9/10 · period (sec): $\frac{1}{f}$

· freq (Hz) : $\frac{1}{\text{period}}$

FUND FREQ: freq. of which harmonics are all int. multiples

ex: 880, 660 → 220

 · sawtooth wave: has every harmonic : $1, \frac{1}{2}, \frac{1}{3}, \ldots, \frac{1}{n}$

 · square wave: odd n's where $1, \frac{1}{3}, \frac{1}{5}, \ldots, \frac{1}{n}$ (even amplitudes = 0) $[1, 0, \frac{1}{3}, 0]$

 · triangle wave: odd n's where $1, \frac{1}{3^2}, \frac{1}{5^2}, \ldots, \frac{1}{n^2}$

· phantom fundamental : take low freq & add harmonics to make it seem like the freq. (fund. doesn't have to be present for us to perceive the sound)

2nd harm

Ex: $[1, 0, \frac{1}{3}, 0, \frac{1}{4}]$ . . .

1st harm

→ in practice we don't generate a full freq b/c human hearing is 20-20k Hz

· octave: double the most recent freq 220, 440, 880, etc.

continuous energy
· white noise: energy distributed evenly across freq. spectrum 

· pink noise: less high-frequency energy, more low-freq

· impulse: instantaneously short sound distributed across freq. spectrum (plucking, striking)

· piano, guitar, string instruments doesn't have much surface area, so doesn't distribute much air.

- energy of string is transferred to the soundboard.

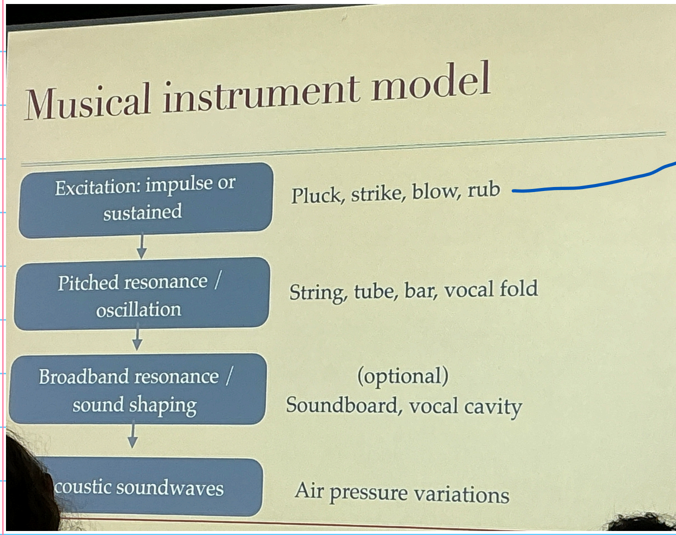
· resonance: vibrate more freely (harmonics)

· freq. spectrum: what frequencies are present in a sound

· freq. response: which sounds will resonate (harmonics) vs. damped out

· if you keep putting energy in @ resonant freq. faster than it can damp, can break things

MUSICAL INSTRUMENT MODEL:

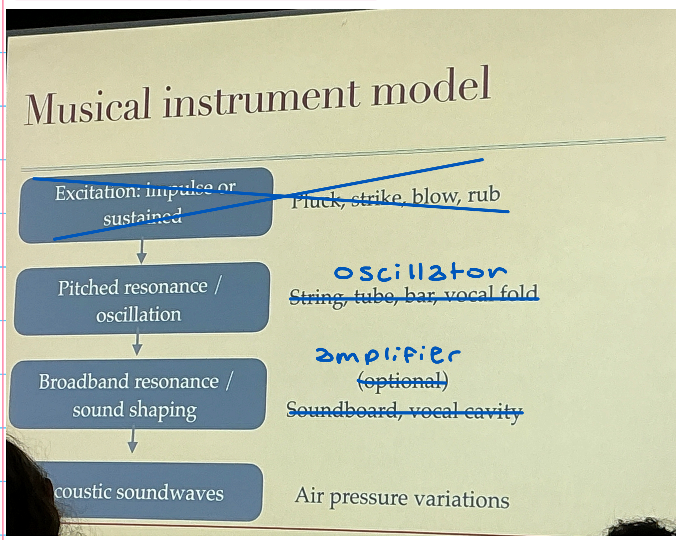


quality of impulse can affect output of sound

9/15: ## ADDITIVE SYNTHESIS

- creating sound by combining sine waves w/ different amplitudes/ freq / phases
- complicated, hard to do irl (sine waves hard to generate)

SUBTRACTIVE SYNTHESIS:



oscillator — modifies

pitch → VCO → VCF → VCA

VCO: voltage controlled oscillator

VCF: pitch / tone shape — voltage controlled filter

VCA: amplitude of out sig. — voltage controlled amplifier

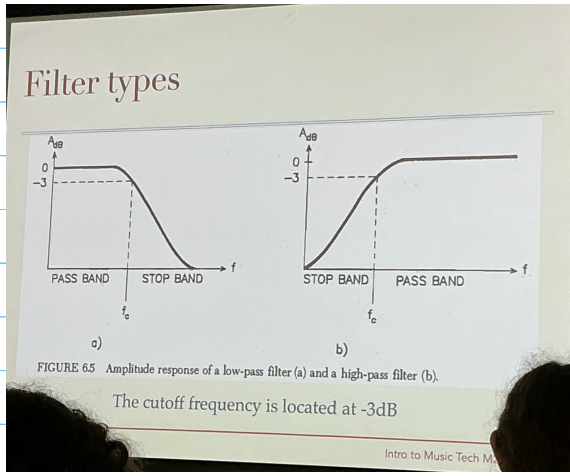
VCO [
Sawtooth: nazzle-y
Square wave: deeper,

