

# A brief introduction to audio systems

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# What are some of the parts of audio systems?

- Sound waves — generated by mechanical vibrations of some sort, travel through the air, and detected by our ears
- Microphones to convert sound waves to electrical signals.
- Methods for storing audio information.
- Circuits and systems to manipulate audio information (signal processing)
- Circuits to amplify audio signals to higher power levels.
- Speakers (and other transducers) to convert electrical signals back into sound waves.

## Design considerations

- Power / efficiency
- Distortion
- Frequency response & signal processing

# Audio frequencies

- All sound involves oscillations — a periodic wagging of some physical quantity (position, sound pressure level, voltage). The frequency of oscillation is a key feature of the sound — frequency is the number of oscillations per second, denoted in units of hertz (Hz).
- Humans can hear sounds at frequencies between 20 Hz and 20,000 Hz (20 kHz). Frequencies below are “infra-sonic” (elephants) and those above are “ultra-sonic” (dogs). (Note, that the human range of hearing changes with age.)
- Low frequencies are known as “bass”, high frequencies as “treble” and in between as “mid-range”.
- All aspects of an audio system should work properly for all frequencies in this range.
- When we measure the performance of an audio system, we often describe it in terms of a *frequency response* — how does the property vary as a function of frequency.

# Audio frequencies

Sine wave

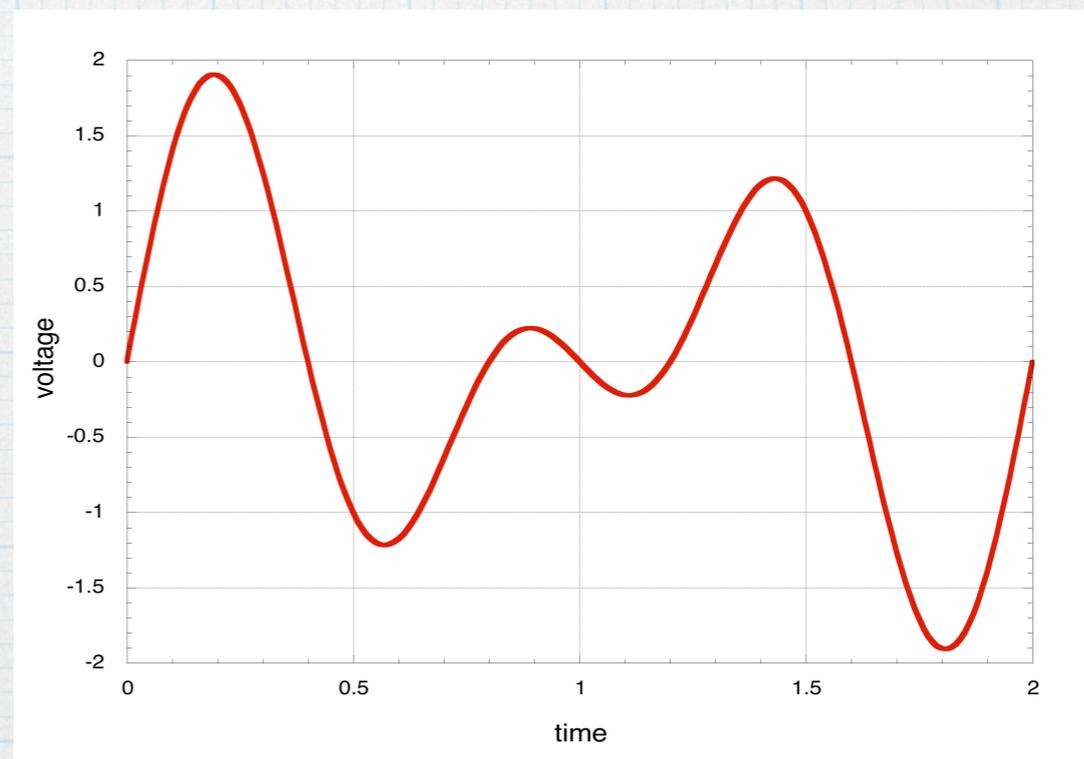
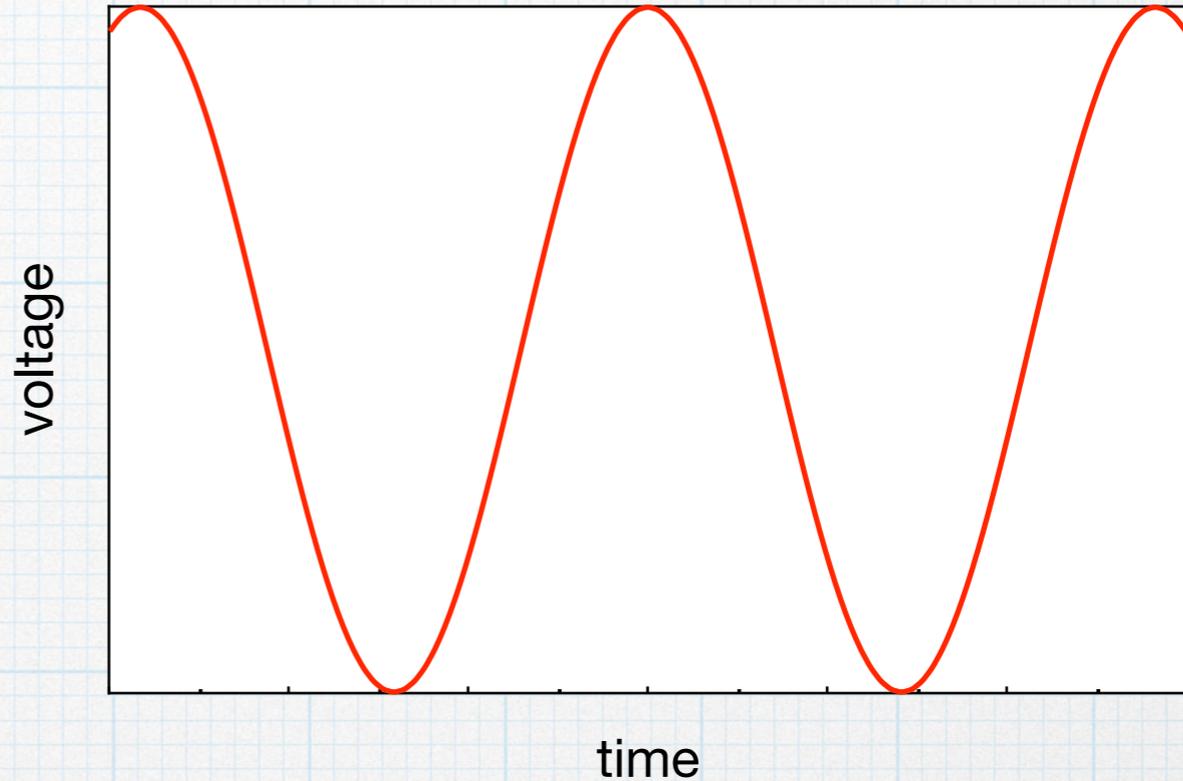
$$v(t) = A \sin(2\pi ft)$$

$f$  is the frequency

$$v(t) = A \sin\left(2\pi \frac{t}{T}\right)$$

$T$  is the period =  $1/f$ .

