Audio & amplifiers

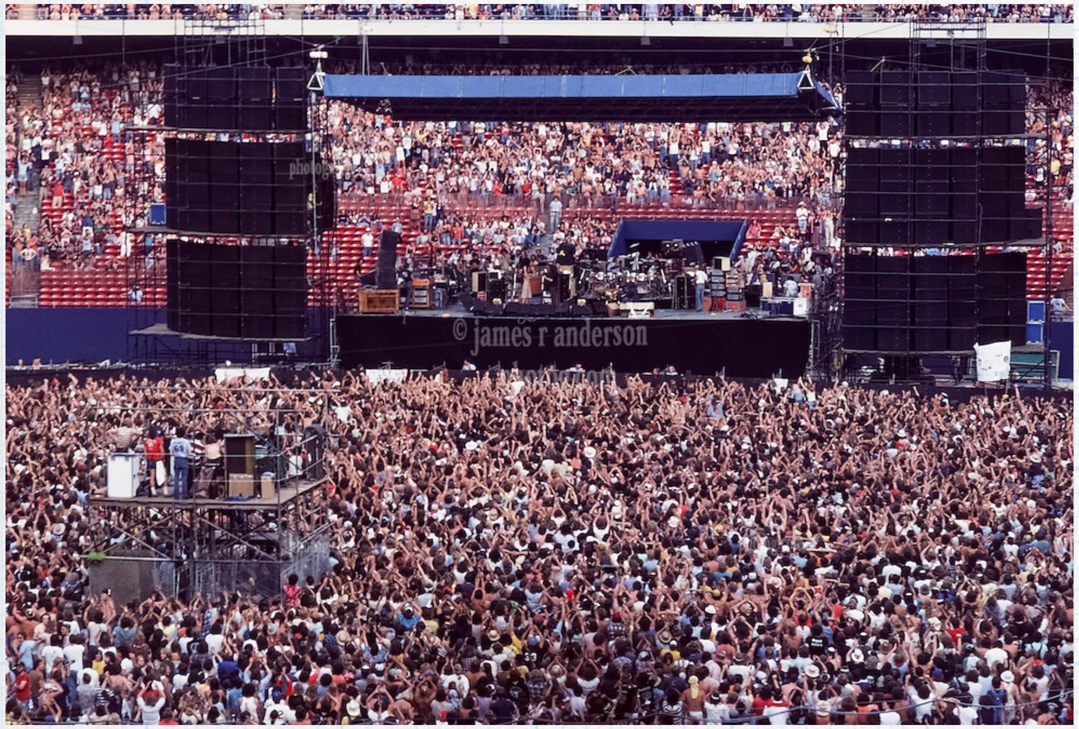
ISU Audio/Arduino meeting
Oct 6, 2025
Gary Tuttle

- Announcements
- Teach-yourself Arduino
- Audio systems - general considerations
- Amplifiers - efficiency, power output, frequency response, and distortion
- Project — GTDT amp
- Project — Cyduino

Arduino tutorials

- New page has been added to the website: Knowledge
- Presentation slides, articles, links to other websites, maybe videos
- Arduino tutorials
  - Serial monitor
  - Timing
  - Digital out
  - Digital in
  - Analog in
  - PWM
  - Serial interfaces — UART, SPI, I2C
  - Displays — LCD, OLED, LED
  - Program loading — Important!
  - Real-time clocks (external timing)
  - Interrupts

Audio systems



Hearing audio – step by step

You hear some sound coming from an audio system. Work backwards to see what all is involved.

- Your ears respond to the sound waves impinging on your eardrum, causing it to vibrate.
- Sound waves travel through the air.
- Some type of speaker is vibrating to generate the sound wave that travels outward.

An amplifier boosts an electrical signal, giving it enough power to drive the speaker.

- There may be some type of *signal processing* that modifies the audio signal to change the way that it will sound when it eventually gets to our ears.
- There is some mechanism for converting the stored audio information into an electrical signal. If the information is stored in analog form (vinyl disc or magnetic tape), there must be a sensor that responds to the stored medium to create the electrical signal. For digital storage (i.e. information stored as numbers), there will be a *digital-to-analog converter* (DAC).

recording audio — step by step

You hear some good music, and you would like to record it. What is involved?

- Humans playing instruments (guitars, violins, pianos, trumpets, saxophones, drums) or singing create vibrations that move outward as sound waves. (Still true with AI? 🤷🏻‍♂️)
- The sound waves travel through the air.
- The sound impinges on a microphone, which converts the pressure variations into an electric signal — probably very weak.
- There is probably an amplifier that boosts the electrical signal.
- There may be some type of *signal processing* that modifies the audio signal. Presumably, this will change the information in the signal to make it "sound better" when someone eventually hears it.
- There is some mechanism for converting the electrical signal into a form that can be stored. If the information is stored in analog form (vinyl disc or magnetic tape), there must be a transducer that encodes the signal into the mechanical grooves of a vinyl record or the magnetic domains of cassette tape. For digital storage (i.e. information stored as numbers), there will be an *analog-to-digital converter* (ADC).

In the two scenarios described, there are some key components, some of which are common to both recording and producing sound.