VCF: getting rid of higher frequency harmonics
- low pass filter (allows low freq. to pass & attenuates higher freqs)
- attenuate frequencies GRADUALLY

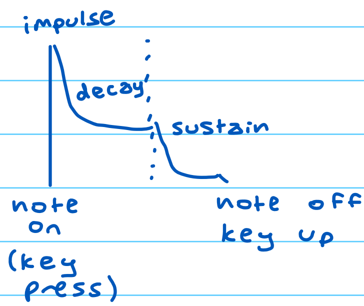
RESONANCE: makes frequency ring out
- high resonance makes higher freqs louder

FILTER TYPES: messes w/ attenuation & how fast sound dims

· cutoff freq = where attenuation begins



· analog to digital depends on filters



ADSR: attack, decay, sustain, release

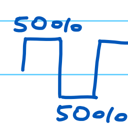
· sometimes release not instant

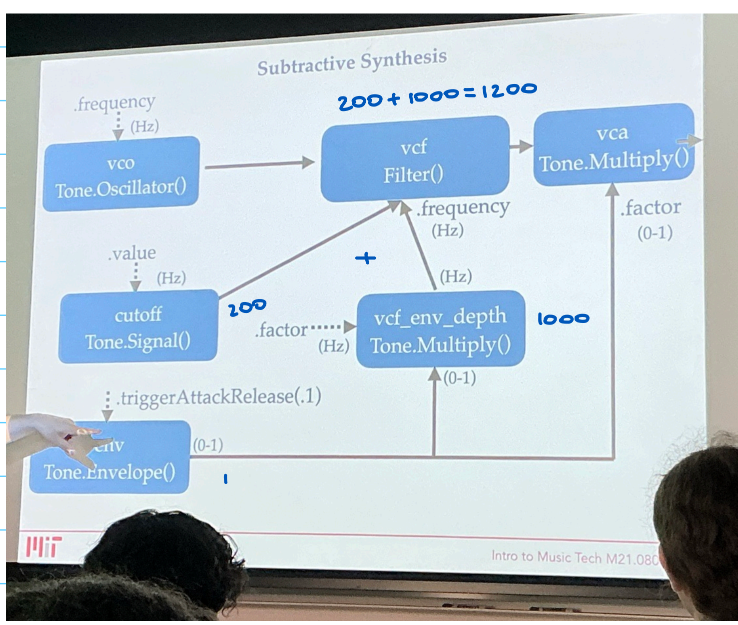
- damping, sustain

DUTY CYCLE:

· pulse-width modulation:

- can speed up/slow

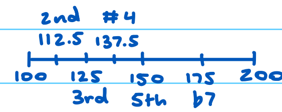




9/17

harmonic series

		100	R
2:1	octave	200	8 (oct)
3:2	fifth	300	5th
		400	8
4:3	fourth	500	3
		600	5
5:4	third	700	7
		800	8
6:5	minor third	900	2
		1000	3
7:6	seventh	1100	#4
8:7			
9:8	second		



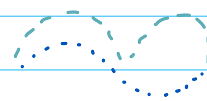
as you go up, intervals keep shrinking,

octaves get larger

FOURIER TRANSFORM:

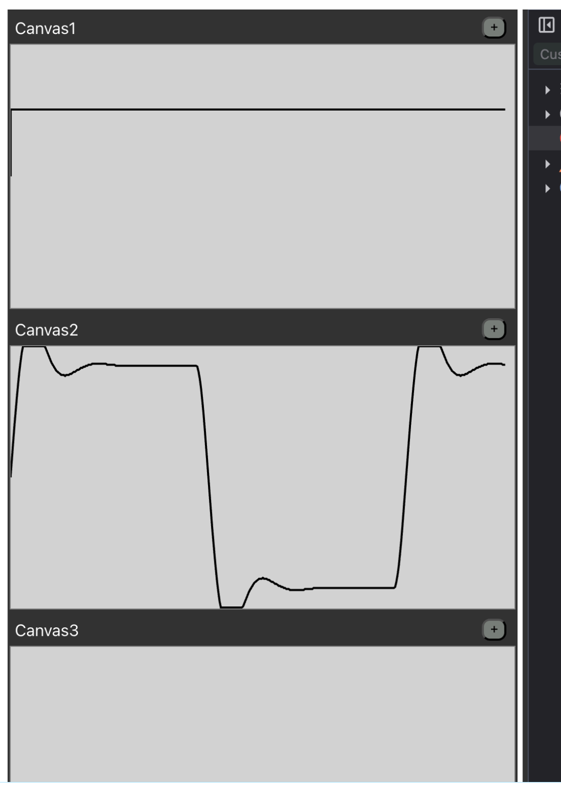
· takes signal & breaks into frequency components within signal