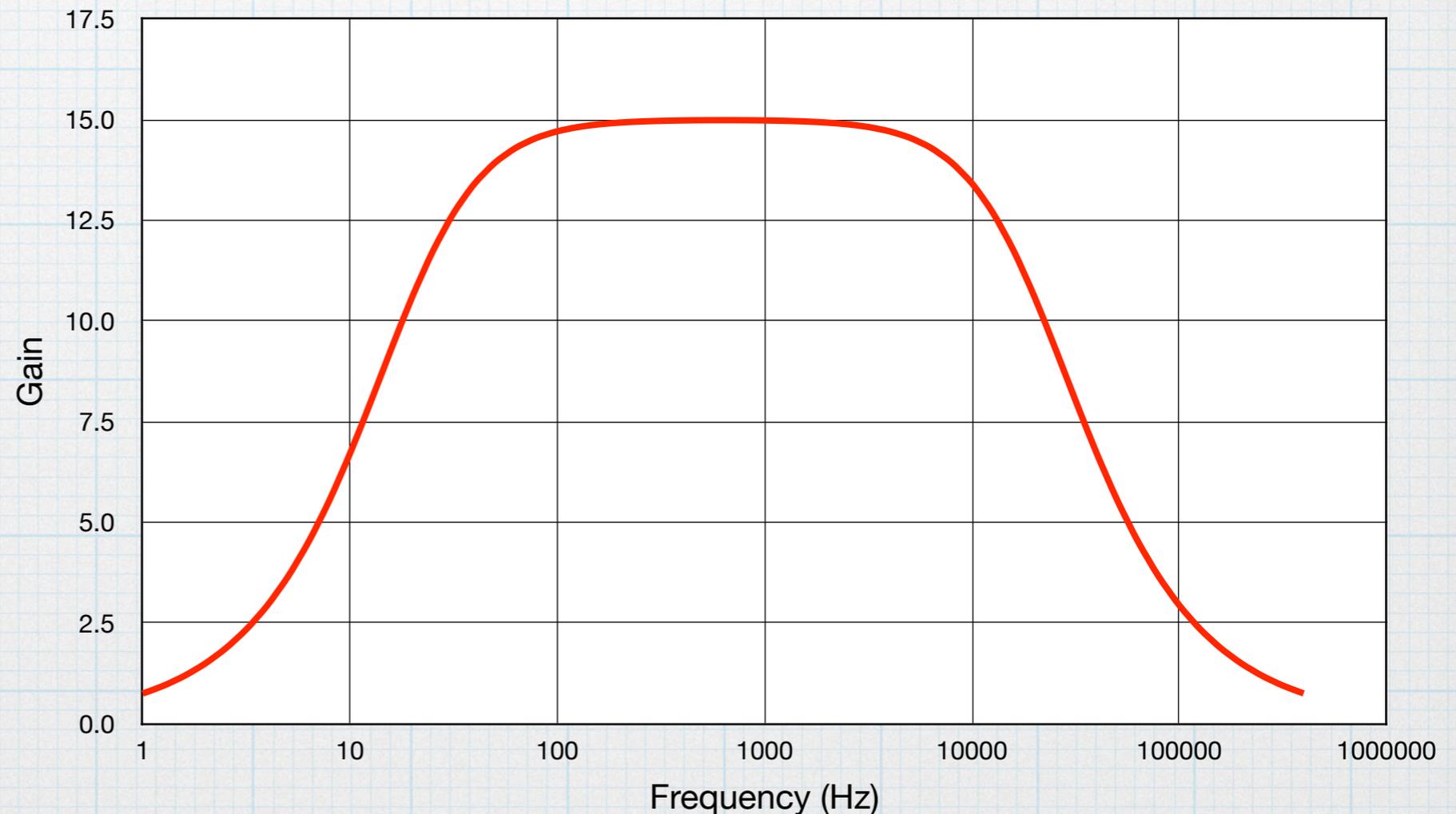
Fourier: *All* audio signal can be represented using combinations of sine waves.



# Audio frequencies

- When we measure the performance of an audio system, we often describe it in terms of a *frequency response* — how does the property vary as a function of frequency.
- For example, when we build an amplifier, we should amplify all of the audible frequencies equally.

