GCT535: Sound Technology for Multimedia

Digital Filters, EQ and Dynamic Range Control

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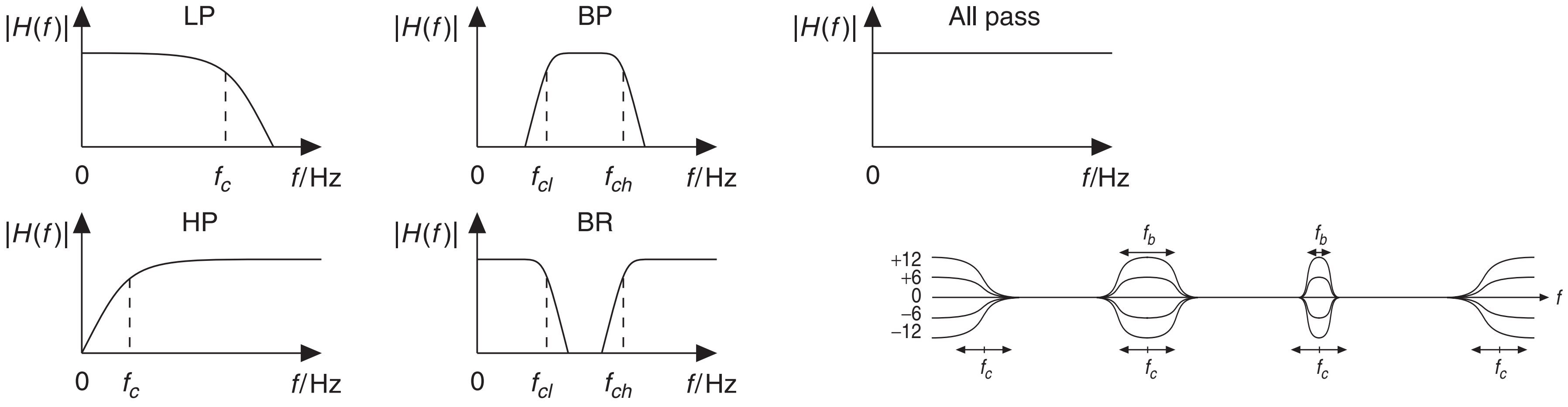
Digital Audio Effect

- Loudness
  - Volume, compressor, expander
- Pitch/Length
  - Resampling, time-scale modification, pitch-shifting
- Timbre
  - Filter, EQ, Chorus, Flanger
- Spatial effect
  - Delay, HRTF, Reverberation

Introduction

- Filters and EQ
  - Control the frequency response of input signals
  - Types: lowpass, highpass, bandpass, bandreject (Notch), equalizers
  - Shape the spectrum of input signals



[DAFx book]

Applications

- Tone control
  - Synthesizers
    - VCF in Analog synth
  - Guitar Effect
    - EQ, Wah Wah
  - DJ mixer, audio Mixers
    - Filter, EQ
  - Audio Players
    - EQ

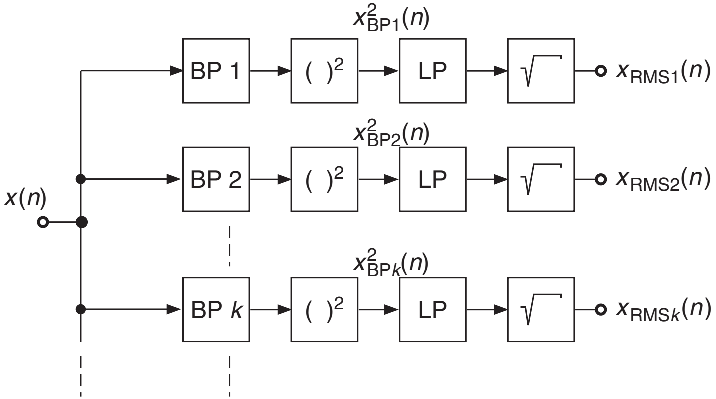
Synthesizer (MiniMoog)

Guitar Wah-Wah Pedal

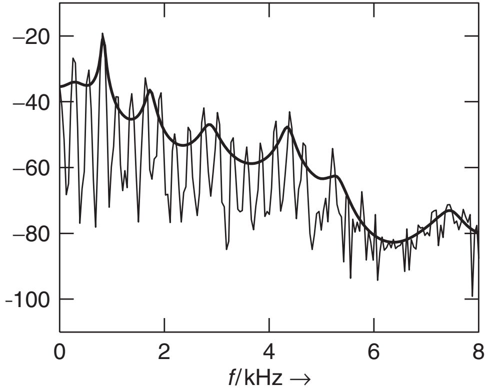
Audio Mixer

Applications

- Communication
  - Speech Coding
    - Vocoder: analysis/synthesis of voice
    - Formant modeling: linear prediction coding
  - Audio Coding
    - Filterbank in MP3
  - Band-limiting
    - Control bandwidth: 8K, 16K
    - Anti-aliasing filters



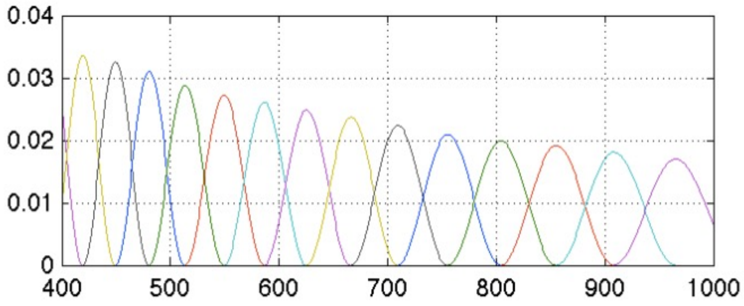
Channel Vocoder



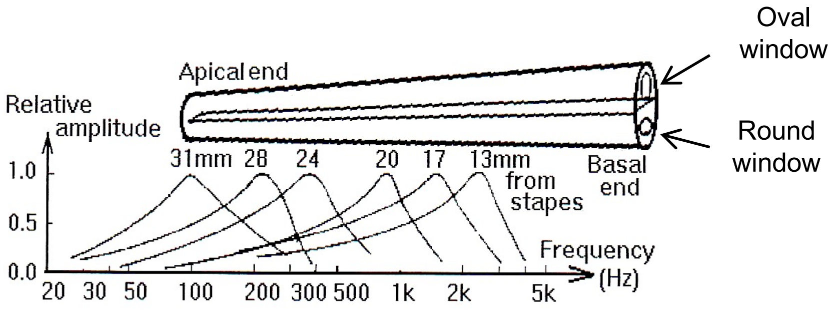
LPC modeling of Formant

Applications

- Audio analysis
  - Constant-Q transform
  - Mel-scaled filterbank
  - Cochlear filterbank in human ears