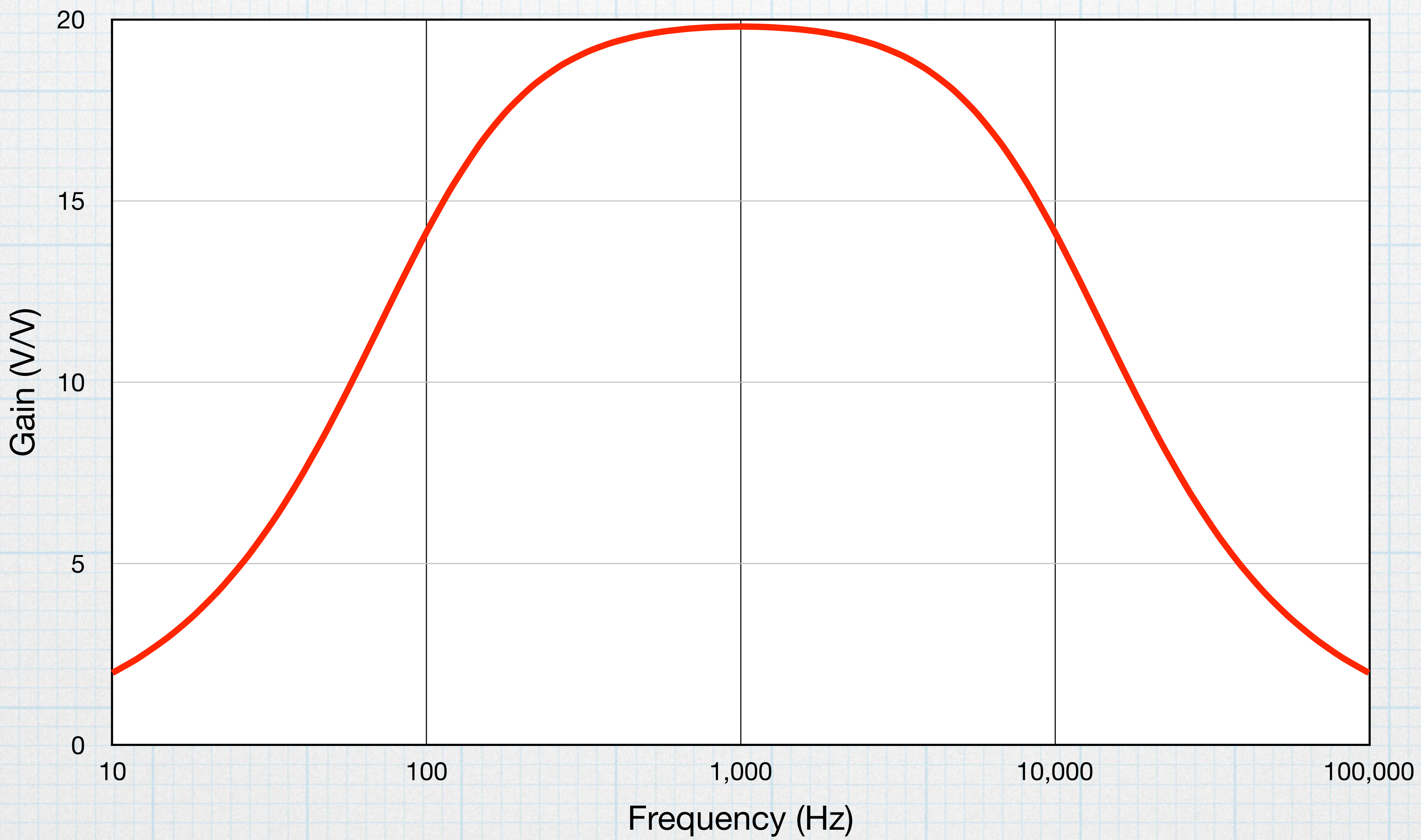
- Our ears. Probably the most important part. Not much we can do about them, other than protect them from damage and see a doctor if there is a problem.
- Sound traveling through the air. Not much we can do about, other than to reduce reflections in the room where we are recording or listening.
- Amplifiers. Somewhat important. Purely electronic. We will make several versions in Audio Club.
- Speakers. For listening to music, these are certainly the most important part.
- Microphone. For recording, certainly the most important part.
- Signal processing. Not necessary, but very commonly used in audio systems. Can improve the *perceived* quality of the sound. (Or make it worse.)
- Transducers to convert between electrical signals and analog storage media. Or if using digital storage, data converters to move information back and forth between digital and analog forms.

Audio frequencies

- All sound involves oscillations — a periodic variation 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 “infrasonic” (elephants can probably hear it), and those above are “ultrasonic” (dogs can hear it). Every human has slightly different hearing ranges, and it changes with age.
- Low frequencies are known as “bass”, high frequencies as “treble”, and in between as “mid-range”.
- In principle, every part of an audio system should treat every frequency exactly the same. In practice, this is rarely true, often because of physical limitations. But sometimes changes are introduced intentionally.
- 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?

Frequency-response plot

Below is a typical frequency response plot. This one shows the gain of some hypothetical amplifier. Ideally, the curve would be absolutely flat between 20 Hz and 20 kHz. This particular curve is “flat-ish” around 1000 Hz, but it is certainly not ideal.