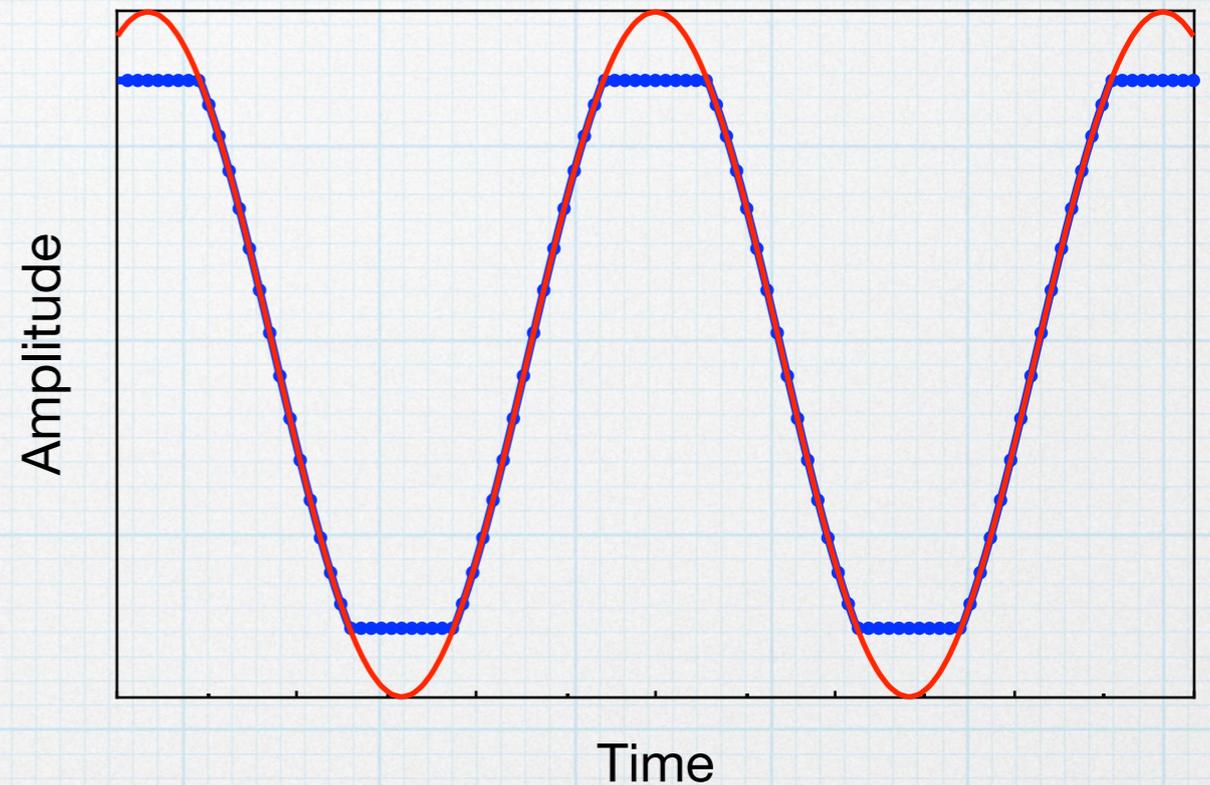
Plot of gain vs. frequency for an amplifier. In the audible range, the gain is “flat” (mostly). And we don’t care what happens at higher and lower frequencies.



# Distortion

- As the audio signal passes through the various parts of the system, the shape of the signal should not change. Any change is *distortion*, which represents a change in the information contained in the signal. (A song will sound different.)