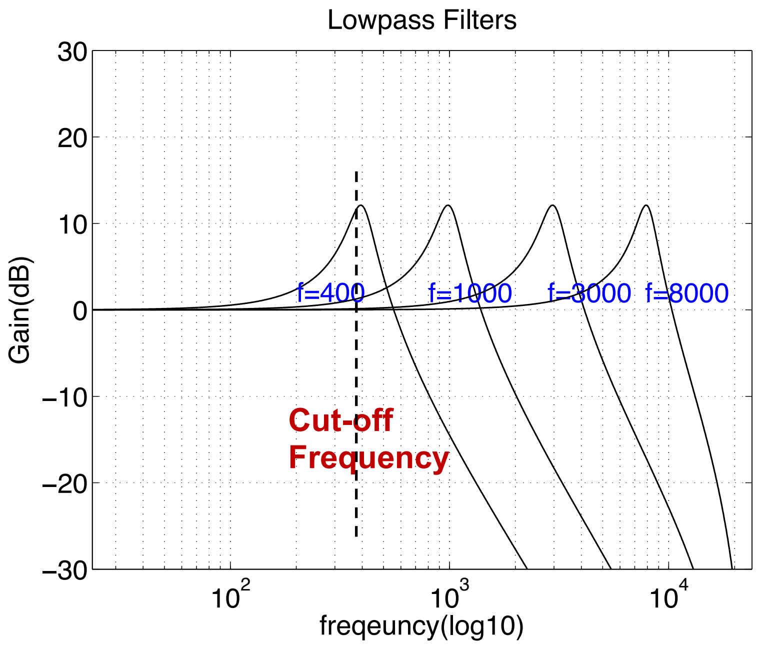
Mel filterbank

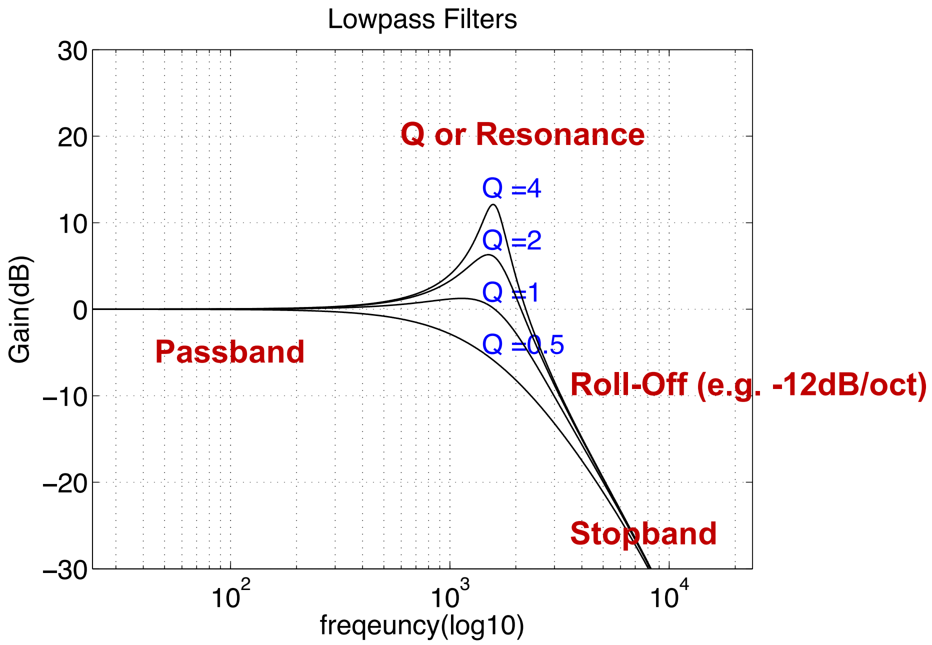


Cochlear Filterbank

Description of Filter Characteristic

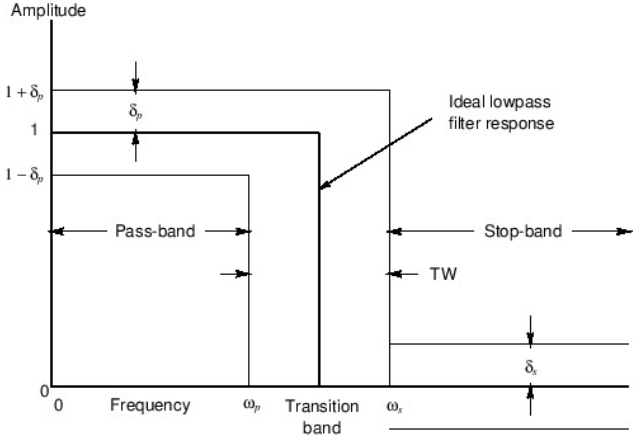
- Bode Plot
  - Log scale in both amplitude and frequency axes





Designing Digital Filters

- Approach 1: find filter coefficients that satisfy “filter specification”
  - Low-order filters
    - Fixed gain at DC (0 Hz) and the Nyquist frequency
    - Cut-off frequency and Q
  - High-order filters (advanced topic)
    - Passband/Stopband margin
    - Width of transition band
  - Methods for high-order filter design
    - Window-based method
    - Parks-McClellan: Iterative search
    - Linear programming or convex optimization
  - Fitting a filter to measured frequency response
    - Linear prediction coding (LPC)



filter specification

Designing Digital Filters

- Approach 2: digitizing analog filters which is already well-developed
  - Low-order filters
    - Use the “prototype” analog filters
  - High-order IIR filters: Butterworth filter, Chebyshev filter, Elliptical filter

- We will focus on designing low-order filters (first and second order) in this course
  - Mapping user parameters to filter coefficients

One-pole One-zero Filters

- Three coefficients (b_0, b_1 and a_1) can be obtained from the following conditions
  - Fixed gain at DC and Nyquist frequency
    - Lowpass filter: $H\left(e^{j0}=1\right)=1$ (DC) , $H\left(e^{j\pi}=-1\right)=0$ (Nyquist)
    - Highpass filter: $H\left(e^{j0}=1\right)=0$ (DC), $H\left(e^{j\pi}=-1\right)=1$ (Nyquist)
  - User parameter
    - Cut-off frequency when the gain is -3dB → $\left\|H\left(e^{j\omega}\right)\right\|^2=\frac{1}{2}$

