# Audio DSP

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## Sound

- Sound: vibration transmitted through a medium (gas, liquid, solid and plasma) composed of frequencies capable of being detected by ears.
  - Note: sound cannot travel through a vacuum.
- Human detectable sound is often characterized by air pressure variations detected by the human ear.
- The amplitude, frequency and relative phase of the air pressure signal components determine (in part) the way the sound is perceived.





## Sinusoids and Sound: Amplitude

- A fundamental unit of sound is the sinusoidal signal.

$$x_a(t) = A\cos(2\pi F_0 t + \theta), \qquad t \in \mathbb{R}$$

  - $A \equiv$ volume
  - $F_0 \equiv$ pitch (more on this ...)
  - $\theta \equiv$ phase (more on this ...)





## Sound Volume

- Volume = Amplitude of sound waves/audio signals
- quoted in dB, which is a logarithmic measure; $10\log(A^2)$
  - no sound/null is $-\infty$ dB
- Loudness is a subjective measure of sound psychologically correlating to the strength of the sound signal.
  - the volume is an objective measure and does not have a one-to-one correspondence with loudness
  - perceived loudness varies from person-to-person and depends on frequency and duration of the sound

## Music Volume Dynamic Range

Tests conducted for the musical note: C6 ($F_0 = 1046.502$ Hz).

| Dynamic Level | Decibels |
|---|---|
| Threshold of hearing | 0 |
| ppp (pianissimo) | 40 |
| p (piano) | 60 |
| f (forte) | 80 |
| fff (fortississimo) | 100 |
| Threshold of pain | 120 |





## Sinusoids and Sound: Frequency

- A fundamental unit of sound is the sinusoidal signal.

$$x_a(t) = A\cos(2\pi F_0 t + \theta), \qquad t \in \mathbb{R}$$

  - $A \equiv$ volume
  - $F_0 \equiv$ pitch
  - $\theta \equiv$ phase (more on this ...)





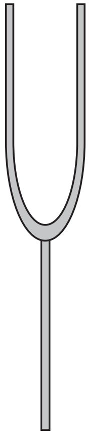
## Pure Frequency

- **Q:** What type of sound does a pure frequency produce?
  - **A:** A pure tone with a single pitch.
- **Q:** Can any instrument produce a pure tone by playing a single note?
  - **A:** No.





## Tuning Forks



- A tuning fork is a two-pronged instrument that is an acoustic resonator. It is usually made out of steel and resonates at a specific constant pitch which is a function of the length of the prongs.
  - Striking the tuning fork will produce the required sounds although initially there may be overtones that die out quickly.
  - A very common tuning fork used by musicians produces the A note ($F_0 = 440$ Hz), which is international concert pitch used to tune orchestras.

# Frequency and Pitch

- Sinusoids can be represented either as:

$$x_a(t) = A\cos(2\pi F_0 t + \theta), \qquad t \in \mathbb{R}$$

or for mathematical convenience when interpreting as Fourier signal components as:

$$x_a(t) = Ae^{j(2\pi F_0 t + \theta)}, \qquad t \in \mathbb{R}$$

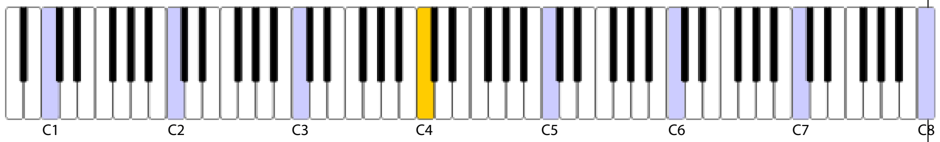
- Pitch is directly related to the frequency $F_0$.
- To be able to hear a frequency $F_0$, it has to be in the human audible range (20 Hz to 20,000 Hz).