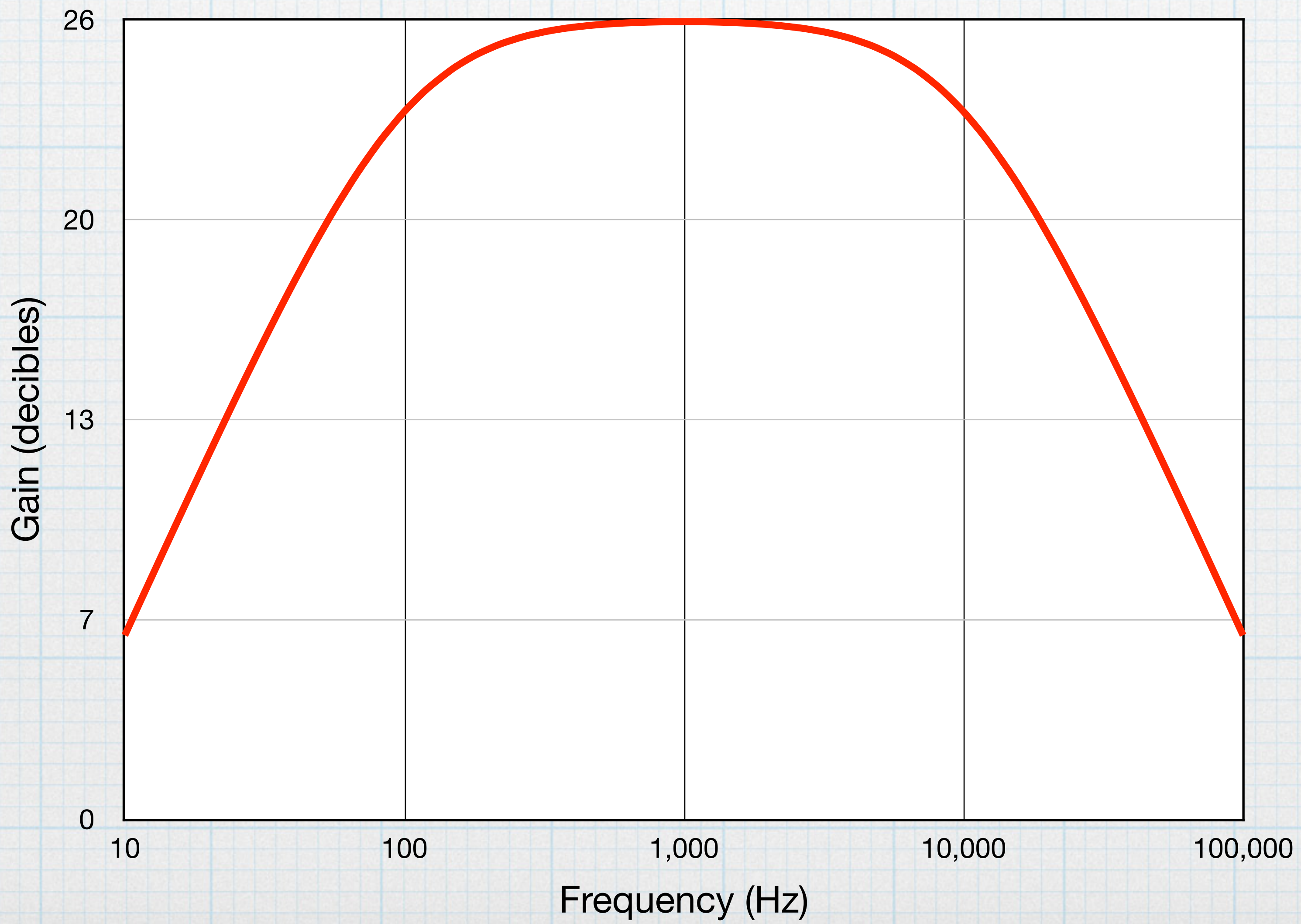
Because the human range of frequencies extends over 3 orders of magnitude, we us a logarithmic scale for the frequency axis.

Decibels

- Usually, when discussing audio quantities, we don't use absolute values (watts, voltage, etc.), but instead use *decibels*, which uses a ratio relative to some standard value.
- This is due to the extreme *dynamic range* over which our ears work.
- At the low end, we might be able to hear the sound of an ant walking across a surface. Above that level, our ears can hear sounds all the way up to that of something like standing near a jet engine at full throttle. (At which point, our eardrums would probably burst.) The ratio of two power levels is about 10^{12}!
- Again, with such a huge range, it is useful to use a logarithmic scale that describes the ratio of the power levels.
- In describing the ratio of power levels, 1 decibel = $10 \cdot \log_{10}(P_o/P_r)$, where P_o is the power being measured and P_r is the reference level. Usually $P_r = 10^{-12}$ W.
- In describing the ratio of voltage levels (like gain of an amplifier), 1 decibel = $20 \cdot \log_{10}(V_o/V_s)$, where V_o is the output voltage and V_s is the source voltage. The factor of two difference is due to $P \propto V^2$.