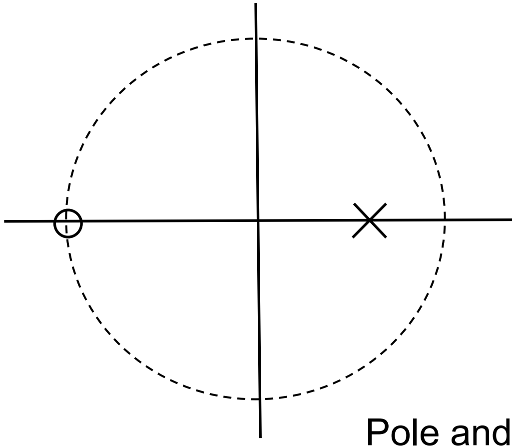
$$H(z)=\frac{B(z)}{A(z)}=\frac{b_0+b_1\cdot z^{-1}}{1+a_1\cdot z^{-1}}$$

Lowpass filter

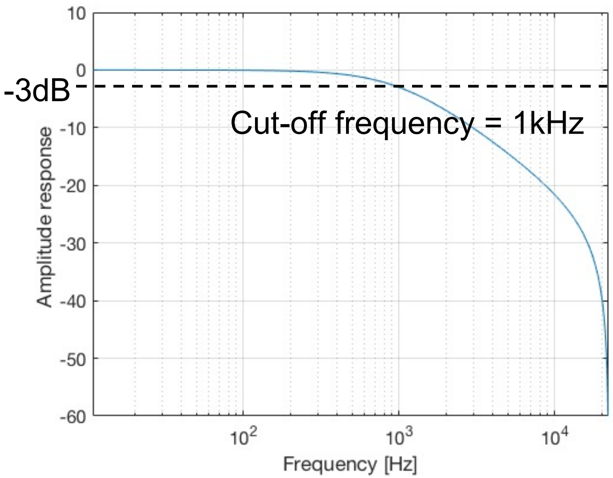
- Fixed gain at DC and Nyquist frequency and cut-off frequency

$$H\left(e^{j0} = 1\right) = 1$$
$$H\left(e^{j\pi} = -1\right) = 0$$
$$\left\|H\left(e^{j\omega_c}\right)\right\|^2 = \frac{1}{2}$$

$$\longrightarrow \quad H(z) = \frac{1-a}{2} \cdot \frac{1+z^{-1}}{1-a \cdot z^{-1}} \qquad a = \frac{1-\tan(\frac{\omega}{2})}{1+\tan(\frac{\omega}{2})}$$



Pole and zero



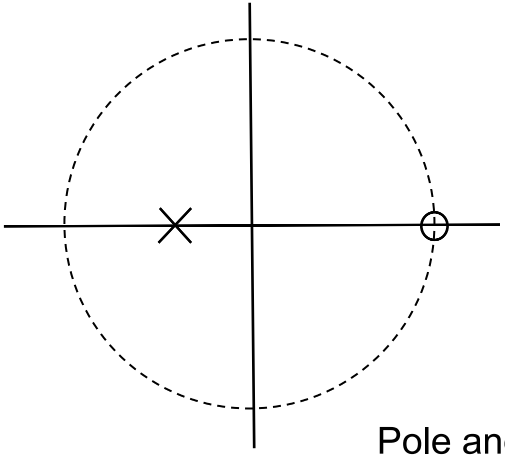
Amplitude response

Highpass filter

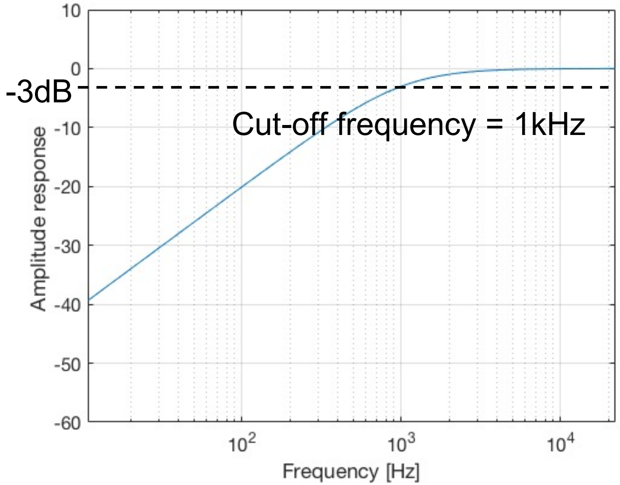
- Fixed gain at DC and Nyquist frequency and cut-off frequency

$$H\left(e^{j0}=1\right)=0$$
$$H\left(e^{j\pi}=-1\right)=1$$
$$\left\|H\left(e^{j\omega_c}\right)\right\|^2=\frac{1}{2}$$

$$\longrightarrow \quad H(z)=\frac{1+a}{2}\cdot\frac{1-z^{-1}}{1-a\cdot z^{-1}} \qquad a=\frac{1-\tan(\frac{\omega_c}{2})}{1+\tan(\frac{\omega_c}{2})}$$



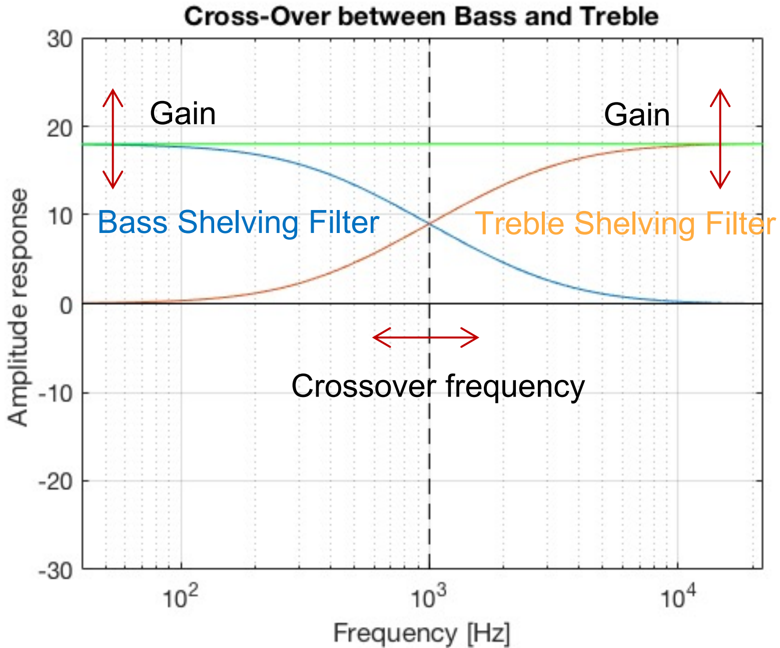
Pole and zero



Amplitude response

One-pole One-zero Shelving Filter

- Equalizers that adjust gain in the low or high frequency range
  - User control: 1) **gain** and 2) **crossover-frequency**
  - Crossover frequency is set **when the gain becomes a half in dB**



The cascade of bass and treble shelving filters renders flat response.

