

ECS614U/ECS749P: Sound Recording and Production

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Audio Signal Chain

Signal Chains

- The signal chain describes all links in the chain from the sound source to the receiver.
- Each component will have it's own **Frequency Response**.
- It is very hard to recreate a sound source accurately.

Sound Reinforcement Signal Chain

- What is the signal chain if we want to reinforce a sound source, e.g. public address system, live performance?

Sound Reinforcement Signal Chain

SOURCE

RECEIVER

Sound Reinforcement Signal Chain



Sound Reinforcement Signal Chain



Sound Reinforcement Signal Chain

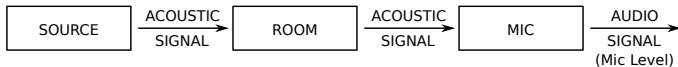


RECEIVER

Sound Reinforcement Signal Chain

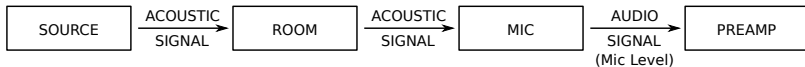


Sound Reinforcement Signal Chain



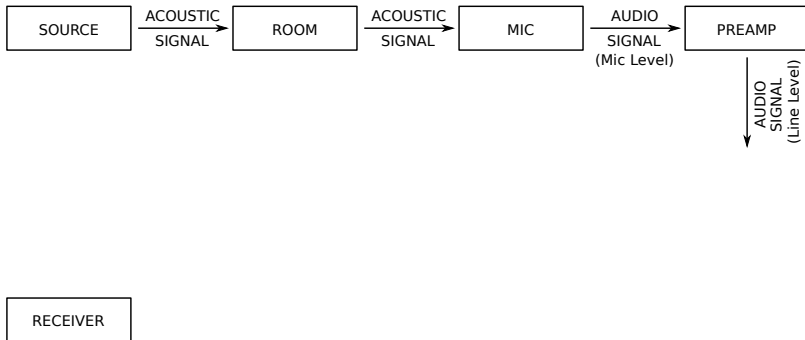
RECEIVER

Sound Reinforcement Signal Chain

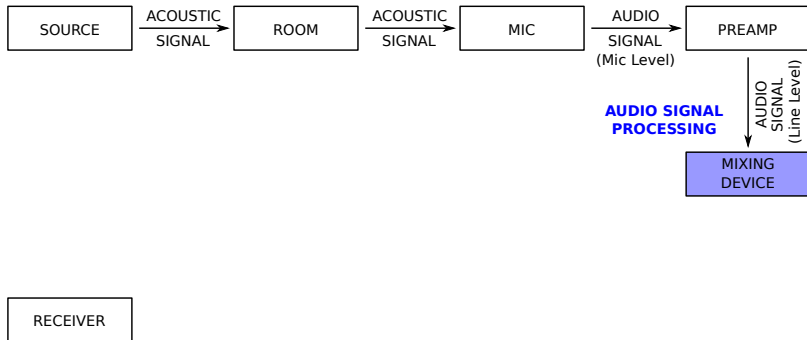


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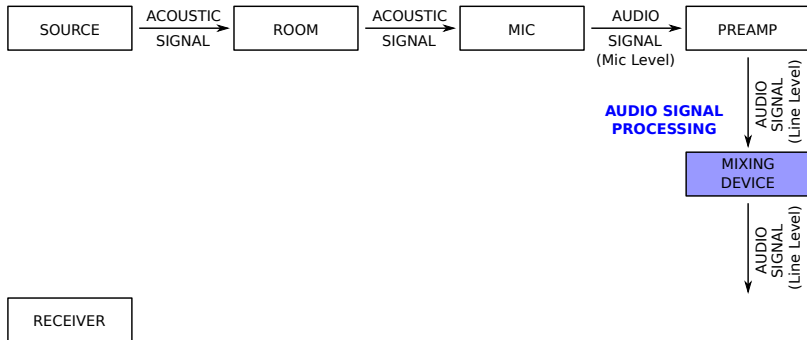
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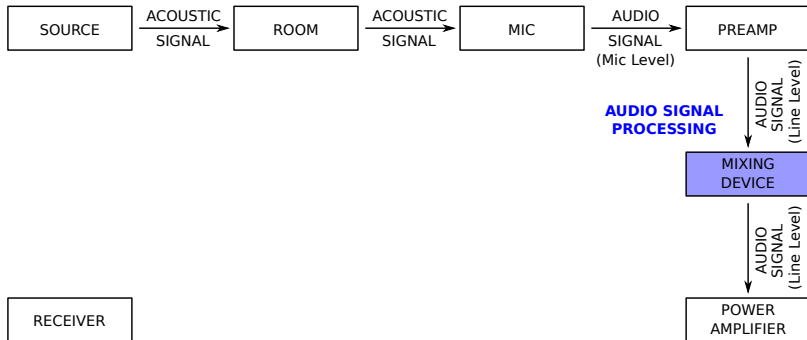
Sound Reinforcement Signal Chain