Bode

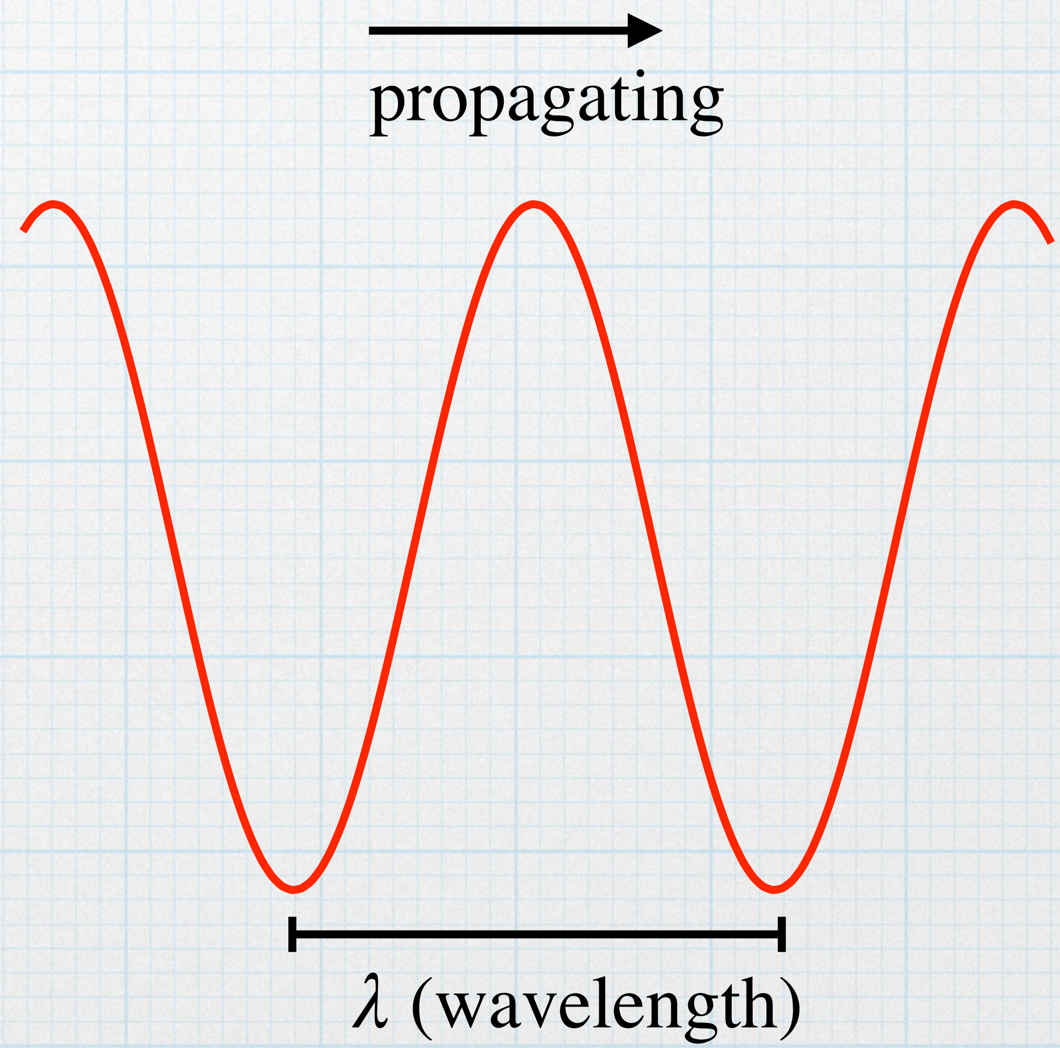
- It is common practice to make frequency-response plots using decibels on the vertical scale rather than a linear unit. In effect, the graph is a log-log plot.
- This is generally known as a Bode plot.
- Taking the logarithm tends to flatten the data, and maybe gives a better graphical view of the useful part of the curve.
- The graph below contains exactly the same information as the one on slide 8.



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 at which the wave travels, $\lambda = v / f$. (v = 343 m/s = 767 mph).

20 Hz	17 m	56 ft
200 Hz	1.7 m	67 in
2000 Hz	0.17 m	6.7 in
20 kHz	17 mm	0.67 in



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 come 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 a big impact on what we hear.