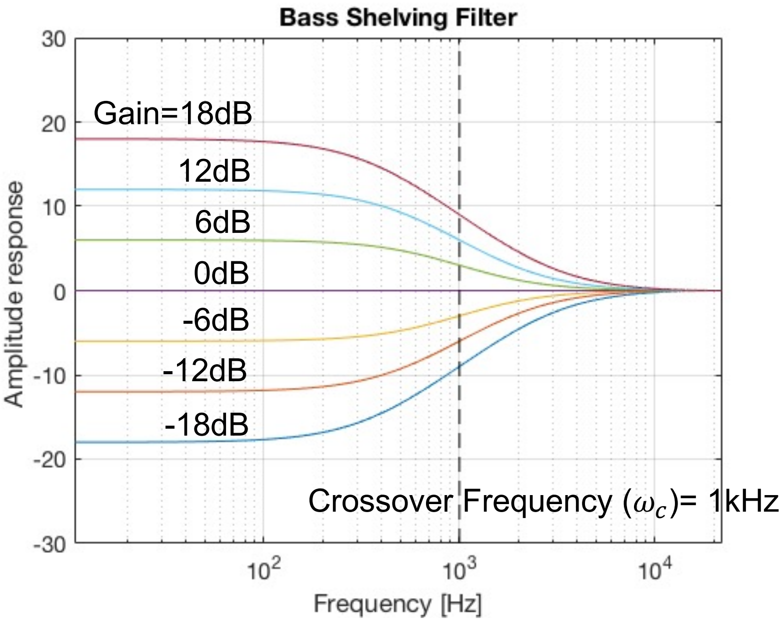
Bass Shelving Filter

- Equalizers that control gain in the low frequency range

$$H(1) = G,\;\; H\left(e^{j\omega_c}\right) = \sqrt{G}\,,\;\; H(-1) = 1$$

$$\downarrow$$

$$H(z) = \frac{\left(G\tan\left(\frac{\omega_c}{2}\right) + \sqrt{G}\right) + \left(G\tan\left(\frac{\omega_c}{2}\right) - \sqrt{G}\right)\cdot z^{-1}}{\left(\tan\left(\frac{\omega_c}{2}\right) + \sqrt{G}\right) + \left(\tan\left(\frac{\omega_c}{2}\right) - \sqrt{G}\right)\cdot z^{-1}}$$



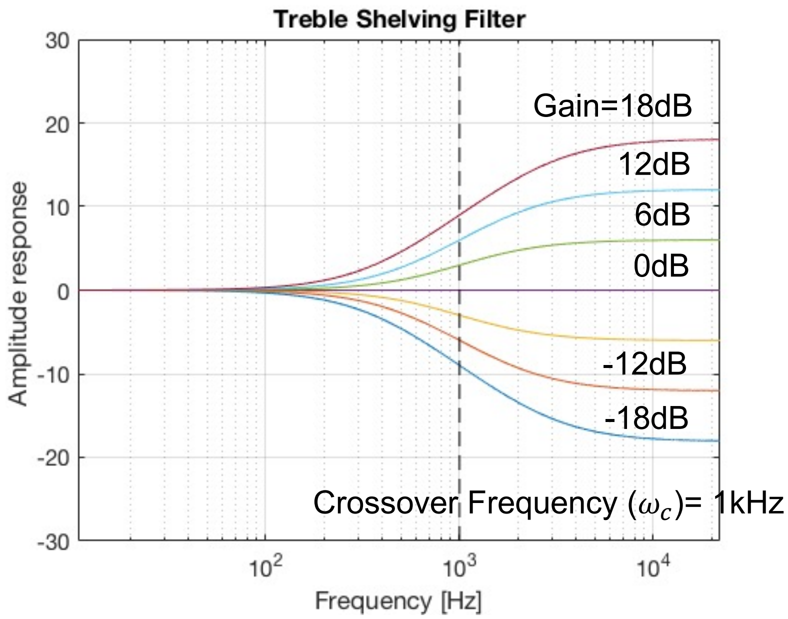
Treble Shelving Filter

- Equalizers that control gain in the high frequency range

$$H(1) = 1,\ \ H\left(e^{j\omega_c}\right) = \sqrt{G}\ ,\ \ H(-1) = G$$

$$\downarrow$$

$$H(z) = \frac{\left(\sqrt{G}\tan\left(\frac{\omega_c}{2}\right) + G\right) + \left(\sqrt{G}\tan\left(\frac{\omega_c}{2}\right) - G\right)\cdot z^{-1}}{\left(\sqrt{G}\tan\left(\frac{\omega_c}{2}\right) + 1\right) + \left(\sqrt{G}\tan\left(\frac{\omega_c}{2}\right) - 1\right)\cdot z^{-1}}$$



Biquad (two-pole two-zero) Filters

- Five coefficients (b_0, b_1, b_2, a_1 and a_2) can be obtained from the following conditions
  - Fixed gain at DC (0 Hz) and Nyquist frequency
  - User parameters
    - Cut-off frequency
    - Resonance
    - Bandwidth (sharpness of peak)

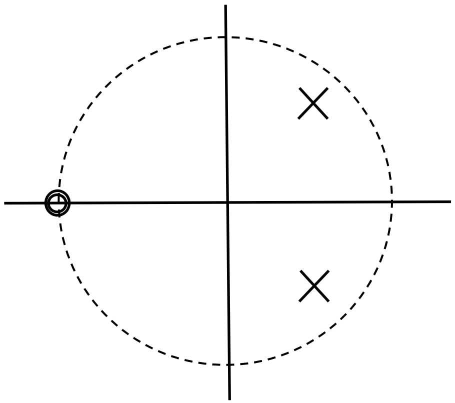
$$H_{bi}(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 \cdot z^{-1} + b_2 \cdot z^{-2}}{1 + a_1 \cdot z^{-1} + a_2 \cdot z^{-2}}$$

- Mapping user parameters to filter coefficients
  - Reson filter can be used but it is quite complicated
  - The art of analog filter design is highly advanced and so we take advantage of it → transform analog filters into discrete-time version

