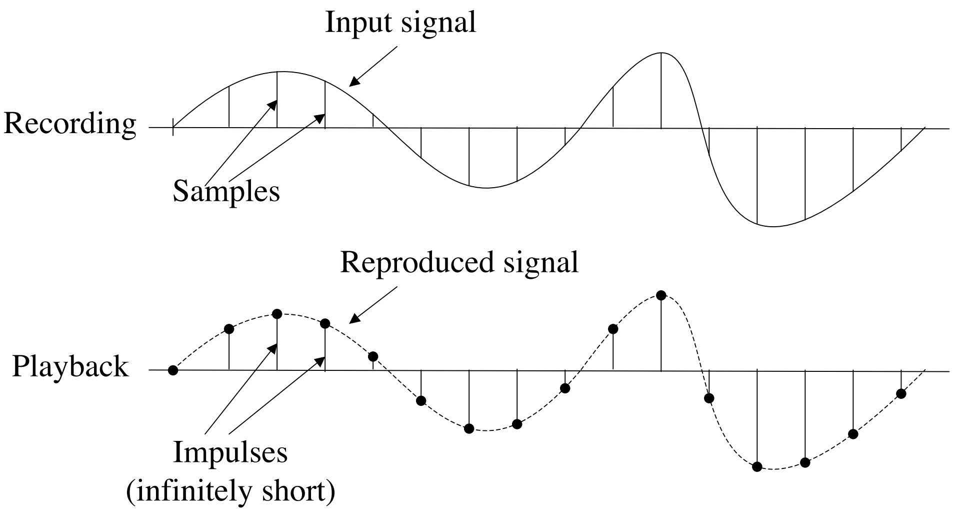
Introduction to Digital Audio

How computers store and process sound

Sampling Theorem

- Based on work by Shannon and Nyquist.
- Provided certain limitations are observed, any arbitrary wave form can be recorded by taking samples at fixed time intervals.
- The wave form is played back by outputting the same samples at the same fixed intervals.
- The output samples must be infinitely short impulses.
- Sampling Frequency (f_s) is the number of samples taken in one second.

Recording and Playback



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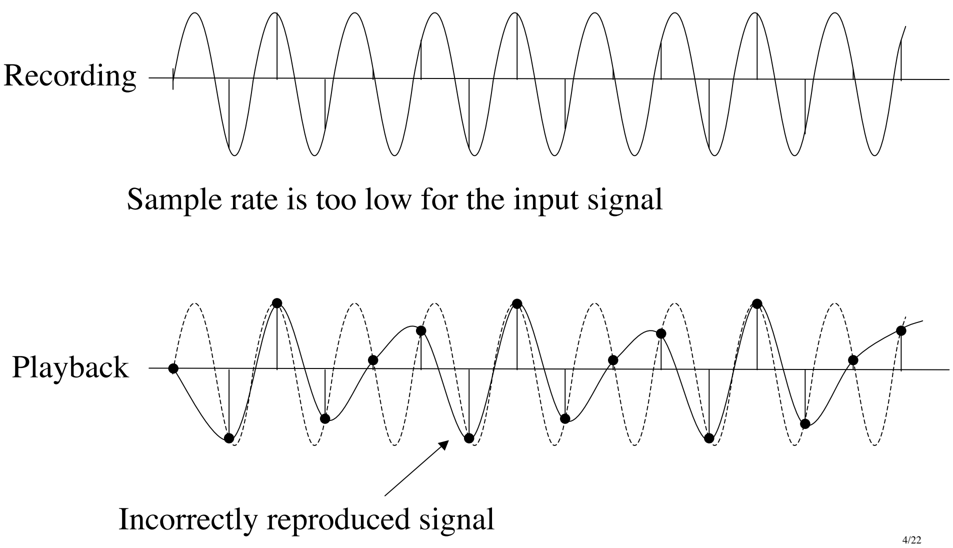
Aliasing



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What is Aliasing?

- Aliasing occurs when the input frequency is greater than half the sampling frequency.
- This point is called the **Nyquist** frequency: $f_N = f_s / 2$
- Audio equivalent of Wagon Wheel Effect seen on movies.
- Any frequency component above f_N is mapped to a frequency below f_N
- Example: A signal 2 kHz above f_N is mapped to a frequency 2 kHz below f_N

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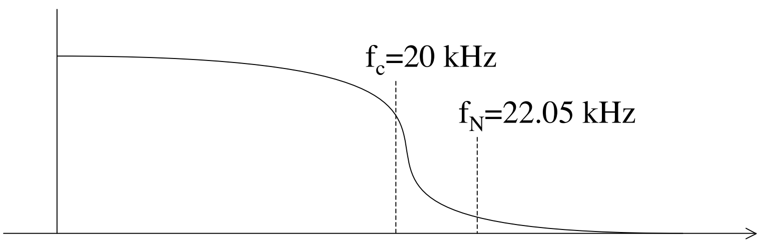
Anti-Aliasing Filter