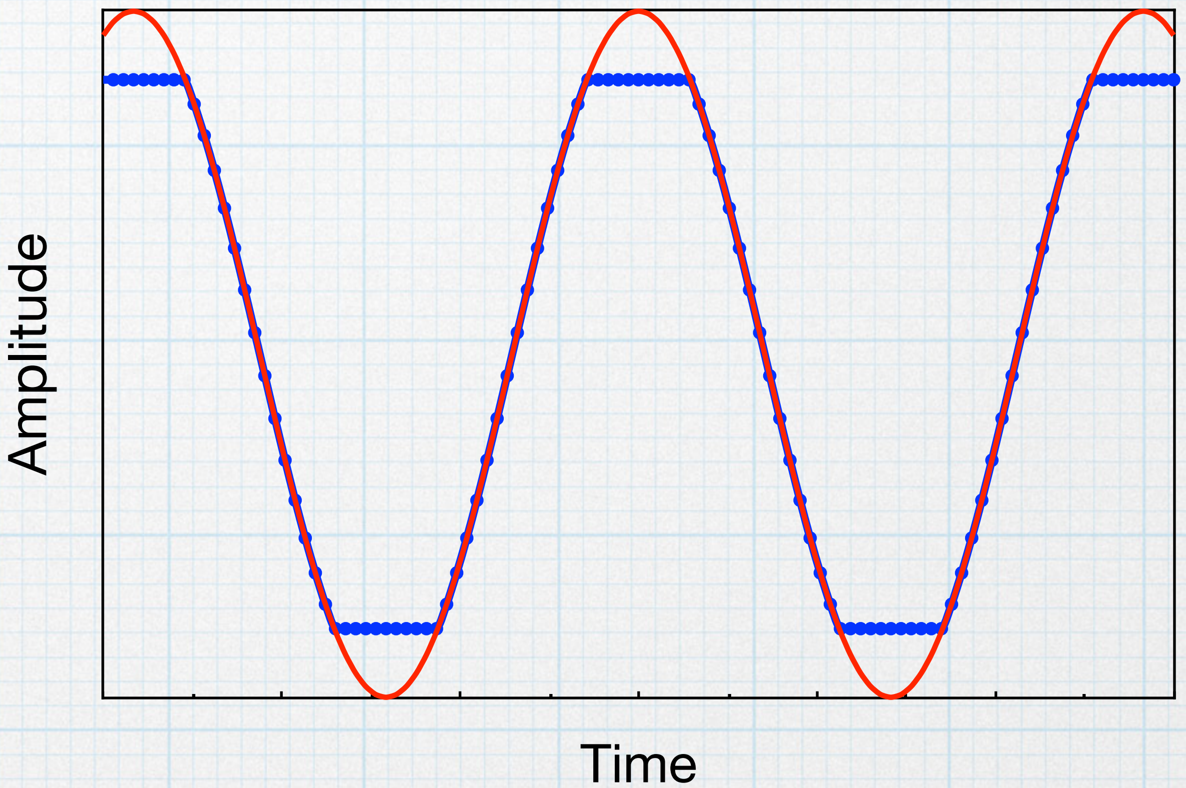
Stereophonic sound

- 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 come 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 be 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.

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 t he 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.

Objective vs. subjective sound quality

- Many aspects of a sound system can be measured using engineering techniques. We arrive at values that can be used to directly compare the performance of two different systems, e.g. frequency response, distortion, output power, efficiency. These are objective measurements, and everyone can agree on what the numbers mean in an engineering sense.
- However, listening to audio, particularly music, can be very much like tasting food — people will have different reactions to the exact same thing. This is the subjective characterization of sound quality.
- Some people like more bass (think electronic dance music), and some prefer more treble, like might be present in classical music. Sometimes people's taste in music quality can change over time. There is nothing wrong with different audio preferences.
- In the audiophile world, the distinction between objective and subjective evaluations can get mixed up, and people can have heated (even overwrought) arguments about the quality of audio systems.
- My approach is to try to make all components "sonically agnostic" — having flat frequency response and no distortion. The idea is to leave no room for argument about the objective properties of the system.
- Then the user can tune the sound produced by an "equalizer", which allows for enhancing or reducing the relative contributions of various frequencies.

Analog

- All sounds are produced by mechanical vibrations of some sort, and these are all *analog* processes, meaning that there is a continuous 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 eardrums. Ultimately, these vibrations are detected by our nervous system and sent to the brain for processing.
- Until the 1980s, all aspects of audio — including the storage media — were analog.

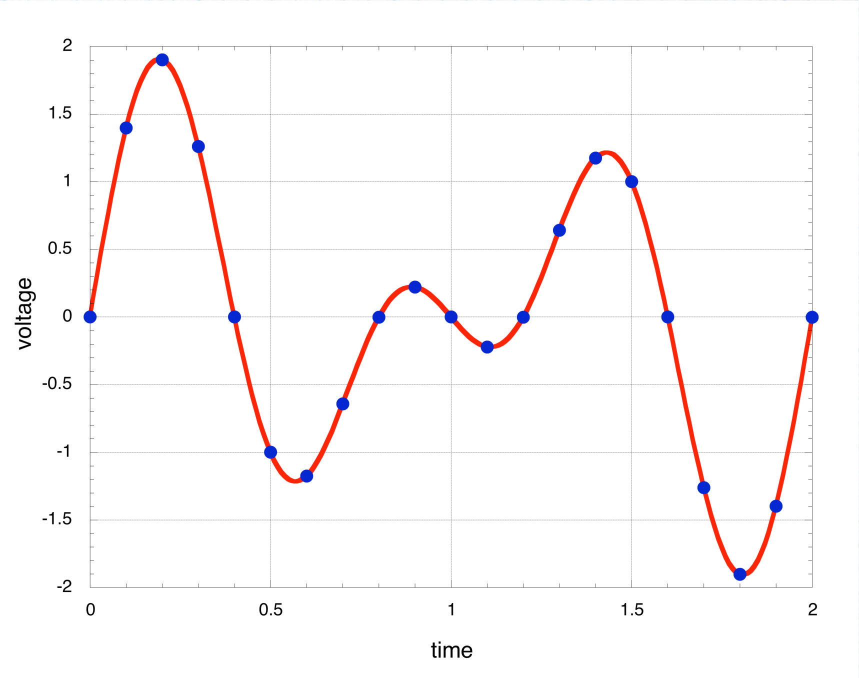
Digital

- 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 modern audio systems.

