent on the Web. This is probably because it plays on the widest number of platforms.

# RA

- RA is Progressive Networks RealAudio format. It is very popular for streaming audio on the Internet because it offers good compression up to a factor of 18. Streaming technology enables a sound file to begin playing before the entire file has been downloaded.

# AIFF

- AIFF or AFF is Apple's Audio Interchange File Format. This is the Macintosh waveform format. It is also supported on IBM compatibles and Silicon Graphics machines. The AIFF format supports a large number of sampling rates up to 32 bits.



# MPEG

- ▶ MPEG and MPEG2 are the Motion Picture Experts Group formats. They are a compressed audio and video format. Some Web sites use these formats for their audio because their compression capabilities offer up to a factor of at least 14:1. These formats will probably become quite widespread as the price of hardware based MPEG decoders continues to go down and as software decoders and faster processors become more mainstream. In addition, MPEG is a standard format.

# MIDI

- MIDI (MID, MDI, MFF) is an internationally accepted file format used to store Musical Instrument Digital Interface (MIDI) data. It is a format used to represent electronic music produced by a MIDI device (such as a synthesizer or electronic keyboard). This format provides instructions on how to replay music, but it does not actually record the waveform. For this reason, MIDI files are small and efficient, which is why they are often used on the Web.