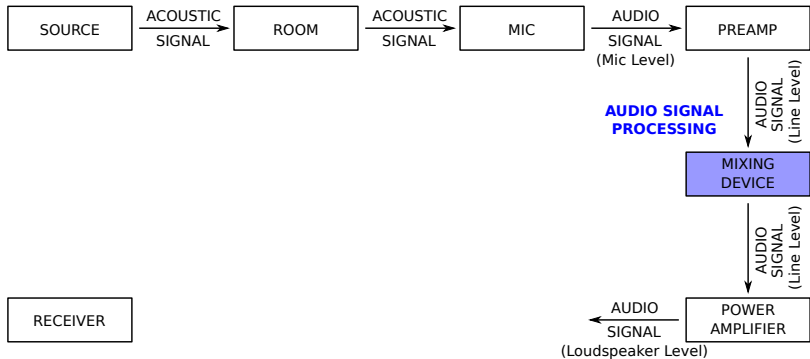
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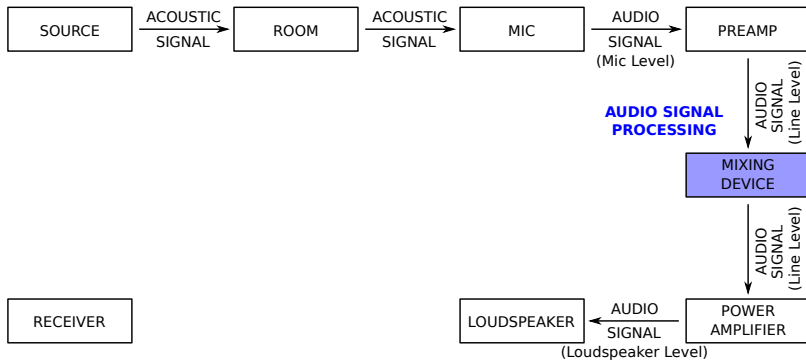
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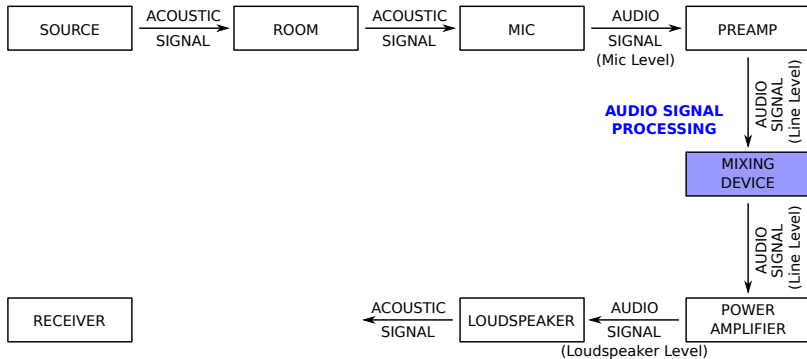
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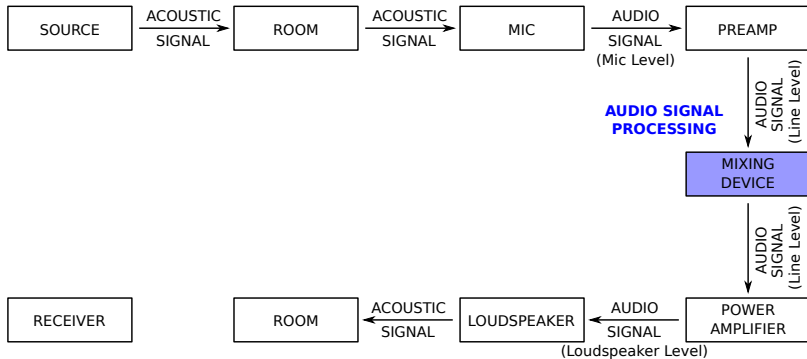
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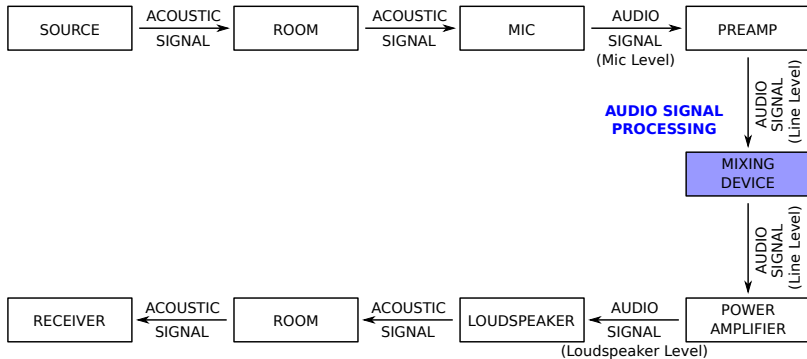
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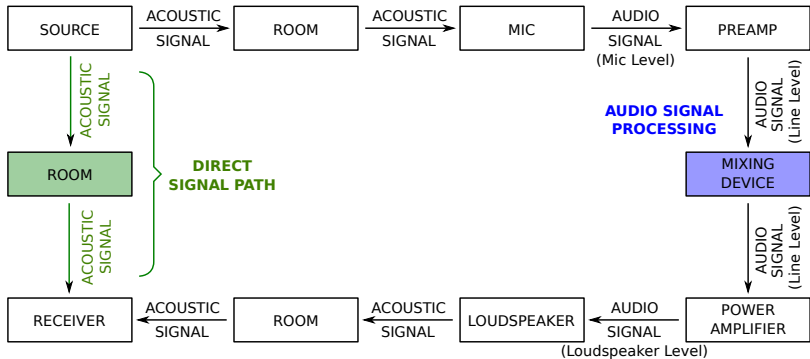
Sound Reinforcement Signal Chain