Analog-to-digital conversion



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

voltages in

ADC

numbers out

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

10011011
01100111
00110010
11010100
01110100

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

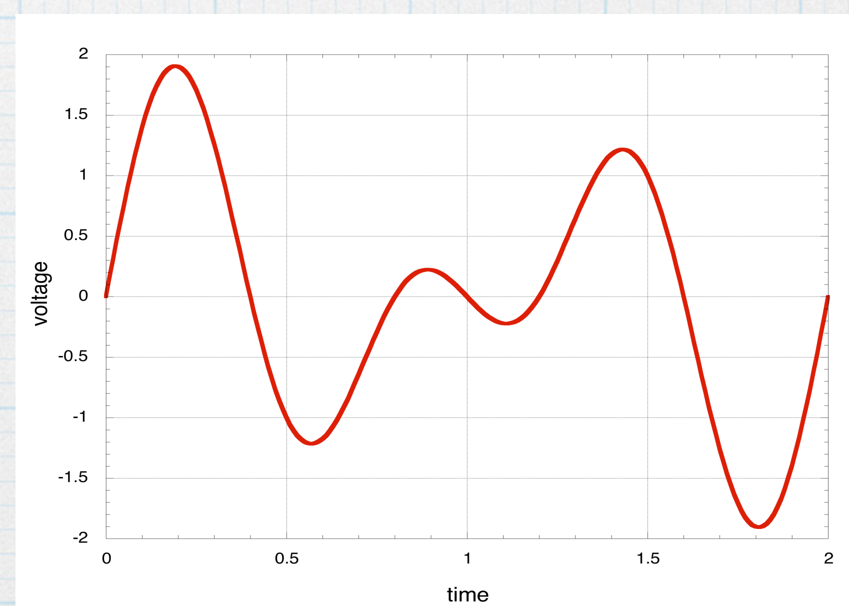
a continuously varying analog signal.

10011011
01100111
00110010
11010100
01110100

numbers in

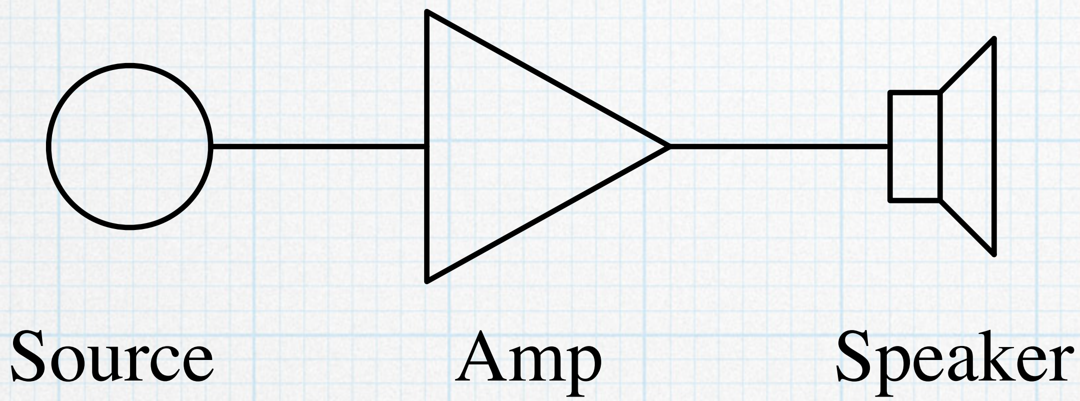
DAC

voltages out



Amplifier basics

The purpose of an amplifier is to increase the power in the audio signal in order to drive the speakers.



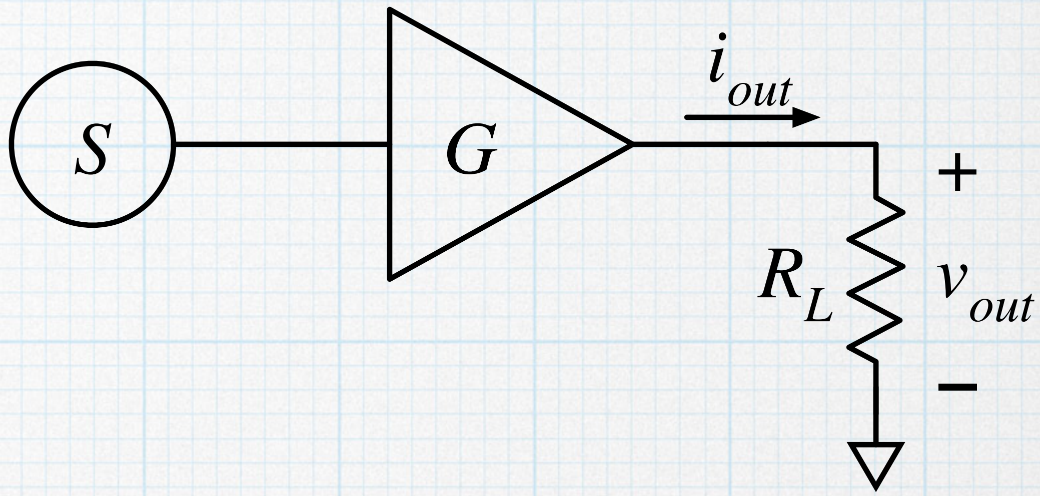
The power that is added to the audio signal comes from one or more DC power supplies, which must be adequately sized for the power level.