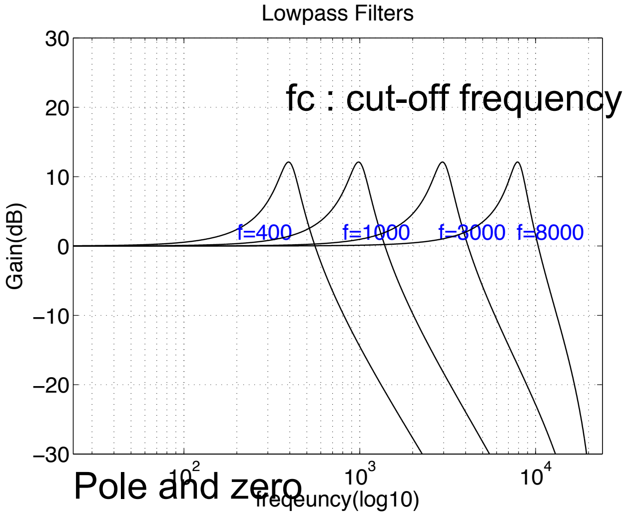
Digitized Resonant Low-pass Filter

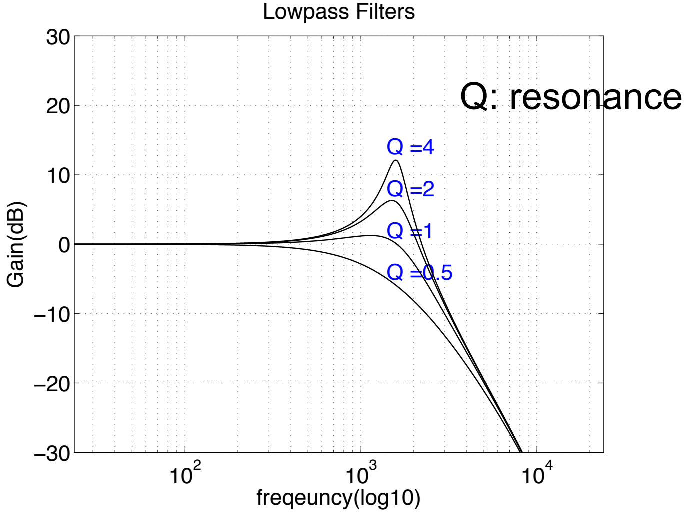
- Transfer Function

$$H(z) = \left(\frac{1-\cos\theta}{2}\right)\frac{1 + 2z^{-1} + z^{-2}}{(1+\alpha) - 2\cos\theta z^{-1} + (1-\alpha)z^{-2}} \qquad \alpha = \frac{\sin\theta}{2Q} \qquad \theta = 2\pi\frac{f_c}{f_s}$$



Pole and zero

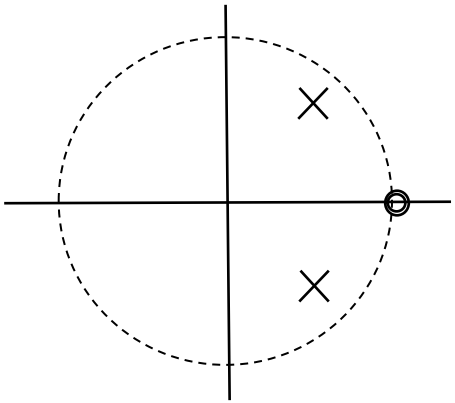


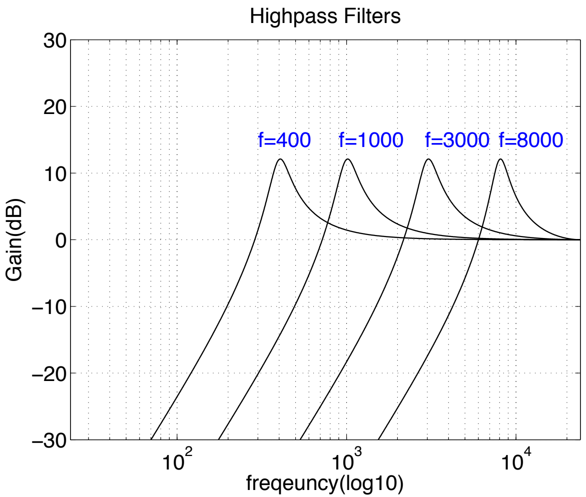


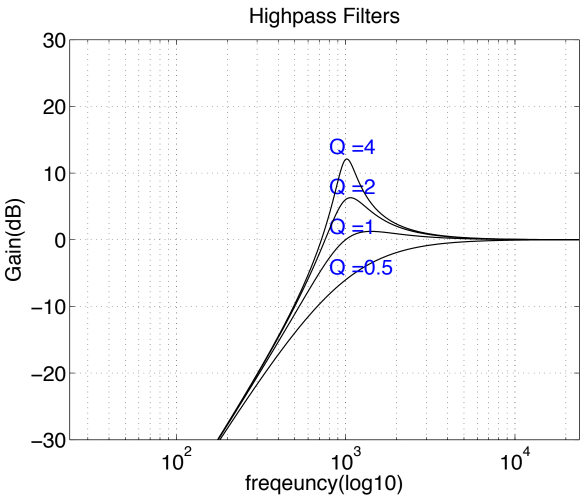
Digitized Resonant High-pass Filter

- Transfer Function

$$H(z) = \left(\frac{1+\cos\theta}{2}\right)\frac{1 - 2z^{-1} + z^{-2}}{(1+\alpha) - 2\cos\theta z^{-1} + (1-\alpha)z^{-2}} \qquad \alpha = \frac{\sin\theta}{2Q} \qquad \theta = 2\pi\frac{f_c}{f_s}$$