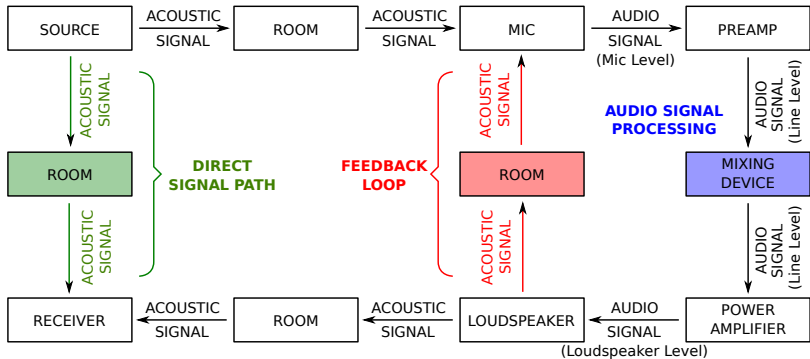
Sound Reinforcement Signal Chain



Sound Reinforcement Signal Chain



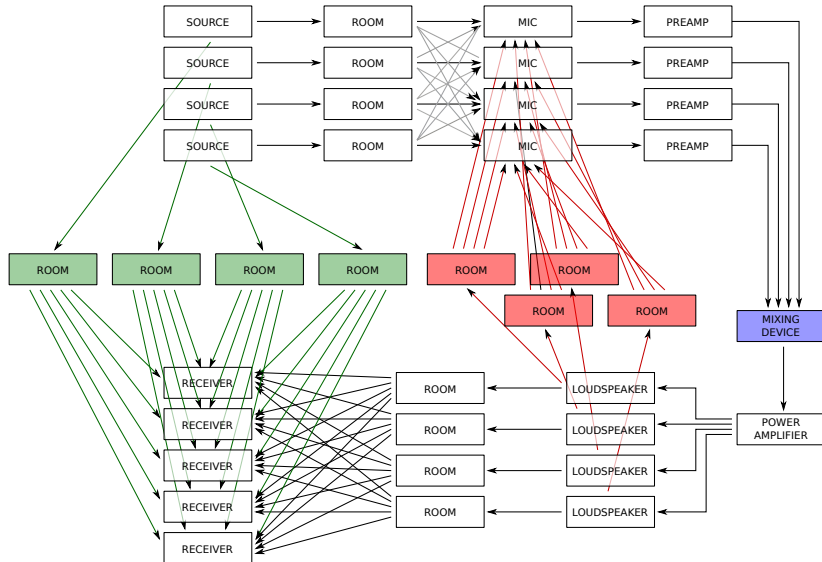
Sound Reinforcement Signal Chain



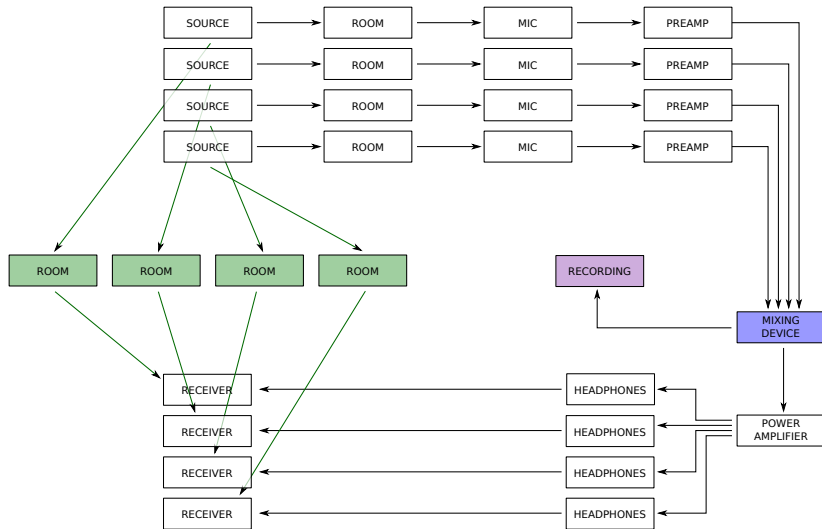
Sound Reinforcement Signal Chain

- In live music we have many sources and many receivers (listeners).
- Things can get very complicated very quickly...!

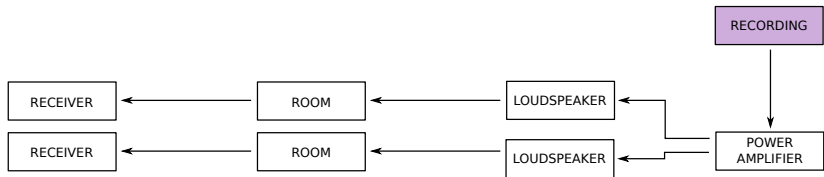
