Tu. 3/5: Ch 16 *Sound Reproduction* **Th. 3/7:** Ch 16 *Sound Reproduction*	HW7:Ch16:1^W,7^W,11^W… Ch16:13^W,18^W,19^W	**Mon. 3/3** or **Tues. 3/4:** Lab 8 *Percussion pt 1 - Drums*
Tu. 3/12: Ch 6 *The Human Ear* **Th. 3/14:** Ch 17 *The Ear Revisited &* http://dx.doi.org/10.1063/1.3603917 http://phys.org/news/2013-02-human-fourier-uncertainty-principle.html	HW8: Ch6: 2,4^W… Ch17: 1,3,4,16,20,23	**Mon. 3/11** or **Tues. 3/12:** Lab 9 *Electronic Amplification & Speakers*

Equipment:

- Function generator
- Speaker
- Speaker diaphragm without the speaker cabinet (and some alligator clips)
- Microphone on stand
- ruler
- O'scope & laptop
- Vacuum tube
- Transistor
- Amplifier (preferably surface mount)
- Disassembled and enclosed speaker
- Ipod and 2 RCA stereo amps to drive speaker
- Torsion wave demo with different length bars for impedance mismatch and match
- 2 magnets, fairly strong but not too strong.

Far too many cool things to talk about in this chapter, so I want you to take a moment and think about what topics you particularly want us to hit upon today.

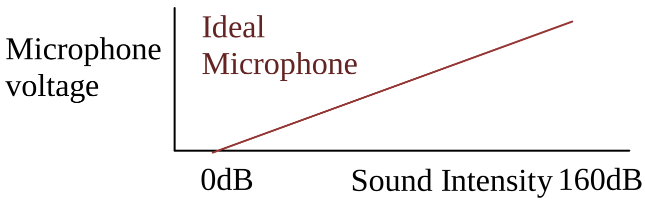
Chapter 16 Sound Reproduction
Recap from last time: Electronics is essentially the manipulation of charged particles using the attractions and repulsions of charges.

- **Electric Interaction.** By electric forces, similarly charged particles are repelled, so forcing charged particles onto a wire at one end dominoes to force ones off the other end – makes electrical current flow.
  - **Ohm's Law.** You can increase/decrease the flow one of two ways – changing how hard you're driving it (trading a 1.5V batter for a 9V battery) or changing how good a conductor it's flowing through.
- **Magnetic Interaction.** By magnetic forces, two parallel currents of charges, say flowing in two parallel wires, pull on each other drawing the current-carrying objects closer, while two anti-parallel currents push on each other, pushing the current-carrying objects apart. That's true whether we're talking two wires, a wire and a magnet, or two magnets (in which the "currents" are built of microscopic charge circulation.)
  - **Magnetic Transducers.** So the magnetic interaction underlies most common transducers. Attach a paper cone to a coil of wire and oscillate the current through the wire at 440Hz, then, if you've got a magnet nearby, the wire and cone will vibrate at 440Hz and thus drive sound

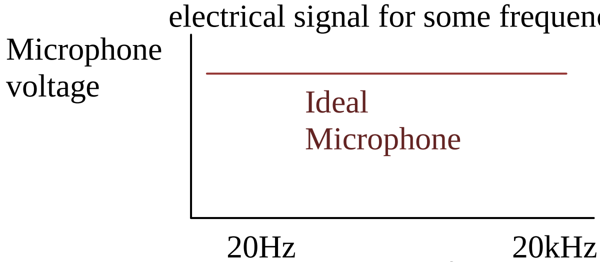
waves in the air with that frequency. Alternatively, sing at the cone, and it vibrated, dragging the wire coil with it, and if that's in the presence of a magnet, a current is driven through the wire, thus an electrical signal oscillates to mimic how the sound wave does.

16.3 Microphones

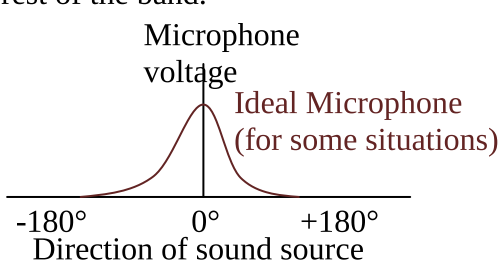
- **Introduction:** Now that we know how something can *work* as a speaker or microphone, we are ready to ask how to make it *work well.* Hearing a microphone played as a speaker and seeing a speaker played as a microphone, it's clear that speakers and microphones have fairly different demands & thus different measures of *working well.* We'll consider microphones first.
  - **Ideally –**
    - **Dynamic Range:** Sensitive to the quietest and loudest sound one could conceivable want to record: 0dB – 160dB, and responds "linearly", that is, if the sound pressure doubles, then the electrical signal's strength doubles.