# Harmonically Related Frequencies and Pitch

| Scientific Designation | Frequency (Hz) | $k$ for $F_0 = 8.176$ |
|---|---|---|
| C1 | 32.703 | 4 |
| C2 | 65.406 | 8 |
| C3 | 130.813 | 16 |
| C4 (middle C) | 261.626 | 32 |
| C5 | 523.251 | 64 |
| C6 | 1046.502 | 128 |
| C7 | 2093.005 | 256 |
| C8 | 4186.009 | 512 |








# Harmonically Related Frequencies

- Recall harmonically related sinusoids have the following analytic form for $k \in \mathbb{Z}$:

$$x_{a,k}(t) = A\cos(2\pi k F_0 t + \theta)$$

or

$$x_{a,k}(t) = Ae^{j(2\pi k F_0 t + \theta)}$$

- They are used in the context of the Fourier Series to build periodic signals:

$$x(t) = \sum_{k=-\infty}^{\infty} X(k) e^{j(2\pi k F_0 t)}$$

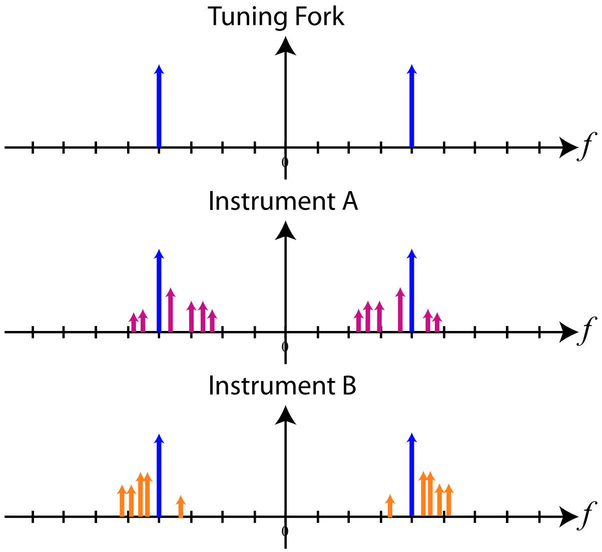




# Signature Sounds

- **Q:** If two different people sing the same note or two different instruments play the same note, why do they sound different?
  - The notes are not pure tones. There are natural overtones and undertones that provide distinguishing signatures that can be viewed in the associated spectra.

Fourier Transforms of the Same Note








## Human Audible Range

- Hearing is usually limited to frequencies between 20 Hz and 20 kHz.
- The upper limit decreases with age.
  - The audible frequency range is different for animals





## Animal Audible Range

| Species | Approx Range (Hz) |
|---|---|
| human | 20 - 20,000 |
| dog | 67 - 45,000 |
| rabbit | 360 - 42,000 |
| bat | 2,000 - 110,000 |
| goldfish | 20 - 3,000 |

Reference: R.R. Fay (1988), Hearing in Vertebrates: A Psychophysics Databook.





## Sinusoids and Sound: Phase

- A fundamental unit of sound is the sinusoidal signal.

$$x_a(t) = A\cos(2\pi F_0 t + \theta), \qquad t \in \mathbb{R}$$

  - $A \equiv$ volume
  - $F_0 \equiv$ pitch
  - $\theta \equiv$ phase

# Phase and Sound

- An audio signal is represented by a real function $x(t)$.
- The function $x(-t)$ represents playing the audio signal backwards.
- Since $x(t)$ is real:

$$\begin{aligned} X(F) &= X^*(-F) \\ |X(F)| &= |X^*(-F)| = |X(-F)| \quad \text{since } |c| = |c^*| \text{ for } c \in \mathbb{C} \end{aligned}$$

- Therefore,

$$|X(F)| = |X(-F)|$$

  That is, the CTFT magnitude is even for $x(t)$ real.





# Phase and Sound

- Recall, $x(t) \overset{\mathcal{F}}{\longleftrightarrow} X(F)$ $\quad x(-t) \overset{\mathcal{F}}{\longleftrightarrow} X(-F)$
- Therefore,

$$\underbrace{|X(F)|}_{\text{spectrum magnitude of } x(t)} = \overbrace{|X(-F)|}^{\text{spectrum magnitude of } x(-t)}$$

Therefore, the magnitude of the FT of an audio signal played forward and backward is the same!