RMS (root mean square) = $\sqrt{\sum_{0}^{n} S_n^2}$



→ avg. amplitude of signal

```
/*
  Alt(option)-Enter: Evaluate Line
  Alt(option)-Shift-Enter: Evaluate Block
*/
let vco = new Tone.Oscillator().start() //
let output = new Tone.Multiply(.1).toDestinatio
let vcf = new Tone.Filter()
let vca = new Tone.Multiply()
let env = new Tone.Envelope()

vco.connect(vcf)
vcf.connect(vca)
vca.connect(output)
env.connect(vca.factor)

env.triggerAttackRelease(3)

env.attack = 1 // need this val for no pop
env.decay = 1
env.sustain = 0.1 // amplitude 0-1
env.release = 0.1

vco.frequency.value = 150
vco.type = 'square'
vcf.frequency.value = 1000

let scope1 = new Oscilloscope('Canvas1')
env.connect(scope1.input)
let scope2 = new Oscilloscope('Canvas2')
vcf.connect(scope2.input)
```

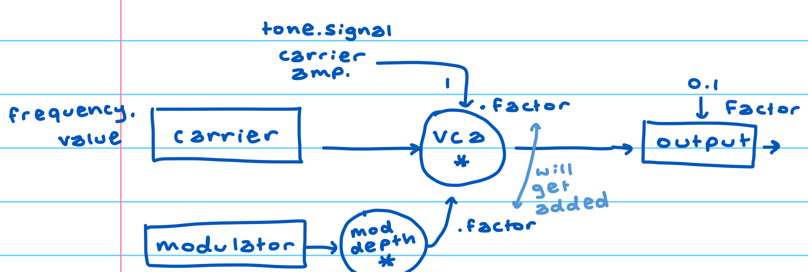


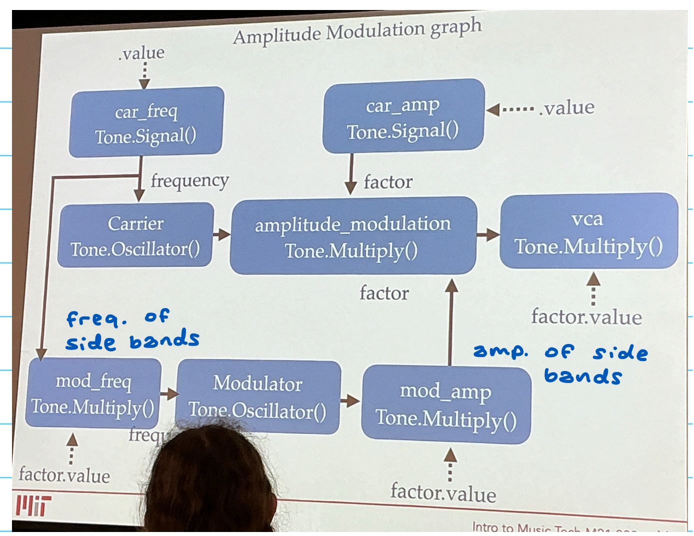
9/24 **MOOG MICROMOOG**

- produced by Moog music in 1970s ; cheaper than Mini Moog
- monophonic subtractive synthesizer (only 1 vco)
- subtractive synth: starts w/ rich waveforms from vco & uses filters / modulators
- additional sources: noise generator (white noise), external audio input
- octave, frequency, doubling, fine-tune frequency controls
  - octave → 32' to 2'
  - frequency → transpose
  - doubling → up / down octaves
  - fine-tune frequency → controls all octaves simultaneously
- cutoff control - sets cutoff frequency where attenuation begins
- modulation: changes params. of sound over time.
  - modulation wheel : controls degree of modulation
- self-oscillation: can make a sine wave → "filter mode" switch to tone
- timbre: 24dB steep filter, higher cutoff = brighter
  - sub-octave generator: adding octaves

ANALOG:

- hard to have another oscillator & keep it in tune
- sub-oscillator: play sound 1-2 octaves below main oscillator





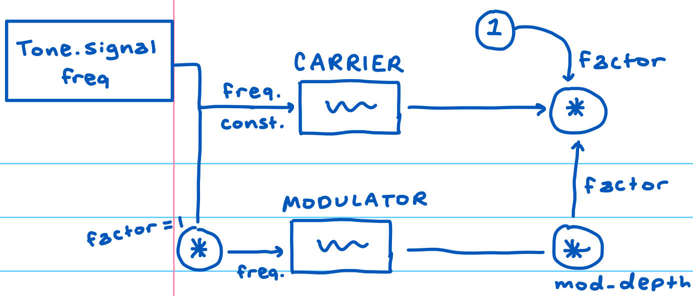
		freq:
Ex:	carrier	440
	modulator	220
	output:	660

going negative: takes waveform & flips : neg =

modulation depth: controls sidebans

AM & RING modulation: (amplitude mod.)

- amount of mod. relates to amplitude of modulator
- ring modulation: carrier dissapears, only sidebands stay

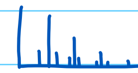


Ex: freq of carrier=100,
mult fact=4

modulator: sideburns 500, -300

100	100
500	300
-300	500

• what if carrier has many frequencies (sawtooth)?



→ every frequency gets it's own set of sidebands.

MIDI: musical instrument digital interface; 1980 tech.

• sharing info b/t hardware music devices

- adopted for lighting/etc.