    - **Frequency Response:** Sensitive to the lowest and highest frequency that we can hear (unless the microphone's for some other application, like recording ultrasonic mouse giggles), so we want a sensitivity to about 20 Hz to about 20kHz, and we want it to have a "flat" response, that is not generating a higher amplitude electrical signal for some frequencies than for others.



    - **Directional:** Quite often, you want to record sounds coming from one direction, but not from another, say from the singer, but not the rest of the band.



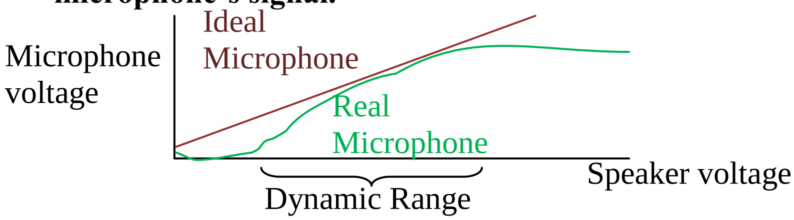
- **Trade-offs and Imperfections**
  - **Dynamic Range:**

Unavoidably, **Too quiet** doesn't record anything on the scope – the diaphragm just isn't getting pushed.

Too loud the signal on the oscilloscope clips. Ideally these limits are below the threshold of hearing and above the threshold of pain.

Demo: Dynamic Range of our microphone

Drive speaker with Function generator, Plot generator's signal and microphone's signal.



- **Trade offs**
  - If you build a very *dainty* microphone, you can get it sensitive to very *quit* sounds, down to the threshold of hearing. But then it's so dainty that a sound at the threshold of pain would demolish it.
  - If you build a microphone that can withstand the threshold of pain, it won't hear down to the threshold of hearing.
  - **Be kind to your Ears**
    - Incidentally, the same holds true for your ears. You have a huge dynamic range, but if you exceed it, you can do damage. Even the morning after a loud concert, your ears may not be fully recovered. For rock stars who play night after night of loud concerts, their hearing may *never* recover.

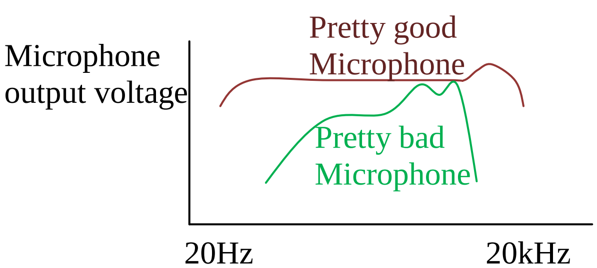
- **Frequency Response**
  - **Demo: Real Microphone response**
    - Pan through frequencies and see O'scope signal's amplitude vary.
  - **Trade offs**
    - **Smaller, higher resonance**. Practically, the microphone diaphragm has some stiffness & some inertia – a springyness. Like a spring, there are some frequencies that it responds well to and some it doesn't. This can be minimized. As you know from vibrations on a string, the smaller the string, the higher the resonance frequency. Thus a microphone head can be made small enough that the resonance frequency is above any frequency that you care about – above 20 kHz.
    - **Smaller, greater higher frequency range**. Also want to make small so that the sound's wave isn't varying across the face of the microphone –pushing one part in while drawing another part out, so smaller than the wavelength.

- **Homework-like example:**
  So different regions of the microphone head do not experience contradictory messages from the sound (pushing in here and out there), want Diameter << Wavelength.

  If you want to be sensitive to all audible sound, how small should the diaphragm's diameter be?

  First, what are the frequency limits of hearing?
  20Hz – 20kHz
  Second, which corresponds to the smallest wavelength?
  20kHz
  What is that wavelength?
  344m/s / 20kHz = 0.0172m=1.72cm
  So, the diameter must be much smaller than this.
- **Demo** – open microphone and measure diaphragm.

    - The book shows two sample plots in figure 16.5: the frequency-response curve for a very good microphone and for a typical microphone.