Key things to consider:

- Maximum output power.
- Frequency response.
- Distortion.
- Efficiency (power to speakers divided by power from DC supplies).
- Thermal control.
- Noise.
- Cost.

Power

In an audio system, the amplifier is probably delivering power to a speaker. Typical speakers have voice coils with either 4-Ω or 8-Ω resistance. (We will use 4 Ω as a standard.)



The resistive nature of the speaker imposes constraints on power.

$$P = v_o \cdot i_o$$

$$v_o = i_o \cdot R_L$$

$$P = \frac{v_o^2}{R_L} = i_o^2 \cdot R_L$$

For a 4-Ω speaker: (RMS quantities)

P	v_o	i_o
0.1 W	0.632 V	0.158 A
1 W	2 V	0.5 A
10 W	6.32 V	1.58 A
100 W	20 V	5 A

DC supplies must be able to supply the corresponding voltage and current!

GTDT amp project

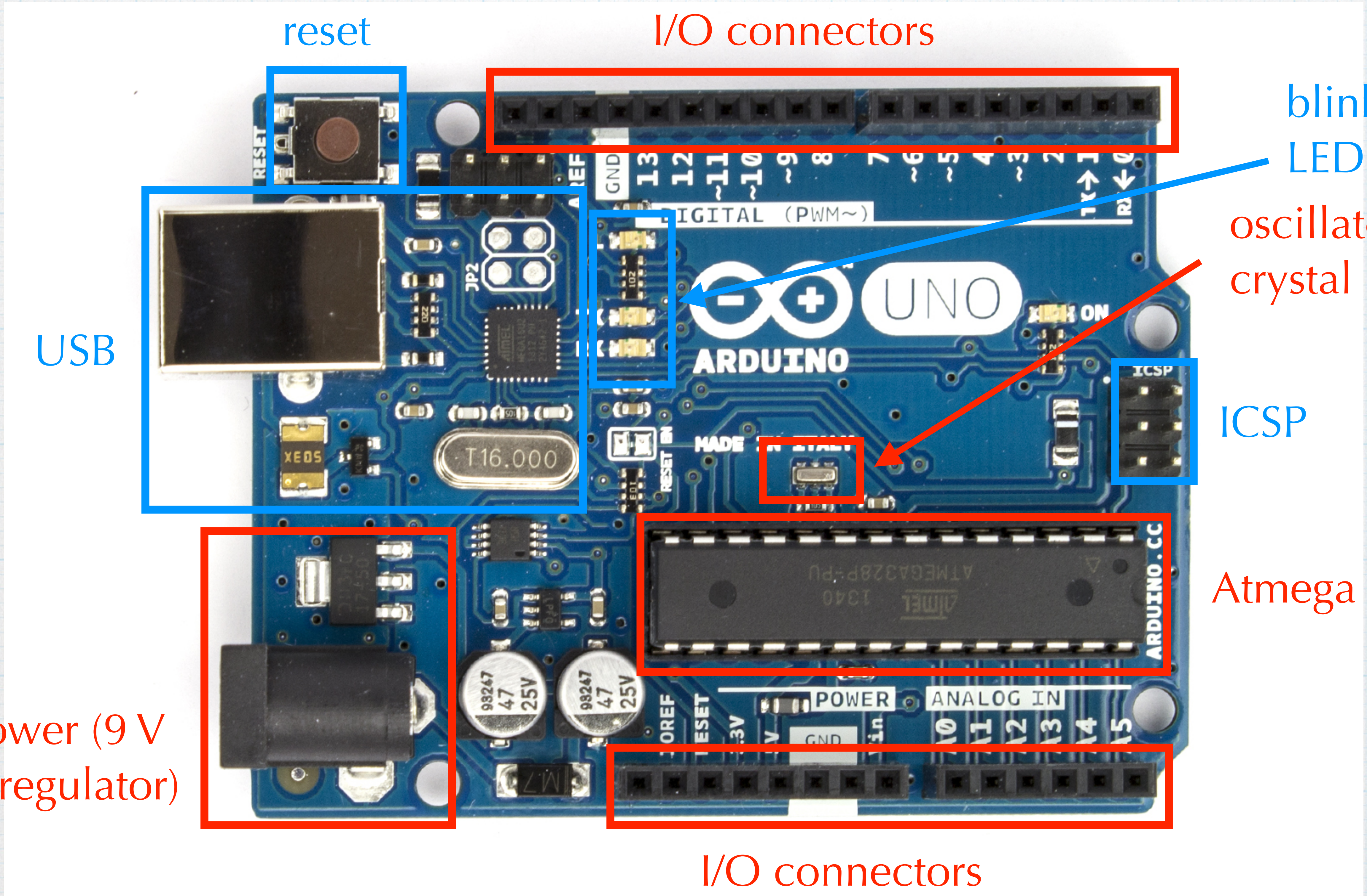
- We built the Altoids amp, which was fun, but of limited value because it cannot supply sufficient current to drive speakers bigger than headphones.
- So the next step up is a small amp that can supply the additional current needed to drive a small set of speakers.
- Take the basic non-inverting circuit from the Altoids and add a Class B output stage to deliver more current.
- To keep it simple, use a single power supply. This requires a change in the way the op amps interface with the audio input and speaker output.
- Still modest power — about 2 W per channel — but adequate to get good sound out of the speakers we will build.