Live Music Signal Chain



Studio Recording Signal Chain



Reproduction Signal Chain

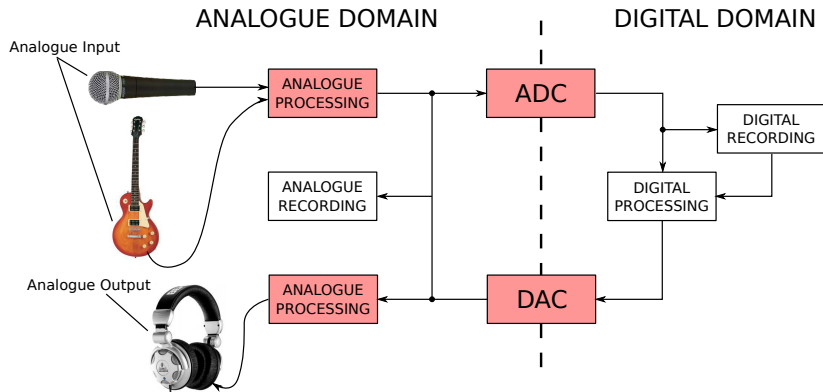


Digital Recording Chain

Analogue vs Digital

- Analogue audio signals are **continuous** in both time and amplitude.
- Digital audio signals contain samples of an analogue signal at **discrete** points in time, and with a **finite** number of possible amplitude values.