  - **bigger=more sensitive.** Then again, if you could get away with even just doubling the diameter, how would that effect your sensitivity? Doubling diameter means quadrupling surface area, which means quadrupling the variations in force for given variations in pressure since F = PS. Assuming that the microphone head responds something like a mass on a spring, quadrupling the Force means quadrupling the diaphragm's displacement, and so the electrical signal produced. Put another way, for the same minimum-threshold electrical signal, now pressure variations that are ¼ as strong will register. Recalling that Intensity goes like pressure amplitude squared, we're talking intensities that are 1/16th, or 10dBLog(1/16)=12dB. That's quite a gain!

  - **Directionality**
    - **Pressure vs. Pressure Gradient**
      - **Pressure.** If the backside of the microphone**'s** diaphragm is enclosed so that the pressure back there can't change quickly in response to the sound wave (as is the backside of our eardrum), then the microphone's response depends on

just how the pressure on the front varies relative to the constant backside's pressure. As some of you already know, and the rest of us will learn next week, that's how he eardrum works. This isn't *very* directionally discriminant (a little though.)

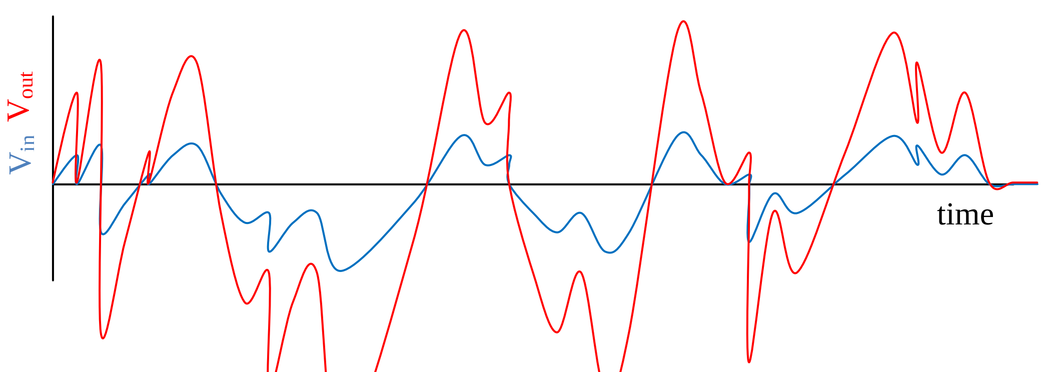
- **Pressure Gradient.** Alternatively, if the speaker diaphragm is open to air on *both* sides, then the sound wave has the ability to get to *both* sides. This allows for directionality.

- **Frequency/Direction.** Microphones that have only one side of the diaphragm exposed are hardly directionally sensitive at all, but they can easily be responsive to all frequencies (shy of resonance and wavelength limitations). Microphones with diaphragms that are exposed on the front and back are directional – only picking up sound that comes from front or back, but not from the sides. However, these are sensitive to pressure *differences* between the two sides, not just the pressure, so they don't strictly record the sound at one distance; in particular, the slight path difference between arriving at the front and back side of the microphone can be a large *fraction* of the total distance from the source when the microphone is near the source, thus the $1/r^2$ drop off in intensity is significant, especially for long-wavelength sounds for which there's otherwise little phase difference. (in general, the longer the wavelength, the smaller the phase difference, so the smaller the response the microphone should have; perhaps that is electronically compensated?)

16.4 Amplifiers

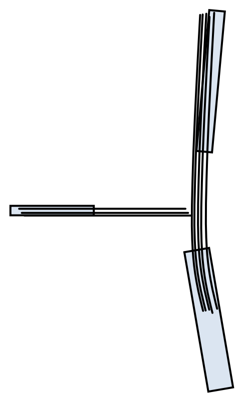
A transducer only gets you so far, it just translates between mechanical motion of air a motion of electrons that mimics it, thus oscillating current and voltage (the 'push' that drives electrons around). The simplest plausible application is electronic amplification – you intercept sound waves with a microphone, and so generate an electrical signal which you then make bigger somehow before using the new signal to drive a speaker and so generate stronger sound waves.

Ideal:
For how easy it is to imagine what an ideal amplifier should do, it's surprisingly difficult to make one. Ideally, regardless of the amplitude or frequency of the signal getting provided by the microphone, you want the amplifier to do nothing else but make it bigger.