- Aliasing is prevented by putting the signal through an Anti-Aliasing (Low Pass) filter **before** it is sampled, removing components above the Nyquist frequency.
- For CD recordings:
  - Sampling frequency f_s = 44.1 kHz
  - Nyquist frequency f_N = 22.05 kHz (half sampling frequency)
  - Cut off frequency f_c = 20 kHz (limit of human hearing)



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Anti-Imaging Filter

- The output signal is made up of impulses.
- An impulse contains all frequencies up to infinity.
- Therefore, the output signal has unwanted spectral components above the Nyquist frequency.
- The unwanted components must be removed using an Anti-Imaging (Low Pass) filter.

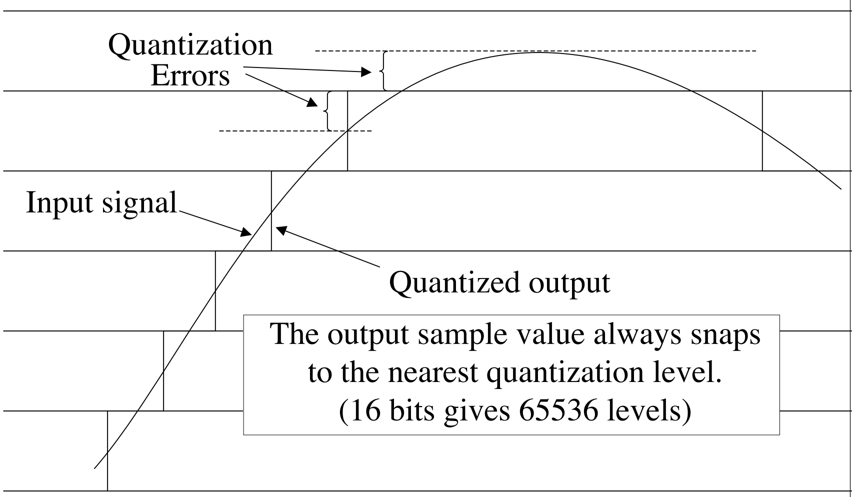
Sampled sound reproduction requires **two** filters:

- Anti-Aliasing filter applied before the input is fed to the sampler.
- Anti-Imaging filter applied to the output.
- Both filters have similar (or identical) characteristics

CD Sample Rate

- The CD sample rate (44.1kHz) poses severe problems for the anti-aliasing and anti-imaging filters.
- The sharp cut off between 20kHz to 22kHz causes severe phase shifts and an audible *metallic* effect.
- Higher sampling rates reduce this problem, hence:
  - Sound is recorded in the studio at a higher sampling rate and then processed down to the lower CD rate.
  - CD players reverse the process and generate intermediate samples which are output at a faster rate
  - DVDs record sound data at 192kHz)

Quantization



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Quantization Levels

- After sampling, the signal is stored as a 16 bit number.
- This only allows a maximum of 65536 levels, hence there will be a difference (error) between the actual signal value and the value being stored.
- On playback, this error is audible as extra random noise which has been added by the system.

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Quantization Noise

- With 16 bit sampling, the quantization noise gives a signal to noise ratio of approx **96 dB**.
- Each sample bit gives approx **6 dB** SNR.
- The best SNR can only be achieved when the signal spans the full range of levels. Weaker signals use fewer levels (bits) and so have a worse SNR.
- Very weak signals have an intrusive noise characteristic (Grainy or Bird Tweets). This can be reduced by adding a small amount of random noise (**Dithering**)

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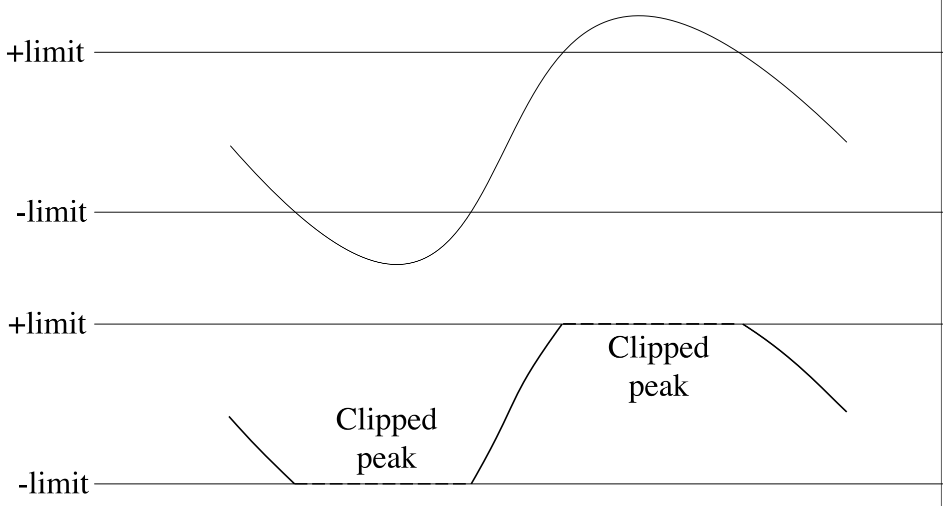
Clipping