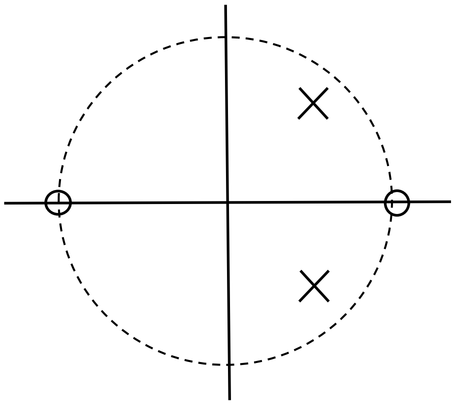


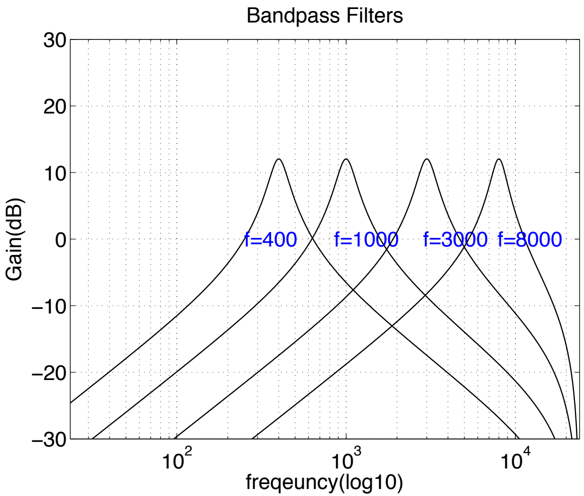


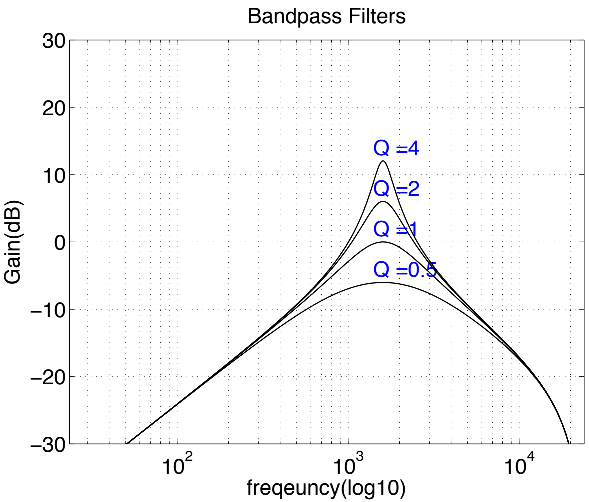
Digitized Band-pass filter

- Transfer Function

$$H(z) = \left(\frac{\sin\theta}{2Q}\right)\frac{1 - z^{-2}}{(1+\alpha) - 2\cos\theta z^{-1} + (1-\alpha)z^{-2}} \qquad \alpha = \frac{\sin\theta}{2Q} \qquad \theta = 2\pi\frac{f_c}{f_s}$$



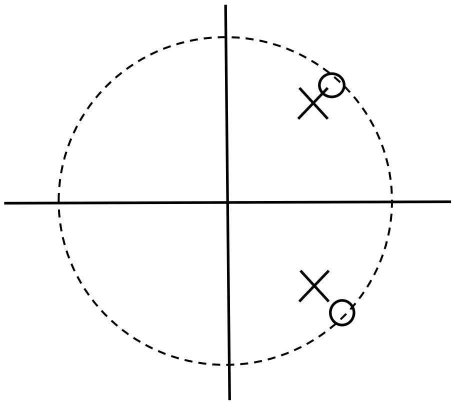


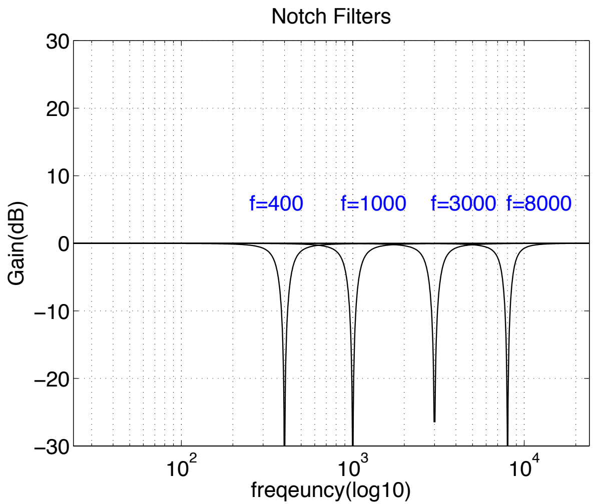


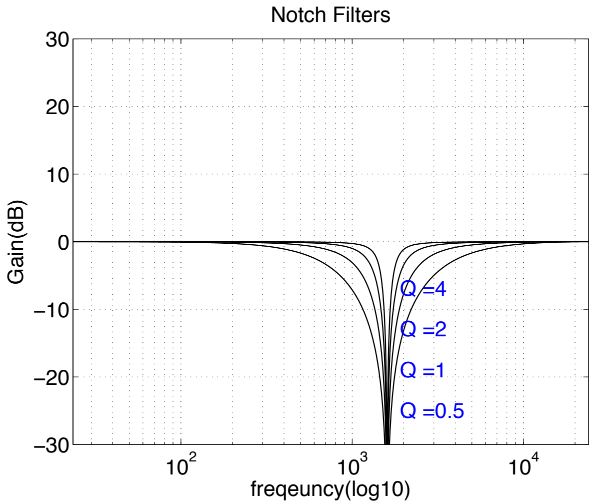
Digitized Notch filter

- Transfer Function

$$H(z) = \frac{1 - 2\cos\theta z^{-1} + z^{-2}}{(1+\alpha) - 2\cos\theta z^{-1} + (1-\alpha) z^{-2}} \qquad \alpha = \frac{\sin\theta}{2Q} \qquad \theta = 2\pi \frac{f_c}{f_s}$$



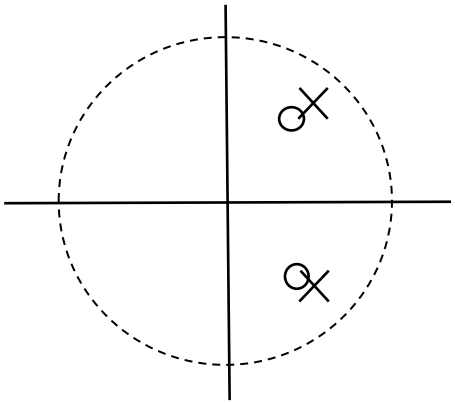


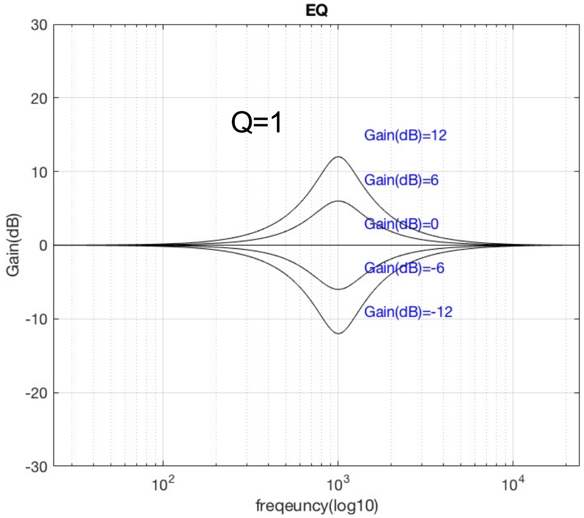


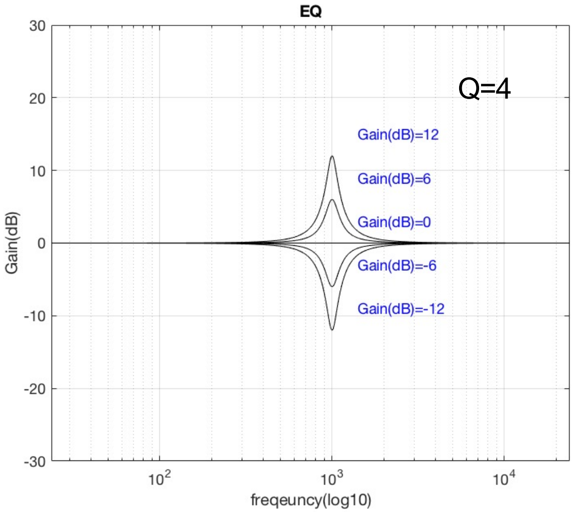
Digitized Equalizer

- Transfer Function

$$H(z)=\frac{(1+\alpha\cdot A)\,-2\cos\theta z^{-1}\,+(1-\alpha\cdot A)z^{-2}}{(1+\alpha/A)\,-2\cos\theta z^{-1}\,+(1-\alpha/A)z^{-2}}\qquad \alpha=\frac{\sin\theta}{2Q}\qquad \theta=2\pi\frac{f_c}{f_s}\qquad A=10^{\left(\frac{\mathrm{Gain(dB)}}{40}\right)}$$