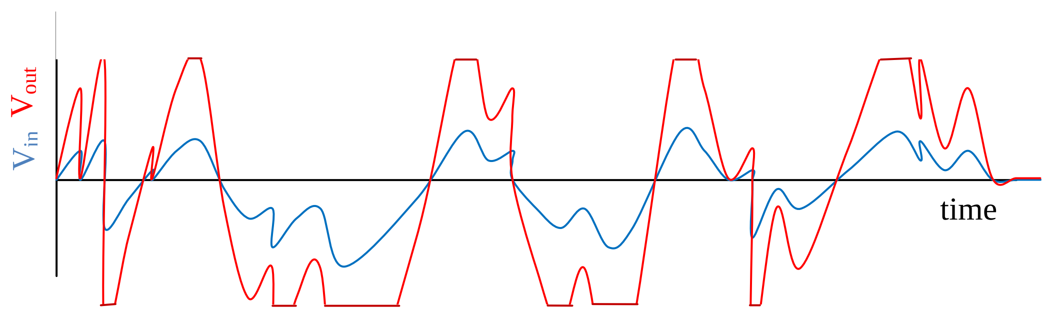
Water-jet amplifier
Remembering that water current flow is a pretty good analogy for electrical current flow, the book suggests a 'water amplifier' in which we can see how an electrical one might operate and see what some of its imperfections might be.

A strong stream of water flows out of an 'emitter' pipe and normally altogether misses a "collector" pipe, but there's a small "gate" hose that can be sprayed perpendicularly at the emitter pipe's stream to push it toward the collector pipe. This is a comparatively *small* control stream that's herding a comparatively large stream. So, as you strengthen and weaken the control stream, you get more or less flow in the collector. That's often how an amplifier works. It's not that your initial voltage magically grows, rather, it's used as the control on a much larger supply.



Imperfections.

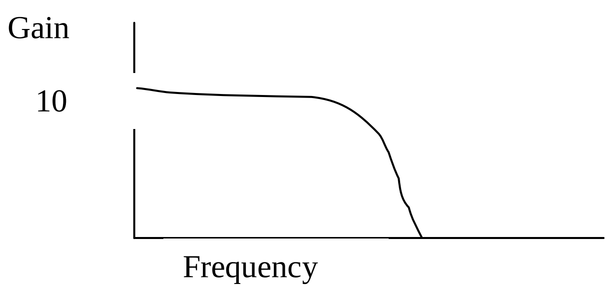
Clipping
One obvious imperfection is that the emitter's beam only has so much to give, once you've got it fully aligned, that's it, you've 'maxed out' or 'clipped' the signal:



If we're processing sound, clearly, if the amplified & clipped signal drove a speaker, the sound wouldn't just be *louder* than the original, it would be comprised of different high-frequency components (recall that the flat-topped square wave sounds like it's built of lots of higher and higher frequency pure tones)

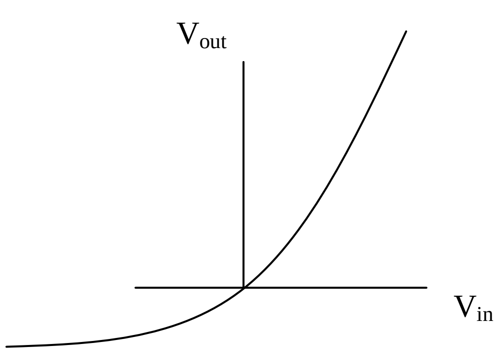
- **What clipping *sounds* like.** Remember, we've observed that simple sinusoidal waves sound "pure" to use, and anything else sounds complicated, and by "complicated" we mean they sound like multiple pure tones being played at once. When a microphone 'clips' while you're recording the signal, then playing it back sounds much more complex. This actually gets used as an electric guitar effect.
- **Demo:** ipod (playing acoustic guitar) played into one amp which is cranked and played into a second which is at a reasonable level and plugged into the speaker.

Response Time / High-frequency cut off
Another problem you can imagine is that it takes some time for the amplification process to play out – you increase the control jet's flow, and that pushes on the emitter's stream which then begins to move to the new position, it takes time. That means that very rapidly changing input signals just wouldn't have the time to fully amplify. The result is that higher frequency signals get amplified less.