# Phase and Sound

- Therefore, for

$$\begin{aligned} x(t) &\overset{\mathcal{F}}{\longleftrightarrow} X(F) \\ x(-t) &\overset{\mathcal{F}}{\longleftrightarrow} X(-F) \end{aligned}$$

- $|X(F)| = |X(-F)| \Rightarrow$ the CTFT magnitudes for forward and reverse sound signals are exactly the same.
- $\angle X(f) \neq \angle X(-f) \Rightarrow$ the CTFT phases for forward and reverse sound signals are different.
- Therefore, the relative phase of the sinusoidal components of sound contains very salient perceptual information much like for images.





# Auditory Masking

- occurs when the perceived quality of one (primary) sound is affected by the presence of another (secondary) sound
  - Simultaneous masking: the secondary sound is heard at the same time as the primary sound
- Can be exploited to mask non-ideal signal processing.

# Why Digitize Audio?

- Fidelity of digital audio is much higher than analog audio.
- Manipulation tools for digital audio are much more sophisticated than those available for analog audio.
- Compression of digital audio provides significantly reduced storage requirements.
- Storage of digital audio (e.g., CDs) are much more convenient and compact.
- Duplication of digital audio is exact in contrast to analog audio.





# Benefits of Digital Audio

- Convenient recording, enhancement, mass-production and distribution.
  - CDs, online stores such as iTunes, etc.
  - data files are distributed instead of physical media storing the information such as records and tapes.





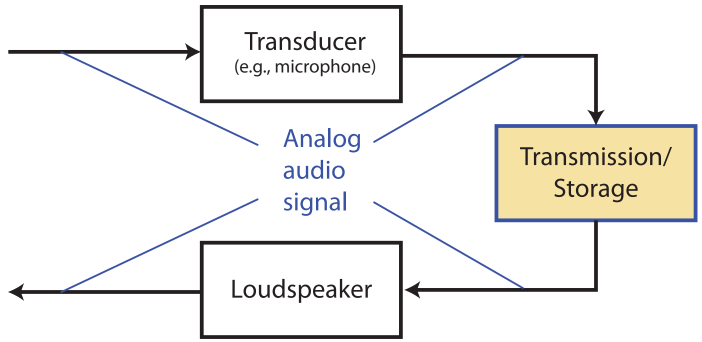
# Concerns about Digital Audio

- Convenient recording, enhancement, mass-production and distribution.
  - unlawful manipulation of recorded audio is difficult to detect
  - piracy: unlawful copying and redistribution of copyrighted content



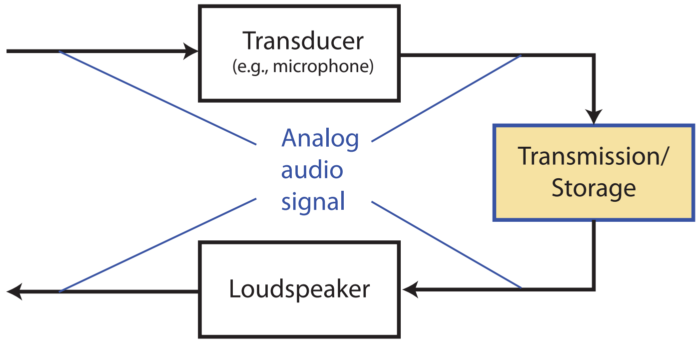


# Analog vs. Digital Audio: Analog Audio System




- microphone: converts sound into an electrical signal;
  air pressure → motion of conductor/coil → magnetic field → electrical signal
- loudspeaker: converts electrical signal into acoustic waves;
  electrical signal → magnetic field → motion → air pressure

# Analog vs. Digital Audio: Analog Audio System

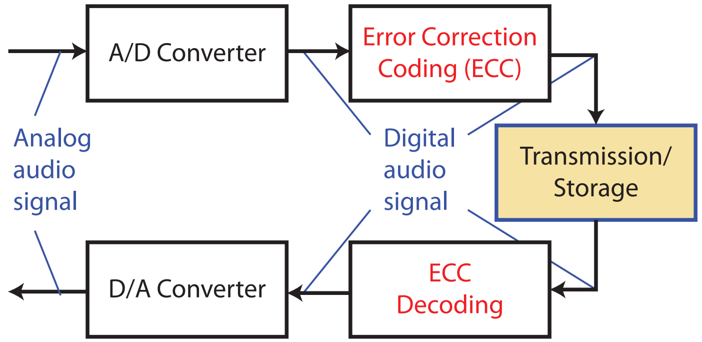



- associated circuits suffer from inherent noise (noise floor)
- capacitance and inductance of the circuits limit bandwidth, and resistance limits amplitude





