Exhibit message

Digital music is often laid down in ‘tracks’, with each instrument or vocal piece being recorded on one ‘track’ as numbers.

Digital music (found in CDs and .mp3 or .wav files) is a stream of numbers that is converted into an analogue sound that we can hear.

Quick Fact

While Thomas Edison is credited with creating the first device for recording and playing back sounds in 1877, the recording he made that year was lost.

A regularly-broadcast phonograph of Edison reciting *Mary Had a Little Lamb* was recorded much later (about 1920).

Graphic panel text

Thomas Edison is believed to have invented the phonograph to record and play back sound in 1877.

As Edison spoke, a needle vibrated and scratched patterns onto tin foil.

This was an **analogue** recording (similar to LP vinyl records and cassette tapes).

Today, a **digital** recording converts sound waves into a stream of numbers.

To play back the music, the stream of numbers is converted back to an analogue sound wave by an electronic converter.
Digital recording is used on compact discs (CDs), mp3 files and the .wav files you can hear in this exhibit.

Want to know more about digital music?

Thomas Edison's 1877 phonograph had two needles: one for recording and one for playback.

When Edison spoke into the mouthpiece, the recording needle (attached to a diaphragm) vibrated and scratched marks into tinfoil wrapped around a cylinder.

The playback needle was attached to another diaphragm, so when the needle moved over the tinfoil markings, the diaphragm would vibrate and generate sound to match the recorded scratchings.

Edison thought that his phonograph could be used for:

- dictation and letter writing
- audio books for blind people
- recording the voices of family members for genealogy
- music boxes and toys and
- recording telephone conversations.

After Edison’s foil-coated cylinder, new technologies were developed to record sound.

These included bakelite, then vinyl LP (long play) records and cassette tapes.

During the 1950s, magnetic tape was used to make high-quality recordings and to edit or manipulate the recordings into new sounds.

Computer music uses electronic technology to generate and manipulate sound. The first time a computer was used to synthesise sounds and music took place in 1957 at the Bell Telephone Laboratories in New Jersey, USA.

From the mid-1960s, electronic oscillators, filters and amplifiers were used to synthesise and process recordings instead of manually cutting and pasting strips of magnetic tape together.

MIDI systems introduced from 1983 meant that music was being stored on computer hard drives or disks rather than magnetic tapes.

MIDI is an abbreviation of **Musical Instrument Digital Interface**. It's a form of communication between electronic musical instruments (usually electronic piano keyboards) and computers.

Digital music recordings (used from the 1990s onwards) need to convert acoustic sound waves into electric voltages (via microphones), then they convert these voltages into a series of numbers to be stored as a digital mp3 file or on CD.

When playing the digital files, the process is reversed. So numbers representing the sound waves are converted into voltages through a digital-to-analogue converter and these voltage steps are amplified and transmitted through stereo speakers so you can hear the music.

This conversion happens every time you listen to an mp3 file or compact disc (CD).

Extra for experts

Technically, digital music files have lower **fidelity** or sound quality than other recorded files or live music, although many people say they find it difficult to pick the difference between digital recordings and other methods of broadcasting.

Sound quality (fidelity) and the frequency (pitch) range of a digital recording depends on how many number samples are recorded (the sampling rate in samples per second).

Humans can distinguish sounds that vibrate at frequencies between 20 to 20 000 hertz (Hz), but human ears are primed to best detect the frequency range of human speech (1000–4000 Hz).

In digital recordings, the upper frequency limit, known as the Nyquist frequency, is equivalent to half the sampling rate.

So older computers that recorded or reproduced 20 000 number samples per second reproduced a frequency range of 10 000 hertz (10kHz).

This sampling rate is adequate for spoken interviews, but not fast enough for musical instruments which need a broader range of pitch frequencies than the human voice.

Modern computers can usually take 48 000 samples per second, allowing the entire frequency range of the human ear (up to 20 000 hertz) to be reproduced and catering to recorded music ranges.

Further information

- *How Stuff Works—analogue and digital recording*
  http://electronics.howstuffworks.com/analog-digital1.htm

- *Grove Music Online*
  http://www.grovemusic.com
  Oxford University Press