• keyboard press: frequency, amplitude

• middle C: midi note 60, 261.63 Hz

- up an octave: +12 → 72

MIDI message format:

• each msg - multiple bits

• status bytes: always start w/ 1

1sss nnnn	0xxx xxxx	0yyy yyyy
status byte	pitch data 1	velocity/amplitude data 2

• w/o top bit, goes from 256 → 128 values; ok b/c piano has < 128 notes, & this value is pretty good for speed.

- criticisms that midi is only 128 resolution

• note on message (note, format) → depends when you press/release

• speed of pressing keys determines sound (velocity)

- can specify note off by sending note on w/ velocity = 0

• can set synthesizers to different channels.

• wire: DIN-5 standard 

• in, out, through msg (goes thru. to output)

CC (continuous control) messages:

• status byte will be diff, Data 1 = value, Data 2:

• pitch bend / modulation - types of CC

- can change pitch w/ -/+ octave — corse on 128
- 14 bits of res. - both data bits are for pitch blend

General MIDI

- 1st way of writing song/sending music around.
- sending MIDI files than music lib.
- has many sound presets - 128 presets
- drums usually on MIDI channel 10.

Problems w/ MIDI:

- low res (7 bit)
- not human readable
  - MIDI data is hex, not binary
- note on/off doesn't easily translate to continuous pitch/microtonal music
- limited support for lots of continuous data
- generally not that extensible

MIDI Updates

MPE:

- every note has its own pitch blend
- 3 dedicated channels for 14-bit cc data
- continuous frequency sweep, not step bend
- move finger side to side for vibrato, up/down for pitch bends.

MIDI 2.0:

- 32 bit

OSC (open sound control):

- no data restriction
- sending packets - transparent, human readable
- Ex: /synth/1/frequency 220.

Roland - Juno synth

presets of them you can quickplay

switch bit with pedal & edit live

- released in 1982
- unique features: arpeggio, chorus, memory (patches).

could go out of tune depending on environment

- uses DCO, not VCO. VCO charges/discharges capacitors. DCO uses digital clock. more accurate
- can mix multiple waveforms @ once, playing up to 6 notes @ same time
- change timbre: HPF/VCF, bender, LFO trigger,
- classic 80's sound
- easy to play

10/1: LINNDRUM SYNTH

· need for drum machines

· 1st programmable drum machine to use digital samples of real drums

· focus on rhythm backbone

- many buttons (pre-recorded sounds)

- many diff. outputs

· not rly a synthesizer... plays digital samples stored on EPROMS

- can swap out EPROMS to change sounds

- variable sample rate

· every drum: can adjust balance / volume

· panning sound for LinnDrum → L/R

· used VCO to change speed of recording / tuning

· hi-hat decay → open/closed clip sound (long vs. short decay)

· multiplexing: have different sounds "take turns" using same DAC.

· Curtis Electromusic

· can use casettees to save

· used by: Prince, Madonna, Paul McCartney, Michael Jackson, Queen

10/6 FM SYNTHESIS

• ADDITIVE SYNTHESIS: adding many sine waves together

• AM SYNTHESIS: oscillators are called carriers & modulators

· modulator affects carriers

• FM: affects frequency

FM:

• carrier = main central freq.

• modulator comes in & modulates around that

• Ex: freq val = 500, mod freq. = 100, amp mod = 400 & 600, 300 & 700, 200 & 900, etc.

• freq. range > mod range → must shift mod amp

Fund. freq: 500
can make them harmonics

• amp mod: 1 pair of sidebands

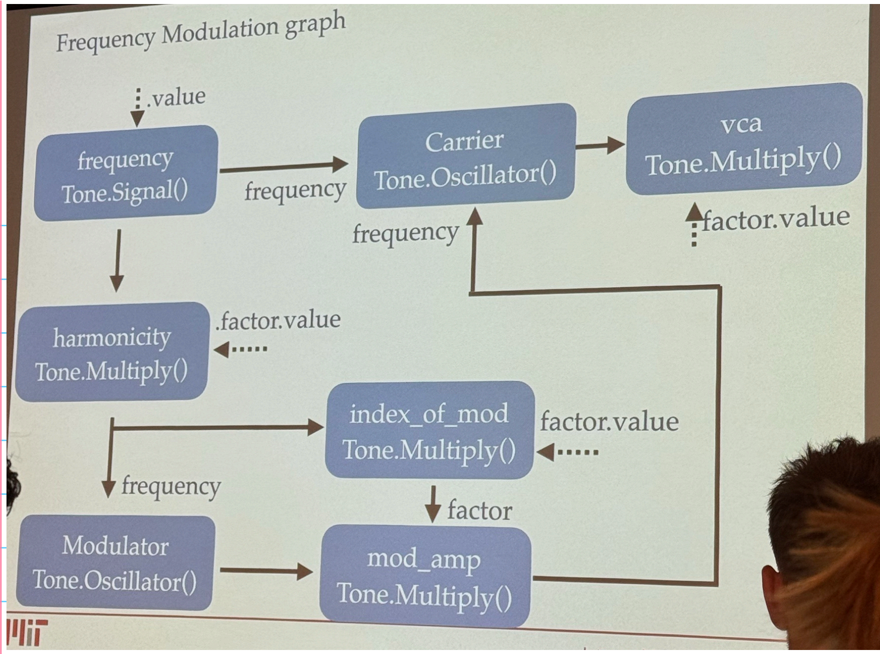
• freq mod: many sidebands

make freq mod = 500 → everything is harmonics

- relationship b/t freq mod & mod-amp controlling #sidebands

Ex: freq. val = 100, mod. freq. val = 1000, mod. amp factor val = 1500

• FM synthesis tends to sound metallic, bright, etc.