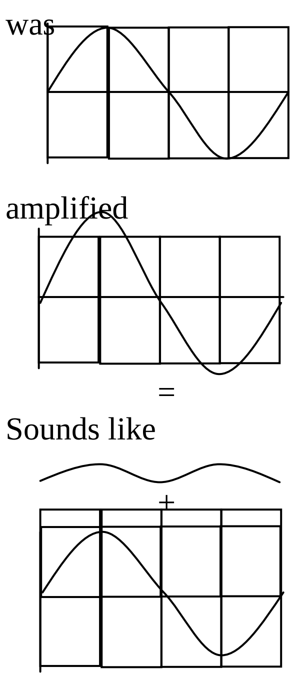


Say your original signal was a mix of a low frequency and a high frequency pure tone. The resulting one would have the low frequency appropriately, say 10 time amplified / louder, but the high frequency would only be maybe 5 times amplified. So the over all sound would be 'darker.'

Non-linear response.
Say the pipes we're dealing with here have your typical circular cross-section, then as first begin turning on the control stream you get relatively little increase in water flow since the pipe's rather narrow out near the edge, but the more you push it in, the stronger the gain because the pipe is fatter near the middle. Something similar can happen for electrical amplifiers, in fact, the simplest amplifiers have *exponential* responses - the first little increase in signal increases the output a little, then next increases the output a lot, and the next increases it a heck of a lot.



This has a funny effect – recall that only a pure sine wave sounds like a single, pure tone; however, any periodic wave that isn't a pure sine wave can be built of / heard as simultaneous pure tones in the same harmonic series. So, an amplifier that essentially gains different strength signals differently would squash/stretch a pure sine wave slightly; it's still got the same periodicity, so it must sound like a composite of sine waves.



So, it sounds rather like it amplified the wave appropriately (perhaps not to the exact intended degree) but then added in a harmonic, so it's a "harmonic distortion."

Phase distortion

Another kind of distortion that we wouldn't easily see in the water-flow example is "phase distortion". Usually the amplifier doesn't produce a wave that's completely in synch with the input signal, rather it's shifted a little, but it's not a uniform shift for the whole wave, rather, component by component, it's shifted a little differently, but that only happens significantly near resonance frequency.

Types of Amplifier

Vacuum tube

The first electrical amplifiers were vacuum tubes. As the name suggests, an evacuated tube, with two oppositely and highly charged plates and a mesh in between, as the voltage on the mesh is varied (with a very small current on and off), it can strongly change how hard it is for charges to flow between the two plates, thus the output current.

Transistor
The same amplifying effect, but achieved within a very small solid of a semiconductor that's been "doped". There are different styles; perhaps the easiest to understand is the MOSFET: Metal-Oxide-Semiconductor Field Effect Transistor. Think of a flexible hose with water flowing through it – the water wants to flow because you're pumping it, but you can squash the middle of the wire to constrict the flow. That's kind of how a MOSFET works – a high voltage is applied between one end and another, so the current *wants* to flow, but there's a relatively narrow bottle neck that you can make more or less conductive with small variations in a third voltage.

Operational Amplifier
Because a single Vacuum tube or a single Transistor is highly non-linear, they get wired up in complicated ways that allow for compensations so the end product is a circuit that is much more linear. Transistors can be made very very small, so they can be built into one "integrated circuit" chip. The Operational Amplifier is one such. A typical "Op-Amp" has tens of transistors all wired up inside.

Ultimately, *some* non linearity will always exist, and "clipping" can happen if you're not careful about the gain, but very smart design can diminish this greatly, whether you're using vacuum tubes or semi-conductor based technology (or a combo).