# Analog vs. Digital Audio: Digital Audio Chain

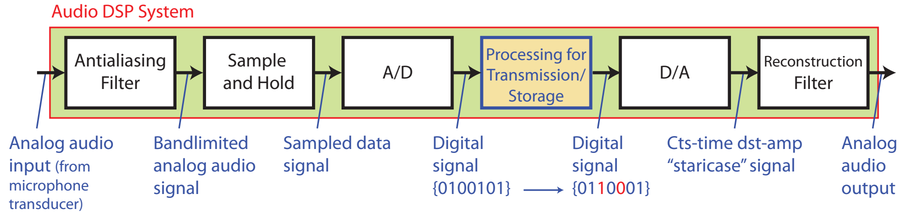



- fidelity limited by quantization noise
- bandwidth limited by sampling rate
- dynamic range limited by bit resolution





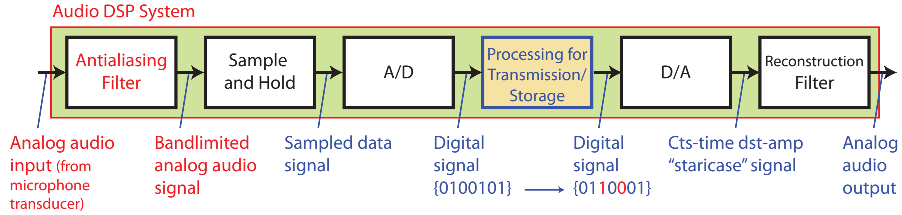
# Digitizing Audio







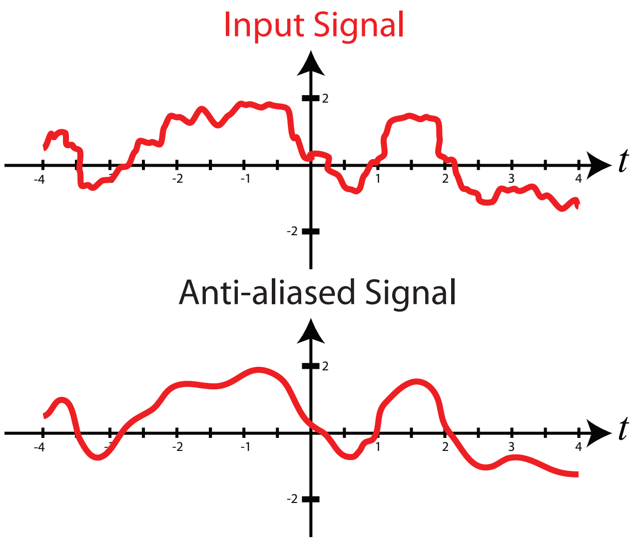

# Digitizing Audio




Anti-aliasing Filter:

- ensures that analog audio input does not contain frequency components higher than half of the sampling frequency (to avoid aliasing)
- Example: C6713 DSP, $F_s = 8$ kHz, therefore anti-aliasing filter must have a passband of 0 Hz to 4000 Hz.