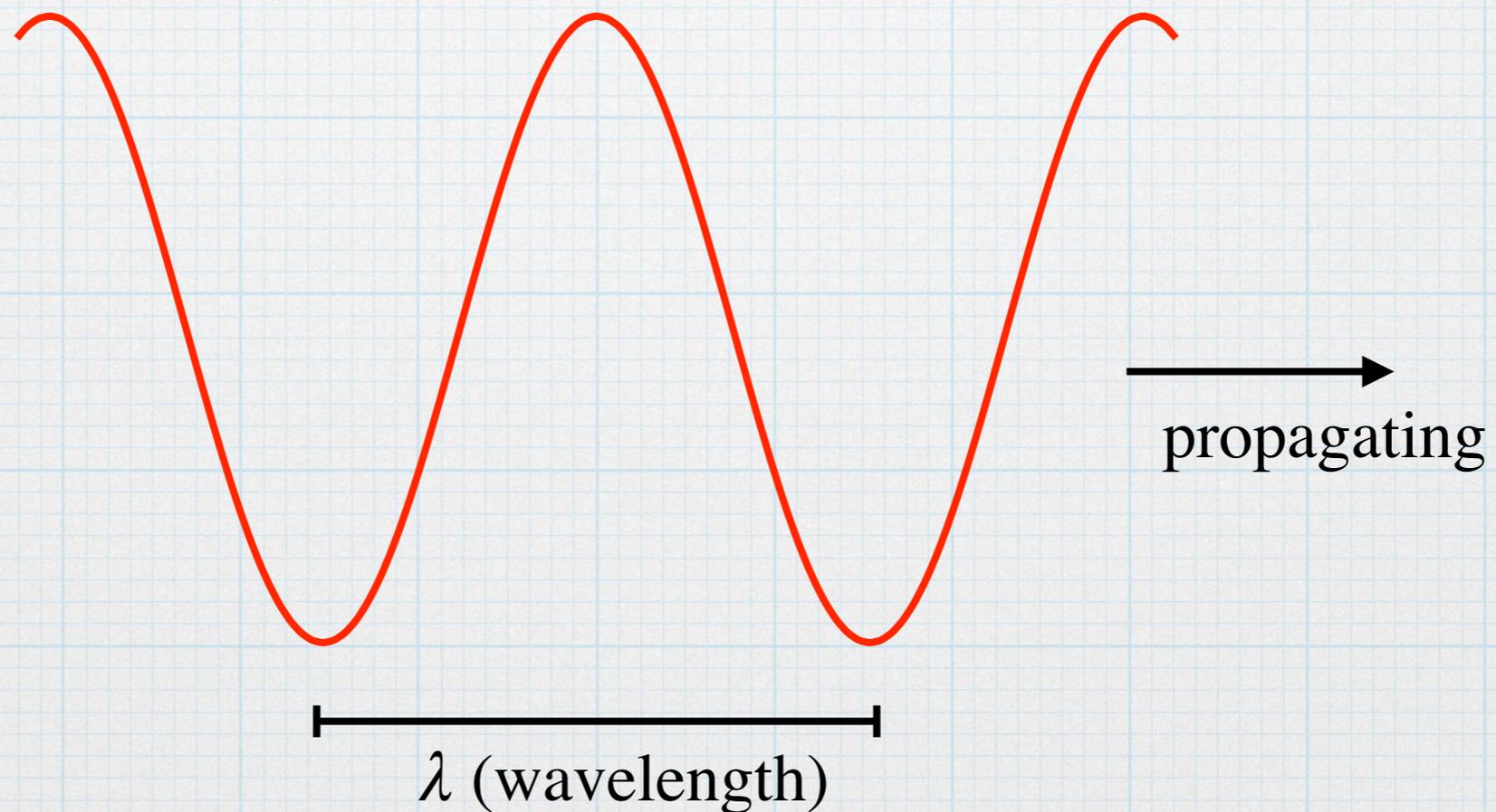
- For example, a common type of distortion that can occur with amplifiers is “clipping”, where the high and low parts of a signal can be cut off.



- There are many other types of distortion. Generally distortion should be avoided.
- One exception is in making guitar pedals, which are designed specifically to distort the signal coming from an electric guitar.

# Sound waves

- The audio signal travels as energy through the air as sound waves — moving pressure variations.
- All waves have a characteristic wavelength that is related to the frequency and speed that wave at which the wave travels,  $\lambda = v / f$ . ( $v = 343 \text{ m/s} = 767 \text{ mph}$ ).
- At  $f = 20 \text{ Hz}$ ,  $\lambda = 17 \text{ m} = 56 \text{ ft}$ . At  $f = 20 \text{ kHz}$ ,  $\lambda = 1.7 \text{ cm} = 0.67 \text{ inches}$ .



# Sound waves

- The three-orders-of-magnitude difference in the wavelengths has important implications for how we perceive sounds. The short wavelengths of the high-frequency sounds mean that they can be viewed as traveling almost like rays. (Similar to viewing light waves as rays.) This means that we can discern the direction the high frequency sounds comes from.
- The long wavelengths of low frequencies make determining directions very difficult.
- The wave nature of sound also means that reflections will lead to constructive and destructive interference. So the shape of a room — and hence the shape of the reflections — can have big impact on what we hear.

# Some important considerations

- Our ears are analog — they respond to a continuous series of pressure vibrations arriving from the surrounding air.
- Most modern audio storage is digital — numbers stored in some form of electronic memory element. Needs to be converted to analog.
- Because we have two ears, we are able to “image” sound and determine the direction that it is coming from. (At least for some frequencies.)
- The range of “loudness” (acoustic power) that we can detect and tolerate is huge. If we assign the smallest possible sound that a typical human can hear as 1, then the loudest that we can handle is  $10^{12}$  (one trillion). (Above this, our ears may be damaged.) Using technical jargon, the range is 120 dB.