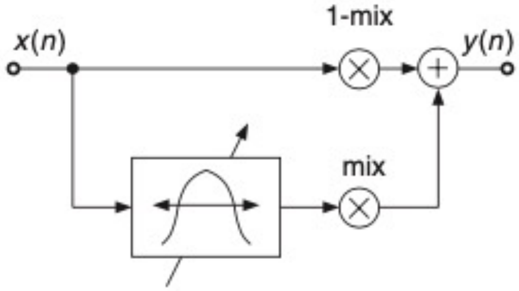






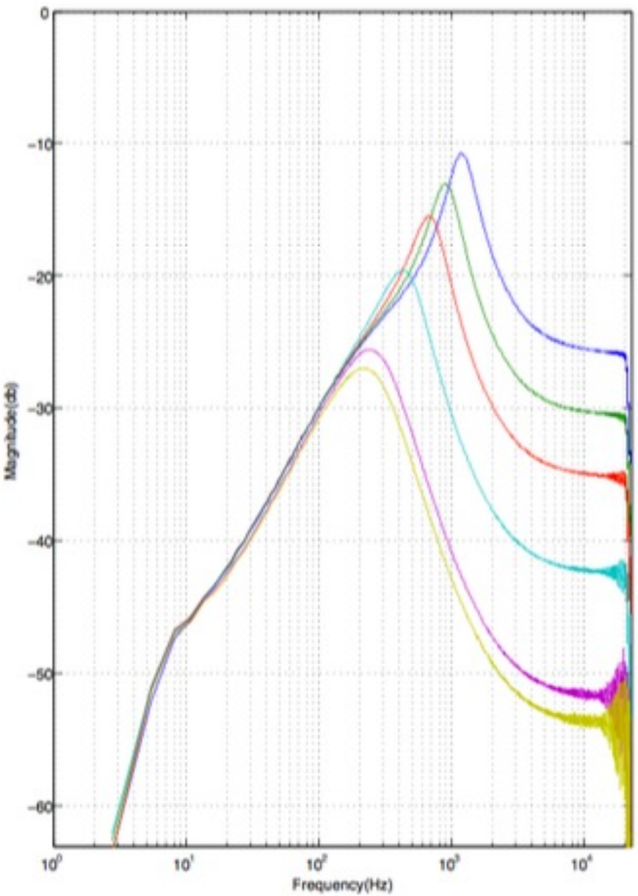
Wah-Wah Effect

- Emulate a human-voice-like sound using resonant filters
  - Bandpass filters or resonant lowpass filters model “formant”
  - The formant frequency ranges between 400 Hz and 2000 Hz, and it is often controlled by a foot pedal
    - https://www.youtube.com/watch?v=NW9Yq99FeTU
  - Implemented with a cascade of bandpass filter and resonant lowpass
    - http://www.geofex.com/article_folders/wahpedl/voicewah.htm



Wah-Wah Effect Diagram (DAFx book)

Wah-Wah Effect



Ampltiude response

Dunlop 535Q CryBaby

Auto-Wah Effect

- The center frequencies can be also controlled automatically by several controllers
  - Envelope follower
    - Compute the trajectory of amplitude envelope
  - Low frequency oscillator (LFO)
    - A sinusoid or sawtooth waveform with a low frequency (typically less than 5 Hz)
  - Used for guitar, bass guitar, clavinet, and electric piano
    - https://www.youtube.com/watch?v=aOmsLKqqJcQ (electric piano)
    - https://www.youtube.com/watch?v=Ws86GIm_jS0 (Clavinet by Stevie Wonder)

Dynamic Range Control (DRC)

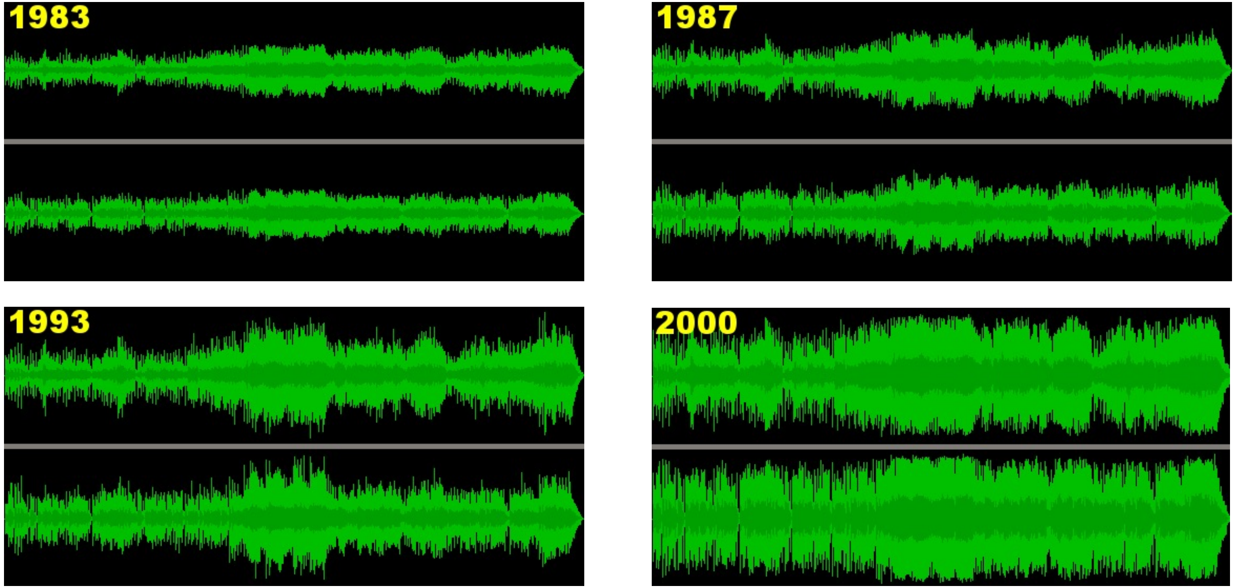
- Control the loudness of sound sources

Automation curve

- Macro-scale DRC
  - Adjust the level or over long segments: e.g. between verse and chorus
  - Remove or reduce short segments: e.g. breathing in vocal
  - Use manually drawn volume curve (automation curves) in DAW
- Micro-scale DRC
  - Automatic gain control or change timbre of a tone
  - Use signal processors: compressor, limiters, expander and noise gate