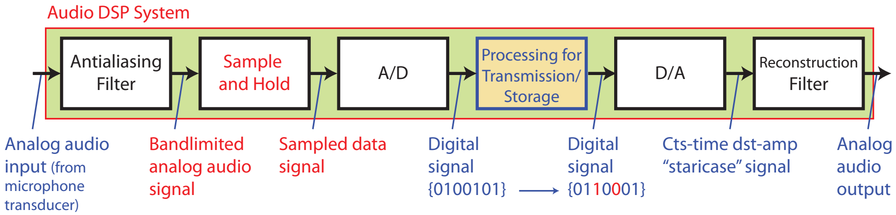
# Digitizing Audio








# Digitizing Audio



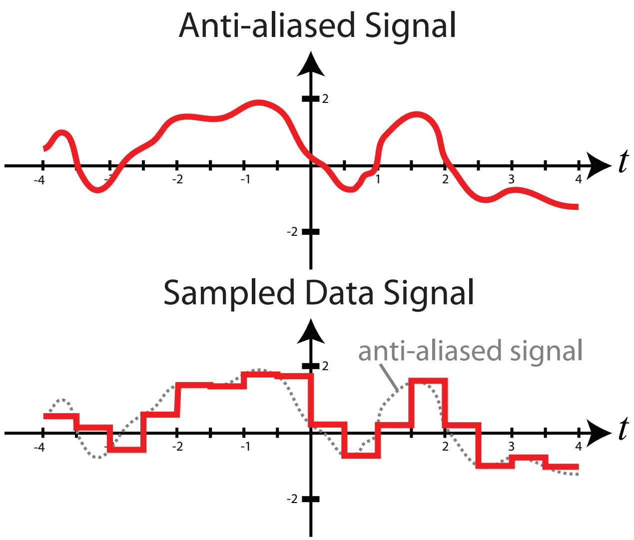

Sample and Hold:

- holds a sampled analog audio value for a short time while the A/D converts and interprets the value as a digital





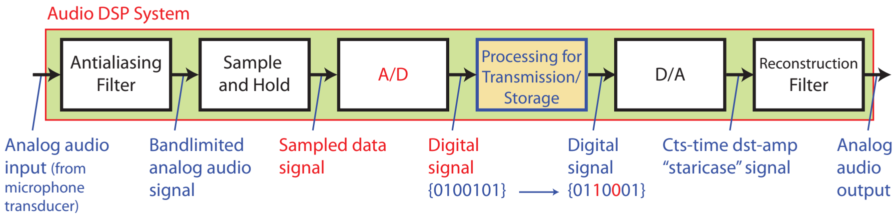
# Digitizing Audio







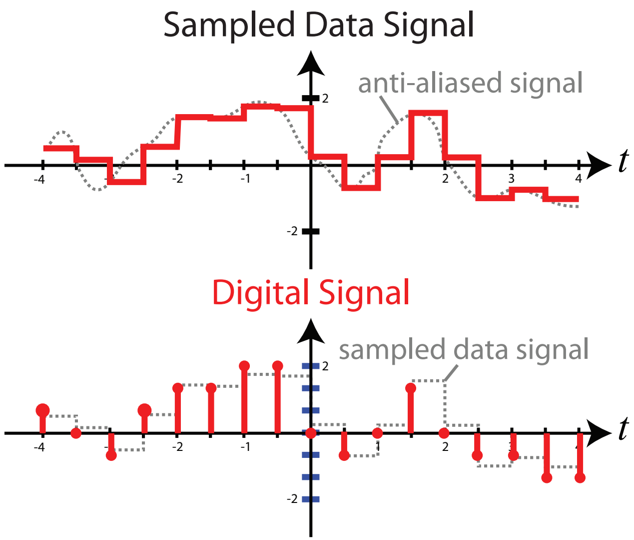

# Digitizing Audio




A/D:

- converts a sampled data audio value into a digital number, in part, through quantization of the amplitude

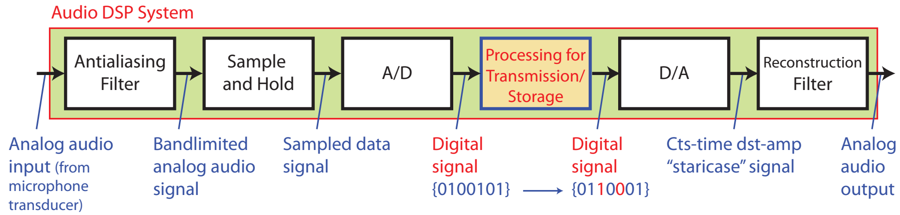
# Digitizing Audio








# Digitizing Audio




Processing for Transmission/Storage:

- transmission/storage contains inherent non-idealities that cause errors in the received/retrieved data symbols
- error correction coding (ECC) is employed to add redundancy to the digital signal so that errors can be compensated for during decoding





# Error Correction Coding

Example: $N$-repetition code

| Input Signal Bit | Coded Sequence |
|---|---|
| 0 | $\underbrace{0\ 0\ 0\ \cdots\ 0}_{N \text{ zeros}}$ |
| 1 | $\underbrace{1\ 1\ 1\ \cdots\ 1}_{N \text{ ones}}$ |

Therefore, for $N = 3$ the following input signal sequence: 0 0 1 would be coded as follows: 0 0 0 0 0 0 1 1 1.