GTDT amp

- Two channels.
- Voltage gain of 16.
- Maximum output power of about 2 W per channel.
- Flat frequency response from 15 Hz to > 20 kHz.
- Minimal distortion.
- Works from a single 12-VRMS wall-plug transformer. (Or an 18-V DC supply.)
- No power switch, no volume control. (Add these if you want.)
- Note that the basic kit does not include the box and connectors. You can make your own. Or IEEE has a standard box configuration that you can obtain.

Cyduino project

Basic Arduino layout



Boil the Arduino down.

Definitely keep:

- Atmega 328
- voltage regulator
- quartz crystal and capacitors for the clock
- pull-up resistor on the reset pin

Maybe keep:

- Momentary switch for reset
- ICSP header for programming
- LED power indicator
- Header pins, terminal blocks, or other types of wiring connectors

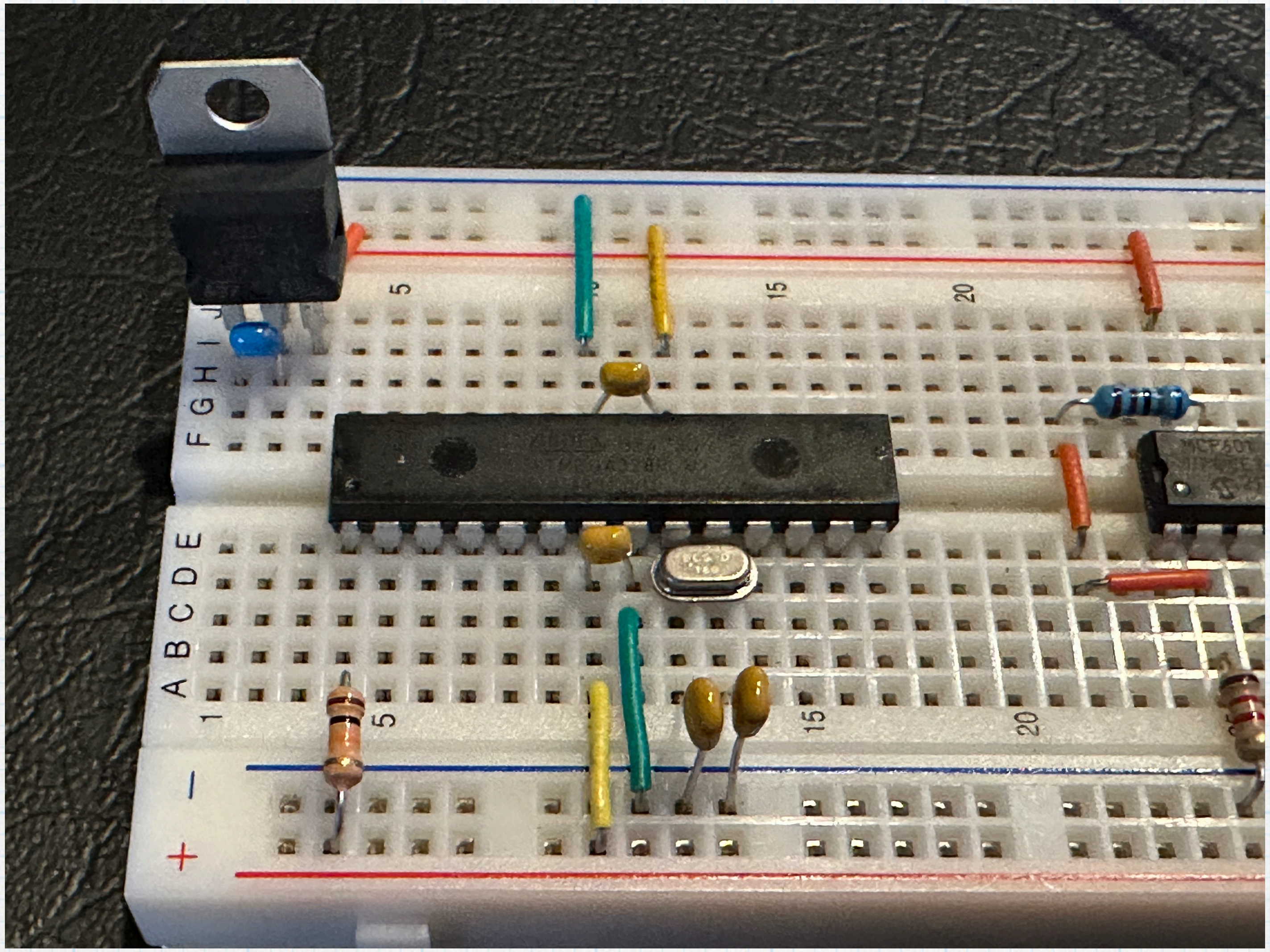
Lose:

- USB interface. Without it, we can't program the Atmega directly. Also, we must use an alternative power input.
- Blinky LEDs. Not a big deal.

Add:

- An OLED display
- Better wire connectors

Breadboard version



- It's easy to build a breadboard version for just a few bucks.

But it is nice to have a prototyping board that includes a display (which I almost always want) and better connectors for the wires. And if it is cheaper than an Arduino, so much the better.

- Will need a good way to program it, though.