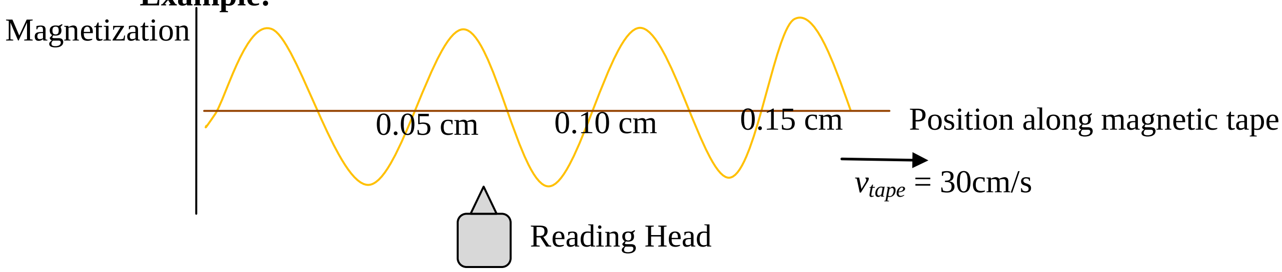
16.5 Recording

Record
Magnetic Cassette

Writing: I've got these two magnets here, they like to line up with each other so that if I bring one near the other it can flip the other over to get them to line up. Of course, a current passing through a coil of wire is equivalent to a magnet. So, if you've got a microphone plugged into an amplifier and *that's* plugged into a coil of wire, as the signal from the microphone oscillates, then so does the current in the coil of wire, *and* so does the force that it would exert on any nearby magnets, making them want to flip one way or another.
Now a cassette has magnetic material on a tape, as the tape moves by the coil, the microscopic magnets pass by the coil and one little magnet gets flipped one way, then next gets flipped another way, the next the other way,… In that way, the oscillations of the coil's current / the microphone's head / the air pressure against the microphone are translated into oscillations in magnets' orientations along the tape.

Playing: Now, this process can be run backwards too – draw the tap past a coil of wire, as the little magnets on the tape pass the coil, they drive a current in it back and forth. If that's plugged into an amplifier, and that's plugged into a speaker, then you get the speaker's diaphragm getting driven back and forth – producing sound.

Example:

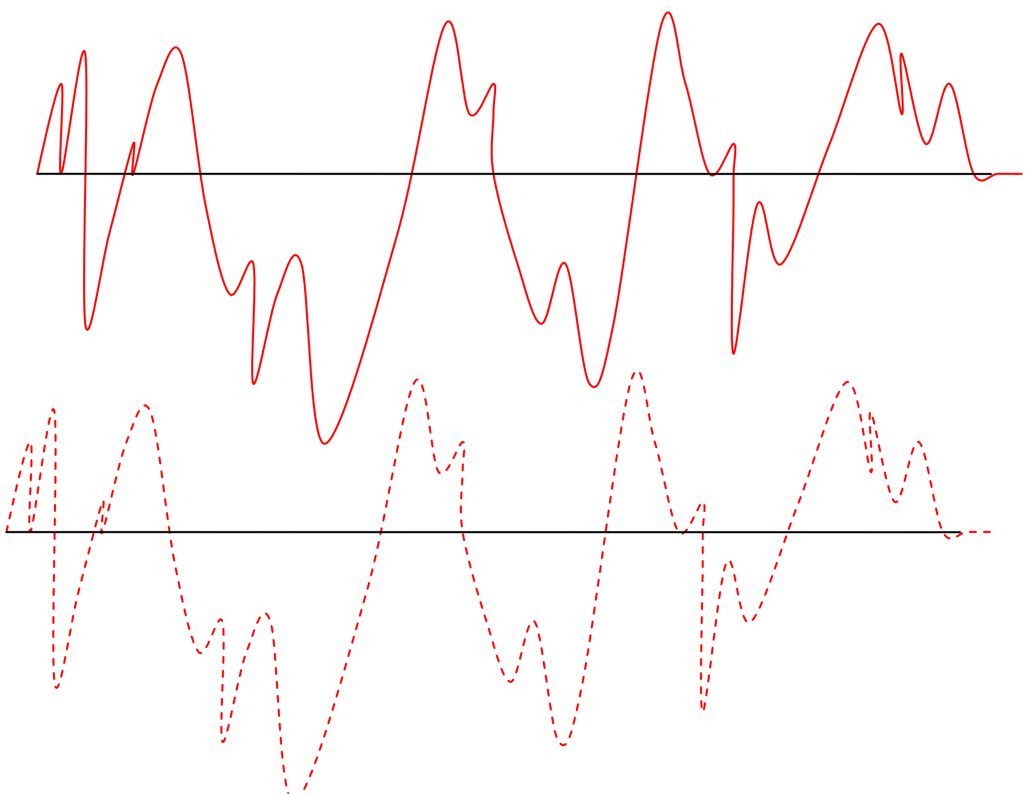


Say the tape moves by the tapedeck's playing head at 30cm/s. Then with what frequency would the magnetization peaks pass the head?

***Digital* disc**

Sampling, "words"& "bits"

The first step in recording something digitally is 'sampling it'. Like a movie doesn't actually record a continuous stream of visuals, it snaps frame after frame after frame, and thus "samples" the images, with a sound wave, you'd take "snapshots" of the wave form.