# Error Correction Coding

**Q:** How would you interpret receiving the following coded sequence (with possible error):

1 1 1 0 1 0 0 0 0?

$$\underbrace{1\ 1\ 1}_{1}\quad \underbrace{0\ 1\ 0}_{0}\quad \underbrace{0\ 0\ 0}_{0}$$

**A:** Decoding can make use of majority vote logic.

# Error Correction Coding

Coder for $N = 3$:

| Input Signal Bit | Coded Sequence |
|---|---|
| 0 | 0 0 0 |
| 1 | 1 1 1 |

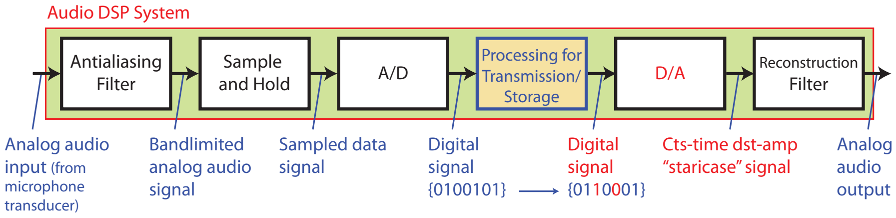
Majority vote logic decoder for $N = 3$:

| Received Coded Seq | Decoded Signal Bit |
|---|---|
| 0 0 0 | 0 |
| 0 0 1 | 0 |
| 0 1 0 | 0 |
| 0 1 1 | 1 |
| 1 0 0 | 0 |
| 1 0 1 | 1 |
| 1 1 0 | 1 |
| 1 1 1 | 1 |





# Digitizing Audio



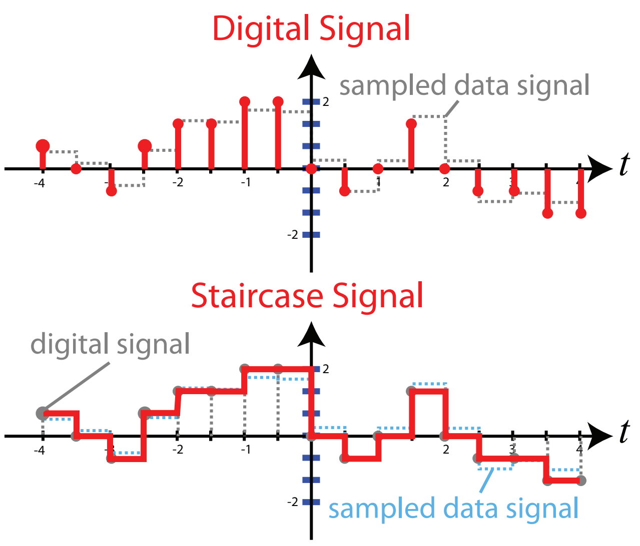

D/A:

- converts a digital audio signal into a "staircase"-like signal for further reconstruction