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Digital Filters

- Digital filters process a digital audio stream to selectively adjust the amplitude and/or phase of its frequency components.
- It is possible to design High Pass, Low Pass, Band Pass and Band Cut filters digitally.
- It is possible to design digital filters that are very difficult to implement in hardware, such as brick wall low pass filters.
- Digital filter mathematical theory is highly advanced and very complex!

The FIR Filter

- A common digital filter algorithm is the Finite Impulse Response (FIR) filter.
- It uses a technique called **Convolution** or Multiply-and-Add:
    - Multiply a block of input samples by values in a table
    - Add the products to produce one output sample
    - Slide the table by one input sample and repeat.
- The table of multipliers is typically very large (more than 200 values).

See separate animation

The IIR Filter

- An extension of the FIR filter is the Infinite Impulse Response (IIR) filter.
- The first stage of an IIR filter is identical to the FIR filter.
- The outputs are reprocessed by a second table of multipliers.
- Can do more extensive filtering than the FIR filter, with fewer calculations.

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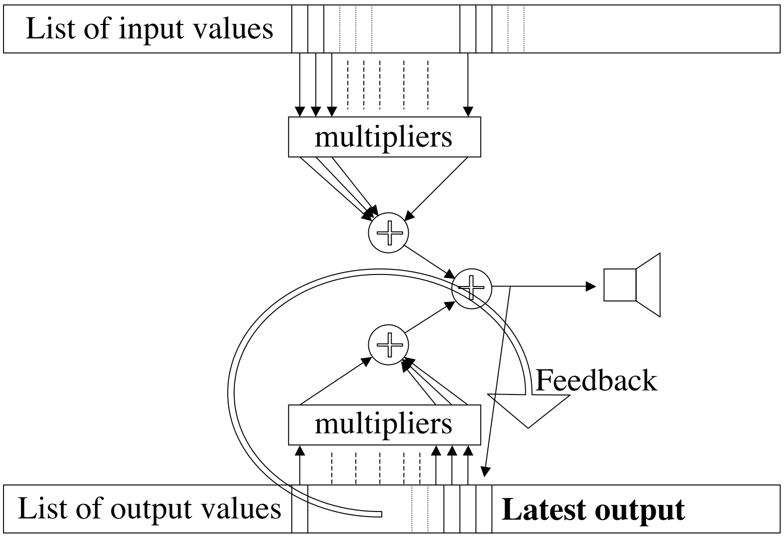
IIR Filter



Sound Card Capabilities

- A modern high-end sound card (Audigy 2 ZS) for PC offers the following features:
  - Sound Input and Output:
    24 bit input/output in multiple channels
    Multiple formats (including MP3)
    Multiple sampling rates up to 192 KHz
  - Real-Time Audio Processing, including 3D effects
  - Wave Table Synthesis:
    High quality sampled sound set
  - MIDI Input/Output ports.
  - Fire Wire Connectivity.

Direct X

- Direct X is Microsoft's technology to allow fast audio and video processing for Windows.
- Originally devised in response to the demands of game programmers.
- Now includes features such as:
  - Direct Sound.
  - Direct Sound 3D
  - Direct Music

Downloading Music

- A typical pop song plays for about 4 minutes and requires:
  - 2 x 2 x 44100 x 60 x 4 bytes = 42336000 bytes
  - Approximately 10 Mbytes per minute
- Downloading over the Internet using a 56kbs modem would take:
  - 42336000 x 8 /56000 = 6048 sec ≈ 100 min
- Such timings would make the Internet an impractical music distribution medium.

MP 3

- MP3 is a technique which reduces the amount of data in an audio file (**compression**).
- It uses a psycho-acoustic model of human hearing to remove inaudible frequency components in a signal.
- This makes MP3 a lossy compression system.
- In tests, compressions of 14:1 cannot be distinguished from normal CD quality audio.
- MP3 has made it practical to distribute music files over the internet.

Streaming

- A compressed song only takes a few minutes to download. This means it is possible to start playing *before* the download has completed. This is called **Streaming**
- Streaming requires a special player program which can access the net and play the file:
    - Windows Media Player
    - Real Player
    - Quick Time
- Many broadcasters provide streamed versions of radio and TV programs on their web sites.