# Analog

- All sounds are produced by mechanical vibrations of some sort, and these are all *analog* processes, meaning that there is a continuous of variation of some physical variable — position, sound pressure level, etc.
- Our ears are also analog — the continuously varying sound-wave pressure level produces a corresponding vibration in our ear drums. Ultimately, these vibrations are detected by our nervous system and sent to the brain for processing. (Note: With two ears, we are able to “image” sound and determine the incoming direction.)
- Until the 1980s, all aspects of an audio — including the storage media — were analog.



# Digital

- However, as digital technology developed, we learned that storing audio in digital form offered many advantages over analog storage methods. Digital storage devices were smaller and digital data is easier to replicate and transmit.
- Eventually, we just started storing audio files on our computers, treating them just like any other digital data on the hard drive.
- Then also on our phones.
- And now in the cloud.

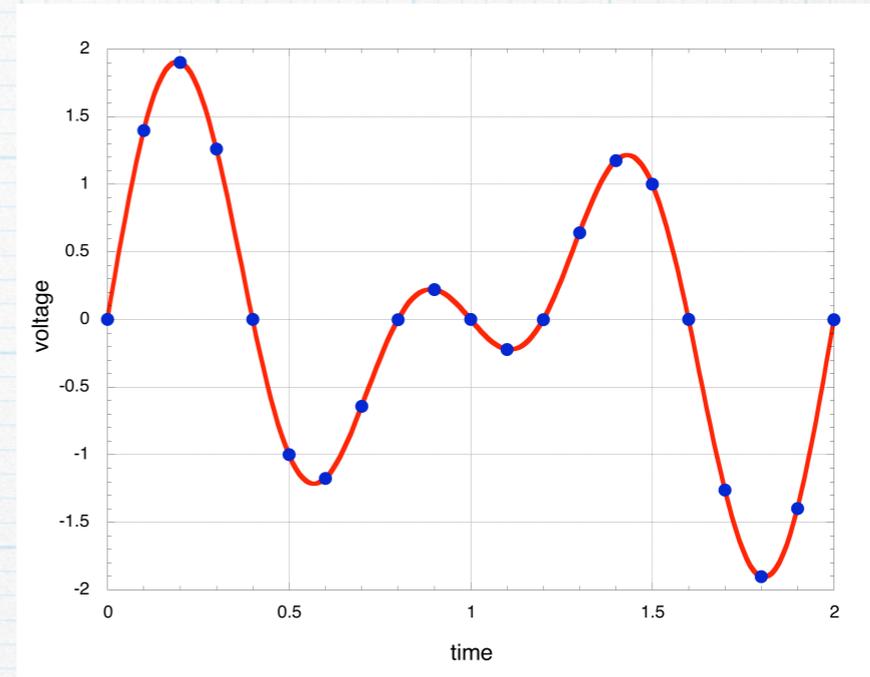


# Analog - Digital Conversion

- All the advantages of digital storage and signal manipulation do not change the fact that sound generation and sound detection in our ears are analog processes.
- To join the analog necessities with the digital conveniences and advantages, there must be converter circuits.
- Analog-to-digital converters (ADCs) take analog signals and generate a stream of digital data — binary numbers in the form of ones and zeros.
- Digital-to-analog converters (DACs) take streams of binary numbers and turn them into analog signals that can be used to produce sound waves.
- ADCs and DACs are essential components in most any modern audio systems.