- harmonicity & index-of-mod multiplies carrier freq,
- mod-amp is also factor
- harmonicity determines freq. of modulator
  - sidebands = carrier-freq ± mod-amp

400 ± 800

-400 → 1200 Hz

Ex: freq = 400 → carrier

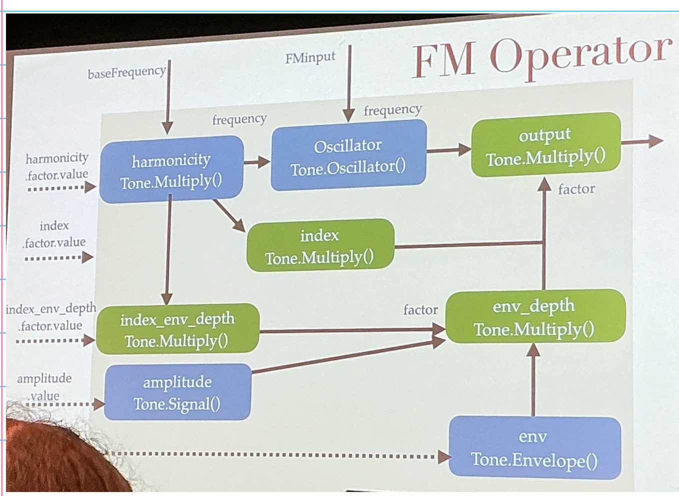
harm. fact = 2 → index-of-mod = 1

modul = 800 → mod-amp = 800

sidebands: 400 ± 800 ;

- high freq: fundamental freq. is almost lowest amp.
  - timbre = bright, have ghost freq.

- can change wave types, but can get insane amt of sidebands b/c multiples will be generated for each freq.
- $I = \frac{D}{F_m}$ D = amplitude of modulator
  Fm = freq. of modulator



YAMAHA DX7

- ppl still release synths based on it

Sampling:

- sample and hold in ADCs
- single note recordings
- music performance
- dj - esque (parts of songs)

Audio Buffers:

- store audio recordings in computer memory
- play back @ original speed / freq.

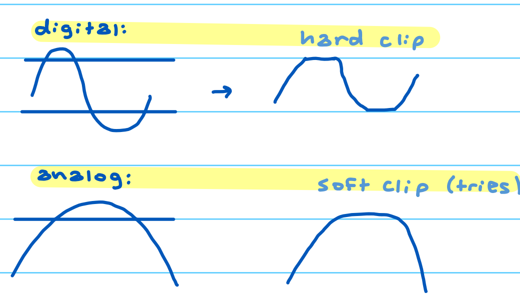
Samplers:

- key mapping : mapping playback speed ratio
- multisampling: recording many different samples of same instrum.
- timbre of instruments change @ diff. amp. levels
  - velocity switching: record each note @ diff. dynamics
- keyswitching: using selected keys to switch articulations
- professional sample libraries are expensive
  - free: philharmonia. co. uk

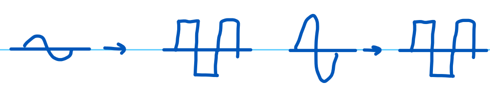
10/15 **Digital Audio Amplitude:**

- env: 0 → 1 
- voltage: -1 → 1 
- same (waveform) harmonics, diff. amplitude
- if amplifying beyond limit, will chop off

  AKA distortion



- Ex: distortion amplified to max:



even when you play soft, will clip.

Distortion: arbitrary alteration of a waveshape

- if sound has pitch, fundamental + harmonics is there
- harmonic distortion: adds harmonics (change shape of waveform)
- adds higher harmonics within limit

WAVESHAPING:

- as soon as we hard clip, many harmonics are introduced
- soft clipping: tanh func
- can use many shapes (sine, triangle). sine a bit softer
- wave folder:  triangle,  sine

PHYSICAL MODELLING:

- model behavior of physical elts (plucking, etc)

KARPLUS STRONG ALG:

- simulates pluck string 
- delay line to model movement

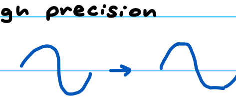
10/20 **SOUND: DIGITAL PERSPECTIVE**

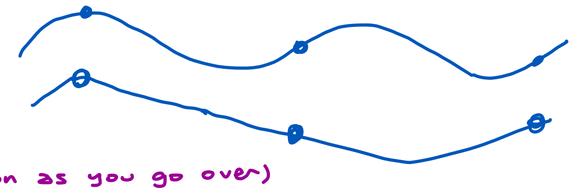
- decibals: how we measure amplitude (dB) - logarithmic
  - 6dB: ~doubling of amplitude (measureable) → for each 6dB there is doubling of amplitude
  - Ex: 12 dB → quadrupled, 18 dB: 2^3, 24 dB: $2^4 = 16x$
- dB SPL (sound pressure level): compared to threshold of perception of 1 kHz soundwave
  - 0 dB SPL = quietest sound ppl heard in lab
  - our classroom ~35 dB (soft whisper)
  - will always be + level
- dBu (professional standard)
- dBV (consumer standards)
- dB FS: negative. is below max noise

CONTINUOUS VS. DISCRETE SIGNALS:

- digital systems manipulate discrete, quantized signals
- time-sampling
- **SAMPLING THM:** continous time signal $x(t)$ with frequencies no higher than fmax can be reconstructed exactly from its samples $x[n] = x(nT_s)$ if sampling rate is @ least 2x highest freq. in signal.
  - have to band limit signal (don't exceed fmax) → $f_{max} \leq \frac{f_s}{2}$
  - not > Nyquist freq.