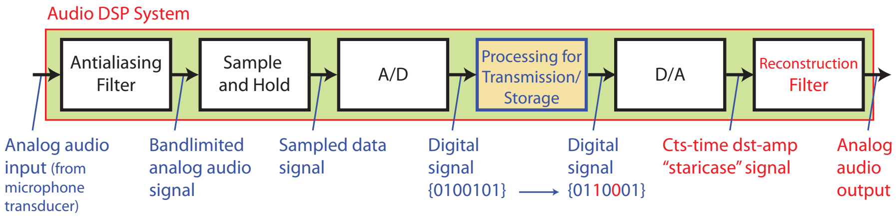
# Digitizing Audio







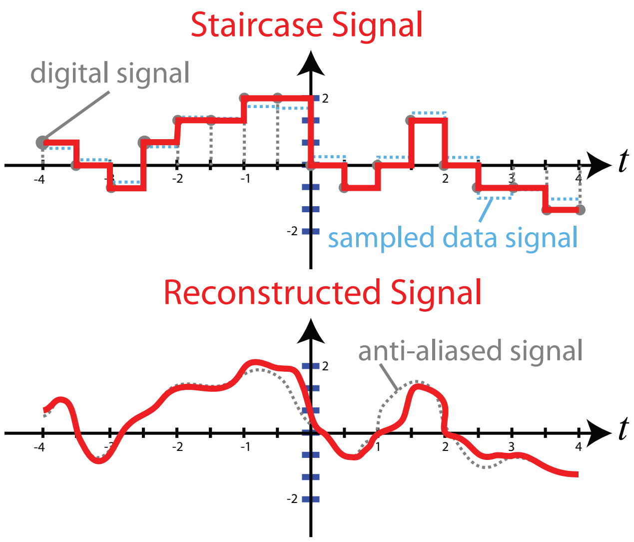

# Digitizing Audio




Reconstruction Filter:

- converts a "staircase"-like signal into an analog filter through lowpass filtering
- depending on the application the filter can be similar to the anti-aliasing filter, or may be very cheap (e.g., compact disk receivers), or may using a different sampling rate for special effects

# Digitizing Audio








# Digitizing Audio

The "quality" of digitizing audio is related to the following parameters:

- sampling rate (Hz)
- bit depth (bits/sample) and dynamic range (related to number of quantization levels)
- mono vs. stereo





# Digitizing Audio

Note: For the same cost, digital audio provides higher signal-to-noise ratio or lower mean-square error between the real sound and what is recorded/played.

- It is less expensive to increase sampling rate and quantization depth (i.e., reduce quantization noise) than to use less noisy analog circuitry (i.e., reduce noise floor)
- When signals are represented digitally the natural noise in the circuits can be circumvented via error correction coding. Thus, it is possible to have near perfect storage/transmission.





# Audio Quality and Sampling Rate

Audio Quality as a Function of Sampling Rate:

| Sampling Rate (Hz) | Quality Similar to |
|---|---|
| 8,000 | telephone |
| 11,025 | AM radio |
| 22,050 | FM radio |
| 44,100 | CD |
| 48,000 | DAT |

## Audio Quality, Sampling Rate, and Bit Depth

Audio Quality as a Function of Sampling Rate, Bit Depth and Stereo/Monophony:

| Sampling Rate (Hz) | Bit Depth | Stereo/Mono | Quality |
|---|---|---|---|
| 8,000 | 8 | mono | telephone |
| 11,025 | 8 | stereo | low |
| 22,050 | 8 | stereo | · |
| 22,050 | 16 | mono | · |
| 22,050 | 16 | stereo | · |
| 44,100 | 16 | mono | good |
| 44,100 | 16 | stereo | CD quality |





## Audio Quality

**Q:** Why do some people insist that analog audio is superior to digital audio?

**A:** What they think sounds good isn't the exact original sound, but a nonlinearly distorted version generated from the analog components.

Note: Some digital audio companies now make digital amplifiers that mimic the distortion from analog audio amplifiers.

Quality of audio is a qualitative and psychological measure that is user-specific.





## Audio Equalization

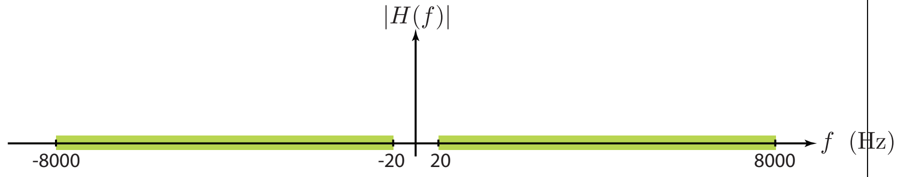
- Equalization $\equiv$ Equalisation $\equiv$ EQ
  - amplifying or attenuation different frequency components of an audio signal
  - Example: bass/treble control in inexpensive car radios
- Common goals of equalization:
  - provide fine granularity of frequency amplification/attenuation control without affecting adjacent frequencies.
  - correct for unwanted frequency attenuation/amplification during recording processes
  - enhancing the presence of certain sounds
  - reducing the presence of unwanted signals such as noise