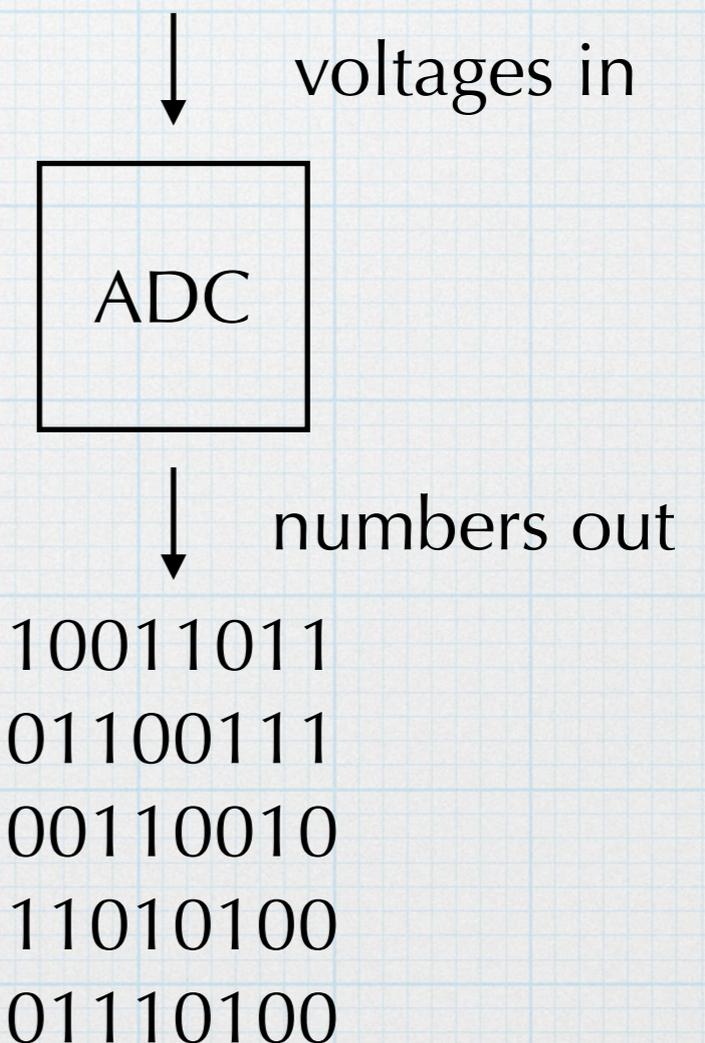
# Analog-to-digital conversion

A continuously varying analog signal is *sampled* at various points in time.



The samples are fed into an ADC circuit that produces binary voltages

that are interpreted as a stream of binary numbers, which can be manipulated and stored with digital techniques.



# Digital-to-analog conversion

A stream of binary numbers is read from digital memory

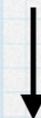
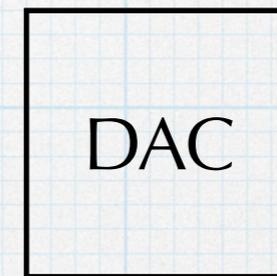
and fed to a DAC circuit, which converts the data stream to

a continuously varying analog signal.

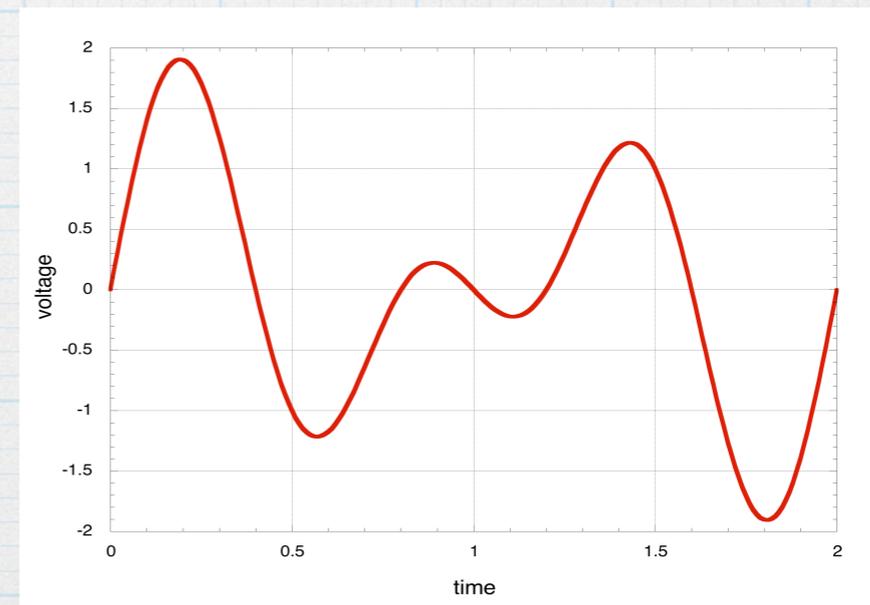
10011011  
01100111  
00110010  
11010100  
01110100



numbers in



voltages out



# Transmission to the amplifier

- Traditionally a wire was used, and still very common. Still probably the best way to transmit the signal with no distortion or data loss.
- However, an alternative method has become available in recent years — wireless transmission using Bluetooth, Wifi, or some other proprietary communication scheme (eg Sonos). With modern wireless systems, the data is transmitted in digital form, so the D-to-A conversion takes place *after* transmission.
- Requires all the elements of a standard communication system — transmitter and receiver circuits and antennas at both ends.
- There are analog transmission systems still in use — AM and FM radio.
- There are also optical fiber transmission systems which send the info via light pulses. These also require transmitters and receivers.
- Wireless transmission often requires some sort of data compression, so there is possibility of information loss.