SAMPLING RATE: rate audio samples must be converted by ADC → DAC w/ high precision

- jitter: variations in timing of ADC or DAC (distorts waveform) 
  - ↳ higher samp. r. = worse jitter
- everything nowadays has ADC & DAC in it
- if you can't process @ sample rate fast enough, dropouts will occur (pops/clicks)



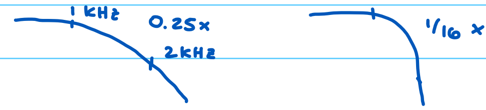
(as soon as you go over)

ALIASING (foldover): what happens when going over Nyquist freq.

· intro. of frequencies not present within source audio

· sampling rate of 40kHz, Nyquist of 20kHz → 21kHz generates aliased freq. of 19kHz

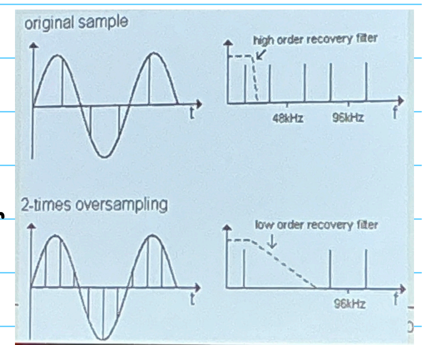
- 25kHz → 15kHz   35 kHz → 5kHz



FILTERING: $\frac{\text{dB}}{\text{octave}}$

· steeper slope = steeper droupout – fast attenuation

· analog filtering easier → do it before converting to digital



taken care of for us in the hardware stage

OVERSAMPLING:

· steeper digital filters after conversion to bring sampling rate down

· most ADCs / DACs use this

AMPLITUDE QUANTIZATION:

· converts continuous amplitude of sample into digital sample

· bit resolution = # bits used to store amplitude

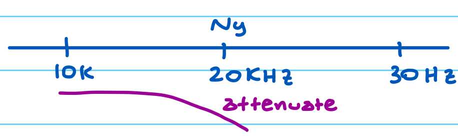
· higher bit depth = better quality

· 16 bits → 65,536 possible representations

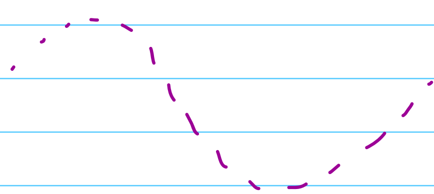
· max error = $\frac{1}{2^{\text{bit depth}}}$

· low bit depths / amp. signals becomes square wave (a bit = on/off)

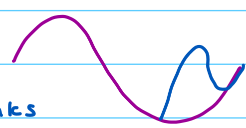
· less word size (bits) → makes sounds more distorted



24 input, Ny = 20, out = 16



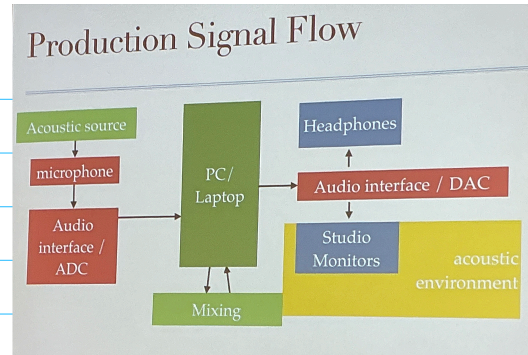
$\boxed{\text{DAC}}$ →



• send chunks

• have a big buffer

10/22 MUSIC PRODUCTION



- many goals directed by personal choices
  - simulate live music, no correct way
  - gear can be important
- maintain fidelity
- develop our ear!
- MONITORING: listening to results of actions/decisions
  - accurate monitoring is important!

CRITICAL LISTENING:

- equidistant from speakers — aimed directly towards ears

10/27 MICROPHONES: dynamic

transducer
- diaphram (body) → vibrations (air pressure variations) cause signal
  - thing that moves; super sensitive
- coil of wire connected to mic induces current (E-field)
- pretty robust, so a little less sensitive

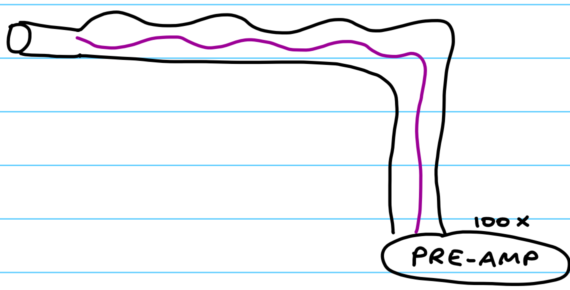
*performers generally louder during performance vs. soundcheck.