Dynamic Range Control (DRC)

- Control overall loudness in the mastering stage
  - Loudness war: competitive escalation of loudness

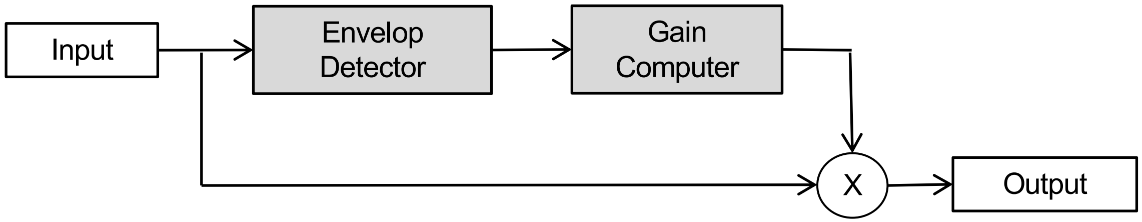


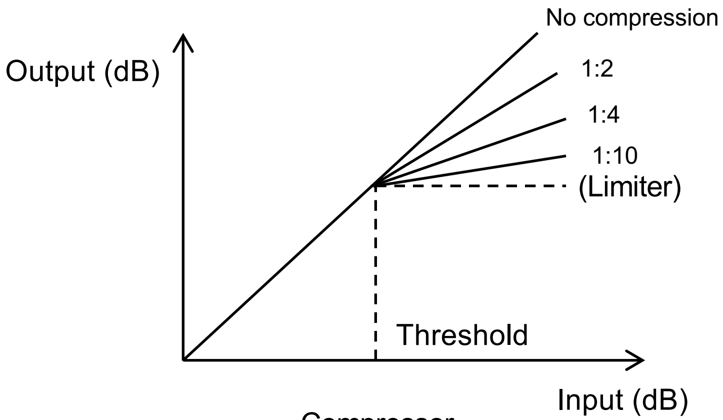
Remastering versions of “Something” by The Beatles

(source: https://commons.wikimedia.org/wiki/File:Cd_loudness_trend-something.gif)

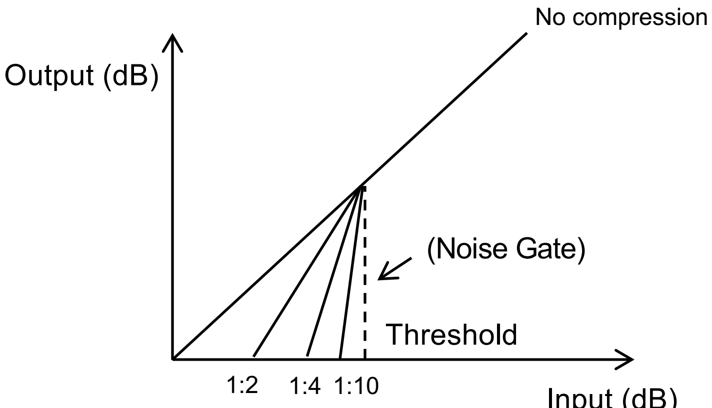
DRC Audio Effect

- Signal processing pipeline





Compressor



Expander

Envelope Detector

- Detecting the level of signal

Input → Full-wave Rectification (|x|) → Leaky Integrator (one-pole lowpass filter) → envelope

- Different sensitivity for increasing (attack) and decreasing (release) levels: control the length of the transient region of the filter
  - During attack:

$$y(n) = y(n-1) + (1 - e^{-1/(attack\_time * fs)})(|x(n)| - y(n-1))$$

  - During release:

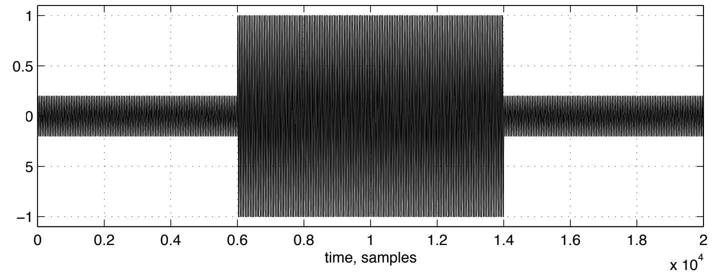
$$y(n) = y(n-1) + (1 - e^{-1/(release\_time * fs)})(|x(n)| - y(n-1))$$

Full-wave rectification

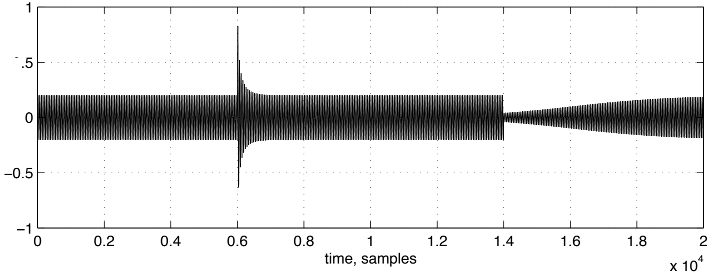
Gain Curve

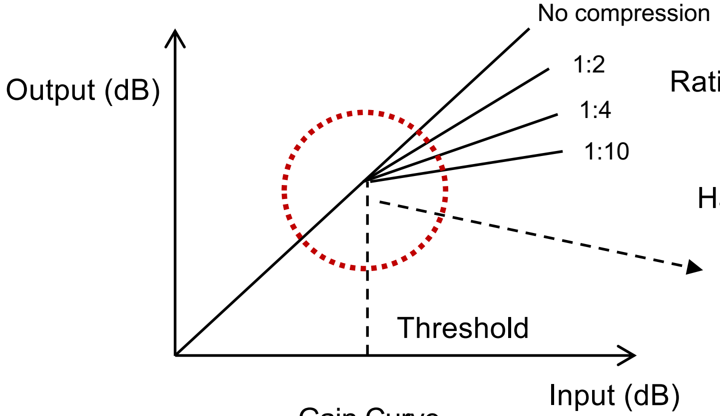
- Parameters
  - Threshold: level
  - Attack/Release: sensitivity
  - Ratio: amount of compression
  - Knee: smoothing

Before compression



After compression





Gain Curve