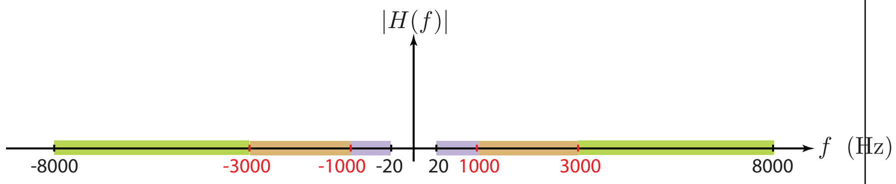
## Equalizer Design Basics

1. Determine the processing band of your audio signal.
   - human audible range is: 20 Hz to 20 kHz
   - if sampling rate of a DSP is $F_s$ then, the bandwidth of the audio signal to process is: 20 to $\frac{F_s}{2}$ Hz
   - Example: $F_s = 16,000$ Hz



# Equalizer Design Basics

2. Determine the granularity of your equalizer (i.e., number of frequency bands to independently control).
   - one approach might be to equally partition the audio signal bandwdith
   - more popular approaches suited to human auditory system models have bands that increase in width by two
   - Example: 3 frequency bands

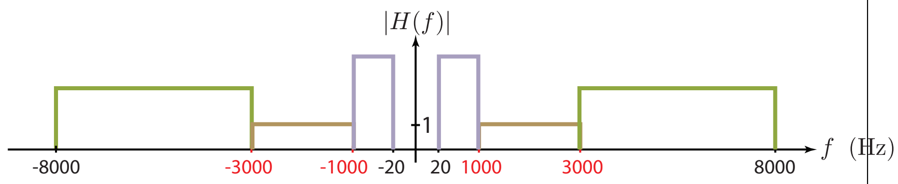







# Equalizer Design Basics

3. Design your bandpass filters.
   - each bandpass filter is independently set/controlled from the others
   - ideally, many people would like shelving EQ
   - Example: Ideal bandpass filters

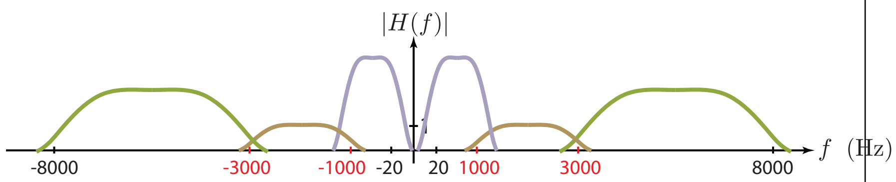







# Equalizer Design Basics

3. Design your bandpass filters.
   - each bandpass filter is independently set/controlled from the others
   - ideally, many people would like shelving EQ
   - Example: Bell EQ

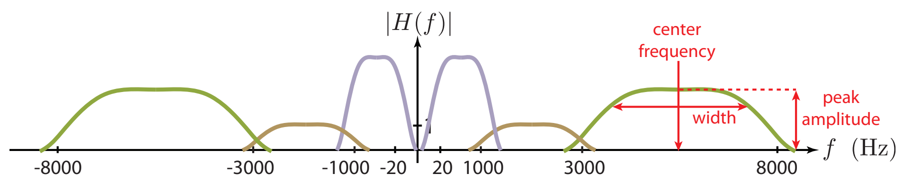







# Common Types of Equalizers

- All bell filters and many other bandpass filters can be characterized by three parameters:
  - center frequency
  - width of the bell curve
  - gain (i.e. peak) of the bell curve



# Common Types of Equalizers

- Parametric Equalizers: the center frequency, passband width and peak amplitude can be independently selected for each filter
  - most powerful EQ, predominantly used for recording and mixing
- Graphic Equalizers: the center frequency and passband width of each filter are pre-set; the gains of each filter can be independently controlled
  - used for live applications such as concerts





# Common Types of Equalizers

- Notch Filters: the passband width is small and fixed for each filter; center frequencies and gains are variable.
  - used in multimedia applications/audio mastering

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