CONDENSOR mic:

- condensor is british for capacitor
- capacitance changes based on distance (diaphram)
- nothing connected to diaphram
- preferred; crisper

RODEO WRAP ("over-under")

- lies down flat
- better for wires (no tension)



Highpass Filter:

- cutoff freq of 100 Hz
- attenuation of low freqs.

Aux Send:

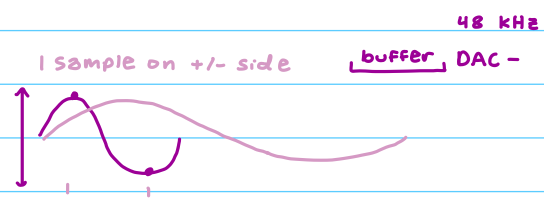
- has reverb setting
- effects on your voice
- reverb is historically expensive - each track can have reverb
- post fader send & pre-fader send
- common use: concert. band doesn't need drums, but audience does

Recording:

1) musicianship
2) good sounding instrument
3) good sounding room
4) mic positioning
5) mic choice
6) pre-amp gain

10/29 **In Reaper:**

- 48 kHz sampling rate
- Nyquist freq of 24 kHz
- can't have a freq. > 24 kHz
- samples capture amplitudes
- 24 bit resolution
- WAV = uncompressed audio file



Microphone:

- GAIN: controls sensitivity (amplifies mic signal)
- TRANSDUCER: device changes energy from one form into another (mic)
  - diaphragm + transducer = capsule
  - mic: air pressure variations → electric signals
  - hydrophone: water pressure variations → electric signals

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (null angle)	—	180°	126°	110°	90°
REAR REJECTION (relative to front)	0	25 dB	12 dB	6 dB	0
AMBIENT SOUND SENSITIVITY (relative to omni)	100%	33%	27%	25%	33%
DISTANCE FACTOR (relative to omni)	1	1.7	1.9	2	1.7

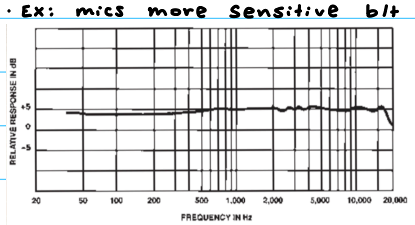
MIC POLARITY PATTERNS:

- **UNIDIRECTIONAL:** mostly sounds in front of mic (good for avoiding feedback, musician)
- **OMNIDIRECTIONAL:** pick up sound equally in all directions
- **BI-DIRECTIONAL:** sounds in front & back of mic
- **CARDIOID:** sounds in front of mic (unidirectional)
  - **PROXIMITY EFFECT:** sound sources move close to cardioid & produce artificial low-end boost

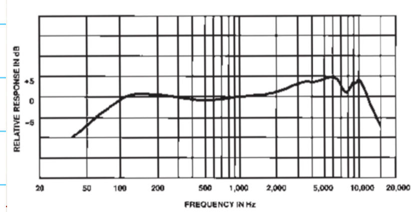
    happens w/ unidirectional mics.

FREQ. RESPONSE: mics will be more/less sensitive @ various frequencies

- Ex: mics more sensitive b/t 2-8 kHz (consonants, high freqs) → called **presence**



Flat frequency response



Shaped frequency response

] almost doubling (+6 dB = double)

DYNAMIC MIC: passive

- ex: shure SM58
- voice coil w/ conductive material connected to diaphragm
- inexpensive, rugged
- often used as stage mics

CONDENSOR MICS: needs charge (phantom power)

- ex: Neumann U87
- diaphragm electrically charged & mounted parallel to metallic backplate
- moving diaphragm changes dist. b/t diap. & backp. electric signal proportional to dist.
- condensor = old word for capacitor
- large diaphragm: highly colored, essential for vocals
- small diaphragm: "pencil mic", less colored (neutral), better for capturing multiple inst. (drum kit, ..)

PHANTOM POWER:

- typically 48 V
- sound is AC signal. power is DC. can provide power & have sound riding on top.
- turn on @ pre-amp.

PIEZO MICS

- structure of crystal generates V when deformed
- frequently used as contact mics (pick up vibrations travelling through solid objects)
- apply charge → deforms.
- Ex: guitar plugged into amp → piezo inside guitar pick up sound
- easy, but not good sound

RIBBON MIC:

- one of earliest bidirectional mic designs
- moving conductive elt = corrugated ribbon of metal (diaphragm)
- extremely thin

MEMS (micro-electromechanical system)

- tiny electronic part containing mic, preamp, ADC
- in phones, etc.
- inexpensive PCB mounted

Mic comparisons

	Dynamic	Condensor	Piezo	Ribbon
Price	Low	High	Very low	High
Sensitivity	Good	Excellent	Low	Varies
Durability	Excellent	Medium	???	Low*
Phantom power	No	Required	No	No
Tolerance to high SPL	Excellent	Med/Good	Variable	No

Mic placement tips

- Your recording is only as good as the acoustic instrument
  - make sure it is in tune, sounds good, is played well
- Use a good acoustic environment: minimal background noise, HVAC
- match frequency response of mic to instrument
  - shaped sound for vocals and bright instruments
- Put on headphones and listen to the sound of the microphone as you move it around
  - acoustic instruments make many different sounds
  - different mic positions will emphasize certain characteristics
- Mic distance will have a big effect
  - close to improve signal level, also emphasize that sonic perspective
  - back off to get the interaction of instrument and room, potentially more natural