Now, the highest frequency sound that you can hear is around 20 kHz, so to capture undulations with that frequency, you'd need to sample twice as quickly – so you could see a peak, then a trough, then a peak, then a trough,… 40 kHz, or once every 0.0025ms.

The next step is to write down a huge data table of the waveform's values for each sample taken, like

Time (ms)	Voltage (mV) (in Decimal)	Voltage (mV) in binary
0	00042	0000000010101010
0.0025	00043	0000000010101011
0.0050	00054	0000000000110110
…	…	…
0.0225	10010	0001001110001101 0
0.0250	10012	0001001110001110 0
0.0275	10011	0001001110001101 1
…	…	…

Word: each voltage value that's recorded is a "word".
Here, I've allotted 5 numerals for digits to represent each voltage and I'm writing it in decimal values. Now, if it's a given that each number is represented with five consecutive numerals, and that each number is a voltage reading another 0.0025ms along, then I could write the whole table's worth of voltage values as one enormous string of numerals:

000430004500054…1001010012100011…

Bits: Actually, the CD player uses 16 numerals and writes it in *binary*, but it's about equivalent to using 5 numerals in decimal. Like the letters of a word, each numeral is a little bit of information that we use to spell out the voltage's value. One numeral in binary is called a "bit."

In this example, our string of numerals would look like

00000000010101010000000000010101011000000000000110110…0001001110001101000
0100111000111000001001110001101 1…

Decimal vs. Binary: The Decimal system has ten different numerals, so we can count from zero to nine before we have to move on from the "ones place" to the "tens place", the "hundreds place" etc. In Binary, there are just two numerals, so you can only count from zero to one before moving from the "ones place" to the "two's place", the "fours place." But the really important thing is you need a vocabulary of just two symbols to be able to write in binary.

CD – chiseled in metal foil. If you look at figure 16.17, each track you see there is like one line on the page of my rambling number. The 0's and 1's are represented by the high and low spots.

00000000010101010000000000010101011000000000000110110…0001001110001101000
0100111000111000001001110001101 1…

One can actually dream up more complicated ways of encoding things, reserving the 16th bit to represent + or – for example, but you get the idea.

As the book notes, for writing a CD, the sample rate is about 10% higher, so 44,100 values are sampled every second. And that's just for mono; if you want *stereo*, you need to keep a record of both the left channel and the right channel, so twice as many total samples per second: 88,200 samples/sec. Since it takes 16 bits to write each sample, that means that (16bits/sample)×(88,200samples/sec)=1,411,200 bits/sec are recorded or read back when a CD is played.

16.6 Loudspeakers
All the electronics stuff about the chapter was brand-new; the last section though, on speakers, largely draws on stuff we've met in previous chapters, so it's probably not as useful for me to go into great depth here, though I'm perfectly happy to answer any questions you may have about them.

Size and diffraction

When I'd introduced the idea of diffraction – the way a sound wave spreads as it travels from a hole, my prime example was a speaker diaphragm. That example's really at home here in this chapter. The rule of thumb that you might recall is that if $\lambda > D$, then the sound spreads out a lot when it leaves the diaphragm; conversely, if $\lambda < D$, then the sound fairly "beams" straight forward. That's why woofers *can* be quite large while tweeters *need to be* quite small. You'll use this threshold relation on your homework.

Enclosure
The book points out the importance of having an enclosure; this so long-wavelength sounds waves emitted from the front don't get canceled by those emitted from the back. That'll happen because when the front of the speaker *compresses* the air, the back *rarefies* the air, and vice versa. So the sound emitted by the two sides is out of phase.

Demo: play music with and without speaker cabinet.

So the cabinet's primary role *isn't* to somehow *amplify* the low notes, it's just to *prevent* them from getting *cancelled* by destructive interference. That said, the book does mention some tricks for using both the front and the back signal, and thus doubling your bang for your bucks.

Horn – impedance matching
As you probably recall, when a wave pulse hits a fixed end, it reflects back. But you may not recall that when a wave pulse hits a free end, it also reflects back. In fact, that was why we could get resonance in our glass tubes that were open at the top so many weeks ago in lab. You may not have thought of it this way, but that reflection also happens with speakers.

Torsion wave demo

Generally, want the transition space, here, the horn, around a wavelength in length.