The Recording Chain

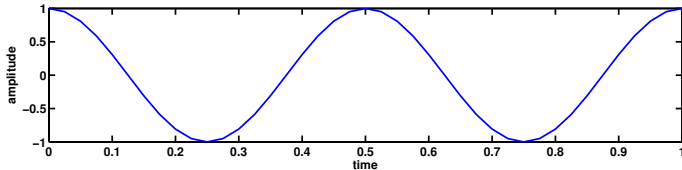


The Recording Chain

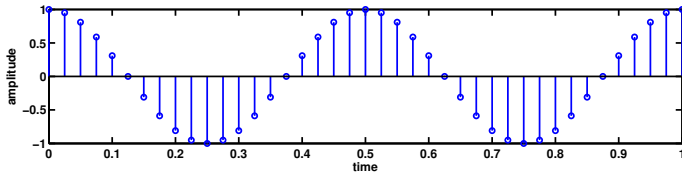
- When we produce sound, e.g. using an instrument, or when we reproduce sound, e.g. using a loudspeaker, the pressure wave is always an analogue signal.
- When we process audio signals within a computer they are digital signals.
- What kind of signals are processed by our auditory system?

A-D conversion: Sampling

Analogue signal:



Sampled signal:

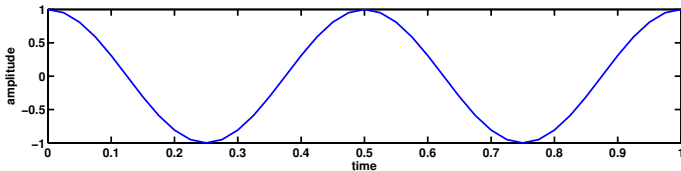


A-D conversion: Sampling Rate

- Data samples of the analogue signal are taken at a fixed rate known as the sample rate. It defines how many data samples are taken per second.
- CD quality audio has a sample rate of 44,100 samples per second (Hz).
- A low sample rate affects the quality of the audio and restricts the usable portion of the frequency spectrum.
- The “correct” sample rate for audio is still a matter for debate!!!

A-D conversion: Insufficient sampling

Analogue signal:



Poorly sampled signal:

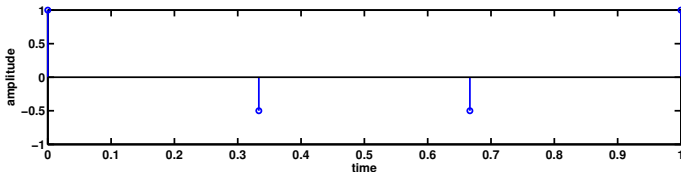
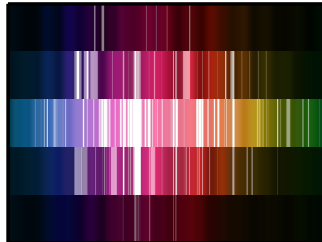
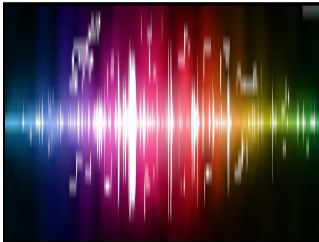
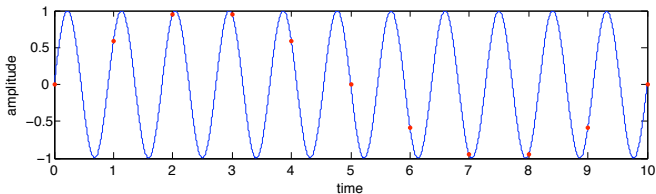


Image resolution

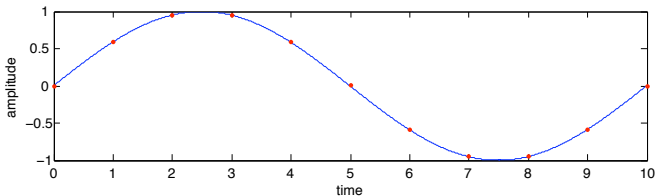


A-D conversion: Aliasing

Original sine wave:

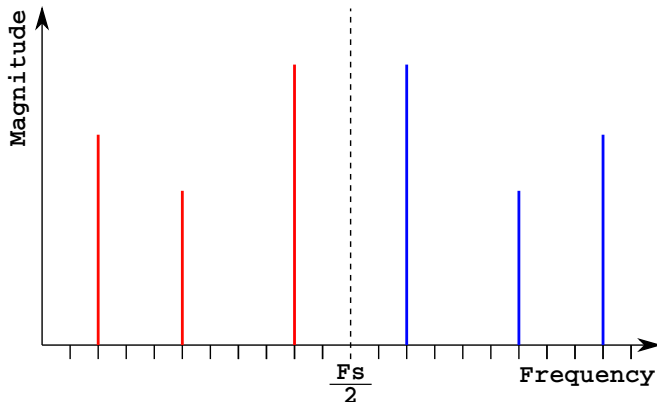


Aliased sine wave:



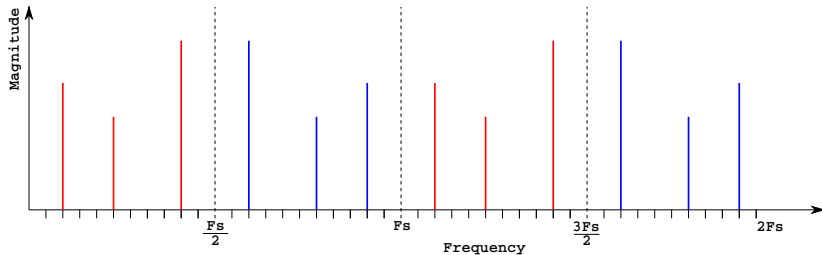
A-D conversion: Aliasing

- The frequency response of any sampled signal will be a mirror image about the middle frequency.



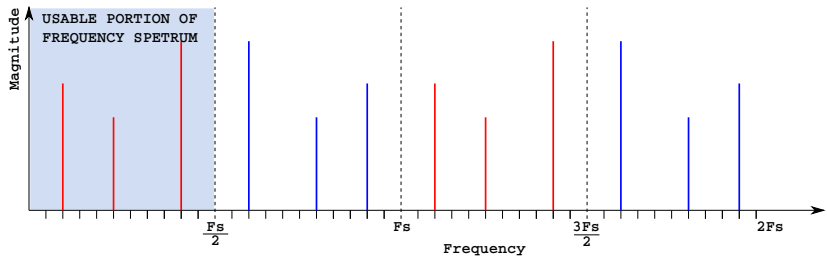
A-D conversion: Aliasing

- The mirror images continue as the frequency increases.



A-D conversion: Aliasing

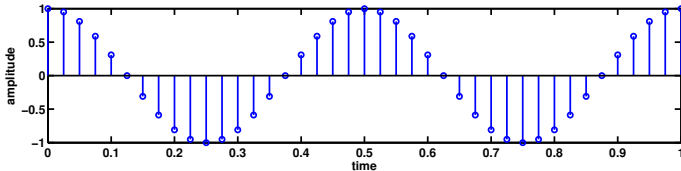
- To avoid aliasing effects, only the first portion of the frequency spectrum can be used.



- The Nyquist frequency is equal to half the sample rate, for CD audio this is 22.05kHz.
- The frequency response above the Nyquist frequency is a mirror image of the response below.
- The signal is filtered using an analogue low pass filter before ADC, and digital filter after, to remove all components of the signal which are above the Nyquist frequency.

A-D conversion: Quantisation

Sampled signal:



Quantised signal:

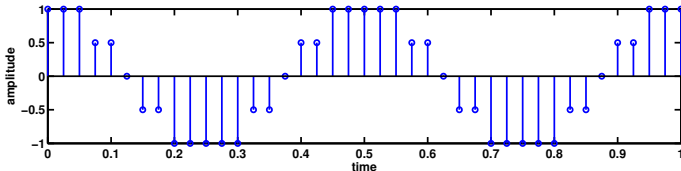
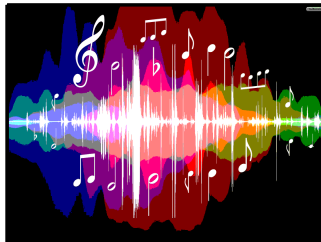


Image resolution

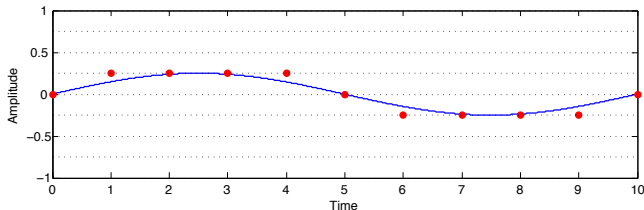
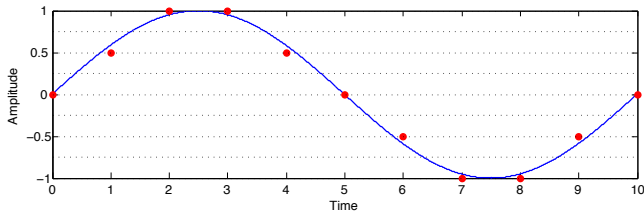


A-D conversion: Bit Depth

- The amplitude range is divided into a number of discrete amplitude values and is defined by the bit depth.
- Each sample is quantised to the nearest amplitude value. The difference between the actual and quantised signal is the quantisation error.
- CD quality has a bit depth of 16 bit -> 32768 amplitude points.
- If the full dynamic range is not used the effective bit depth is reduced and the errors introduced by quantisation become more significant.