11/13 **STEREO RECORDING:**

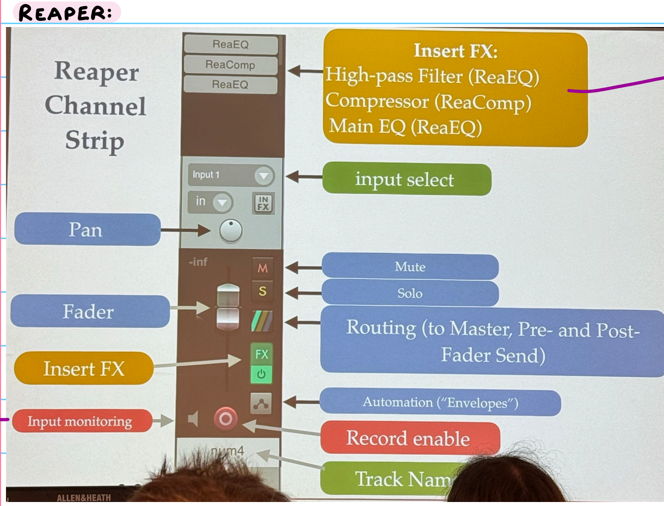
- XY technique
- mid-side recording: uses 2 mics – one directional for mid, and one figure-8 with audio src on side

VOLUME UNITS:

- arbitrary unit to visualize loudness

DAW decidels: (full scale) for digital system, where 0 indicates max amplitude before clipping

REAPER:

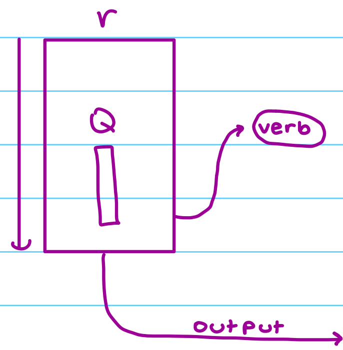


whether you want to hear your voice while singing (going through system can have latency)

click Fx in Reaper

- VST: ReEq
- VST: ReaComp
- VST: ReEq

in order



track → verb
track ↓ output
verb ↙ output

track
↓
verb
↓
output

REVERB: feels more natural

- dry signal: sound before adding reverb
- wet signal: sound after adding reverb

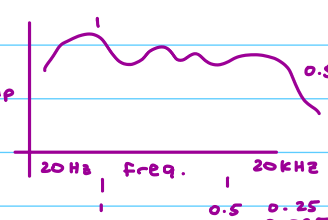
limiting
(happens slowly)

11/12 **DELAY:**

- short delay usually destructive
- delay-based reverb

CONVOLUTION:

- impulse: brief strike of noise (theoretically ∞ short imp. w/ energy = dist. throughout freq. spec.)
- impulse response: system's reaction to impulse (can also use sine sweep
- useful when we want to cross synthesize
- often used in reverb, IR guitar pedal
- convolution (impulse resp.) is good way to model guitar amplifier
- modelling guitars, filtering, noise red.



11/17 **MASTERING:**

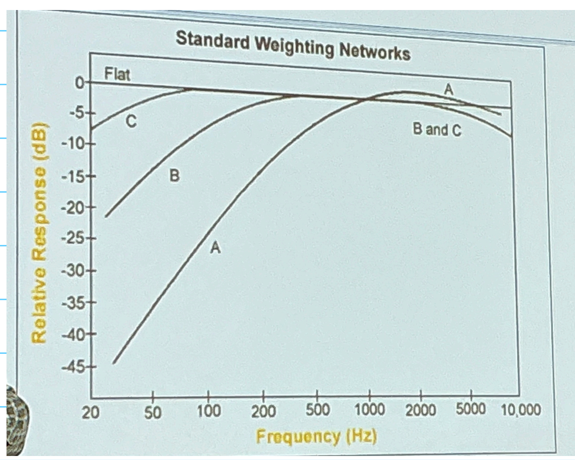
- prepping track to be distributed (optimize)
- classic problems w/ vinyl: too much bass → causes needle to skip.
  has weird freq. curve

LOUDNESS WARS:

- louder sounds have diff freq. resp.
- overcompressing music loses dynamic transients
- listener controls final volume, not producer

LOUDNESS UNITS (LUFS)

- make up for dB RMS weaknesses
  - monitors amp, freq. weighting, measurement techniques
- goal: objective measurement of how loud track is



*most use A weighting (standard)
C is for measuring super loud things

GOALS OF MASTERING:

- make sure track sounds good on all systems
- matching commercial loudness levels
- sequencing & balancing an album
- last chance to sculpt sound of your track

CRITICAL LISTENING:

- correct setup
- ~ 85 dB (not too loud, but overcoming loudness curve)

MASTERING CHAIN

- compression (optional)
- EQ
- multiband compression (optional)
- limiter

LOSSLESS COMPRESSION: reduces file sizes while allowing for perfect reconstruction of original size

- LOSSY: losing info (mp3 / aac)

12/1 HEAD MASKING:

- higher freq: tends to get into head
- lower freq: tends to not reach ears

12/3: AI IN MARKOV CHAINS:

- MARKOV CHAIN: predicts what comes next based on what just happened

USING AI / STYLES:

- timbre transfer
- vocaloid (Hatsune Miku)