# Amplifiers

- Typically, the audio signals coming from the source are too weak to drive most speakers. Making a “normal” desktop or living-room speaker give out some sound requires at least a few tenths of a watt of power. The audio signal from a DAC might have only a few milliwatts of power available. The amplifier adds power to the signal. (Remember:  $P = V \cdot I$ , so both voltage and current must be boosted.)
- The amp takes energy from a DC power supply and adds it to the audio signal. The DC power can come from a battery or from an AC-to-DC converter that is plugged into a wall outlet.
- Power. More power requires bigger power supplies and bigger components. This adds cost and size.
- Efficiency. If running from a battery, efficiency is crucial. Inefficient amps may require significant heat sinking to protect from getting too hot. This also adds to cost and size. (class A vs. class B vs. class D.)
- Frequency response. As discussed, an amp should boost all frequencies by the same amount.
- Distortion. Amplifiers are inherently non-linear, so distortion is always a concern.

# Speakers

- Once the audio signal has enough power (voltage and current), it can be used to drive a speaker.
- The speaker is essentially a linear motor, converting electrical energy into mechanical energy and then into acoustic energy.
- Speaker design requires consideration of several inter-related topics: filtering of the audio signal, magnetics, mechanical vibration, and interference of sound waves. For something that is so common and outwardly simple, the detailed operation is quite complex.
- Multiple drivers. Because a single driver cannot produce all frequencies (due to the limits of mechanical vibrations), a speaker may have multiple drivers of different sizes. Each size will work well for a particular range of frequencies. In this case, the electrical signal has to be modified so that only the appropriate frequencies are passed to each driver.
- Speakers are most effective when their physical size is commensurate with the wavelengths they are producing. While a 56-ft diameter speaker is not practical — although it would be awesome! — the general trends hold. Small diameter speakers are used to produce higher-frequency sounds and bigger speakers produce lower sounds. A three-way system is common, a small “tweeter”, a medium-sized “mid-range”, and a big “woofer” or “sub-woofer”.

# Speakers

- Power. The coil, magnet, and speaker cone must be sized to match the desired acoustic power output — more power = bigger (and more expensive) components.
- Distortion. Like with the amplifier, the speaker should not change the signal as it is changing from one form to another.



# Stereophonic sound

- The fact that we have two ears separated by the width of our head means that sounds reach our two “sensors” at slightly different times. We can use the time difference to determine the direction that the sounds comes from, at least for higher frequencies.
- When listening to a live performance, we can discern where the drums are, where the strings are located, and where the horn section is seated on the stage.
- We can use two speakers (stereo) when playing recorded music to “fake” the effect of listening to a live performance. It is easy to separate the various sounds from a live performance and record those into two separate “channels” — some sounds go into the “right” channel and some into the “left”. If the two channels are played back through two speakers that are spaced apart in front of us, we will hear slightly different sounds coming from each speaker. Our brain will combine the sounds in such a way that we perceive a spread-out orchestra in front. It is an illusion, but a very potent one. Almost all music is recorded in “stereo” for this reason. (If the sounds are not separated, then the audio is said to “mono”).
- The effect can be extended to more speakers — a quadraphonic system or surround sound that comes with video. But stereo is the most common for straight audio applications.

# To review:

For any audio system, we need:

- Microphones to record sound.
- Stored stereo audio source
- Digital to Analog conversion (probably)
- Transmission to an amplifier (wire or bluetooth)
- Amplifier
- Speakers
- Transmission through the air to the listener's ear

Each of these things will affect the quality of the sound we hear. The important factors are distortion and frequency response.

Which is most important?