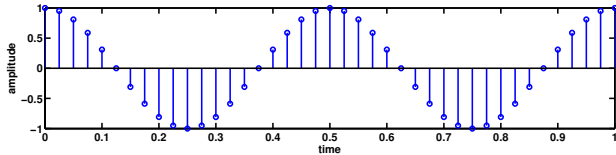
A-D conversion: Effect of dynamic range

- The full dynamic range should be used to minimise quantisation errors.

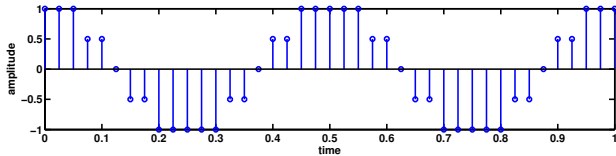


A-D conversion: Quantisation as signal plus noise

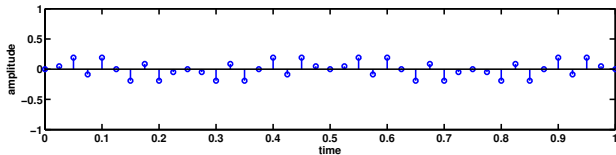
Sampled signal:



Quantised signal:

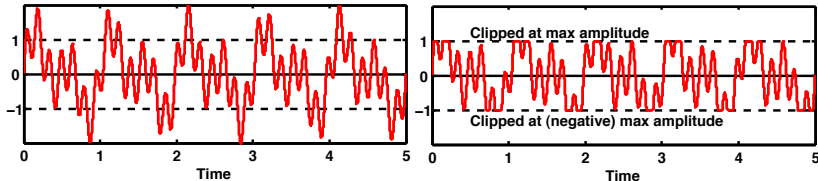


Error signal:



Clipping and Distortion

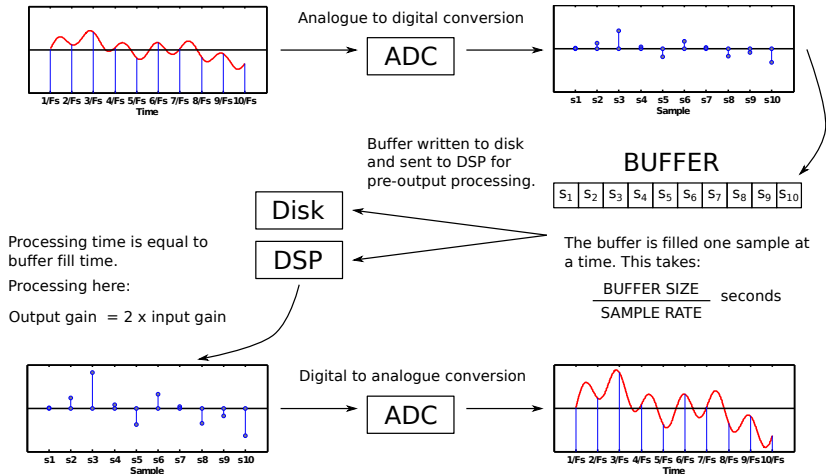
- Whenever we are dealing with an electrical signal, whether analogue or digital, there is a limit on the available amplitude.
- If the amplitude limit is exceeded the signal will be clipped. This leads to a distorted sound.



Clipping and Distortion

- Analogue clipping is more gentle (particularly if tube/valve amp) and is simulated to make a distorted sound.
- Digital clipping is destructive and gives a very harsh distorted sound. It sounds horrible and is never desirable.
- It is essential that the (analogue) input gain on the audio interface is set so that the input signal lies in the available amplitude range. If the sampled signal is clipped this cannot be corrected in the digital domain.

The digital part



Latency

- Latency is an unavoidable disadvantage of digital recording.
- Reducing buffer size will reduce the latency but will increase the processing load on the computer.
- Direct monitoring of the analogue input is the only way to completely remove latency when recording.