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Acoustics II

Audio effects

Reto Pieren

2024

Audio effects

[Zölzer, Udo. et al. 2002. DAFX – Digital Audio Effects. John Wiley & Sons.

Weinzierl, Stefan 2009. Handbuch der Audiotechnik, Springer.

Pohlmann, K.C. 2005. Principles of Digital Audio. McGraw-Hill.

Smith III, Julius O. 2010. Physical Audio Signal Processing: for Virtual Musical Instruments and Digital Audio Effects. W3K Publishing.]

Audio effects: Introduction

- Audio signal processing
- Today mostly <u>digital</u> audio signal processing (PC or DSP)

- Alteration of the original sound for:
  - adjustment of a transmission effect (e.g. frequency response)
  - reduction of the audibility of unwanted signal components
  - mimiking a source behavior or location
  - …
  - creation of new sounds

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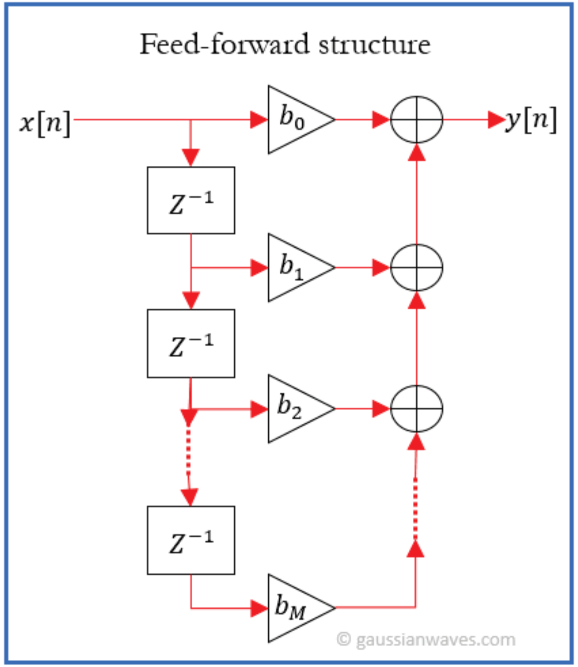
Digital filters: Finite Impulse Response (FIR)

- Finite IR length
- Only forward processing
- Always stable
- Computationally expensive
- Simpler to design

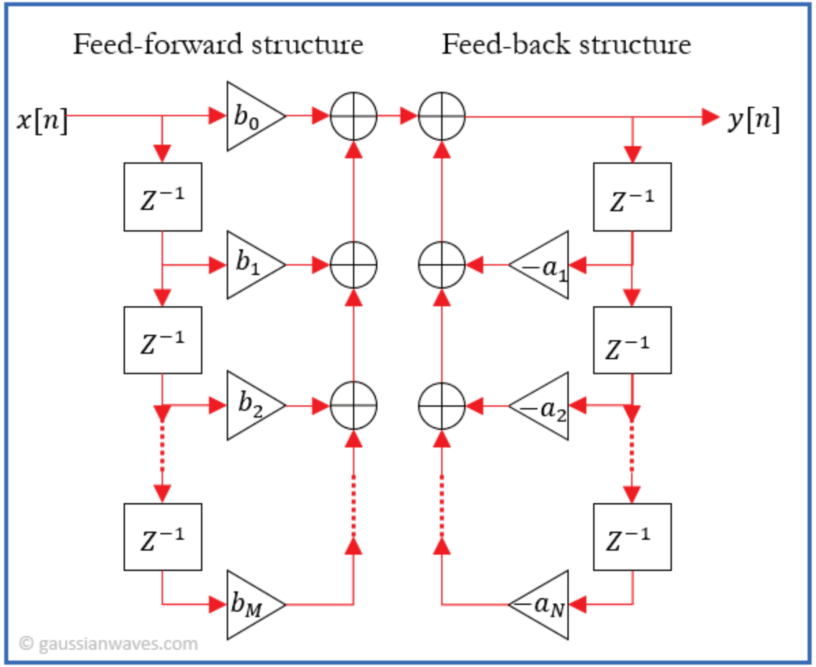
- Order = number of delays = IR length -1



Digital filters: Infinite Impulse Response (IIR)

- Contains feedback loops
- Generally IR does not drop to 0
- Compact
- Efficient
- May be unstable

- Order = min. number of needed delay elements



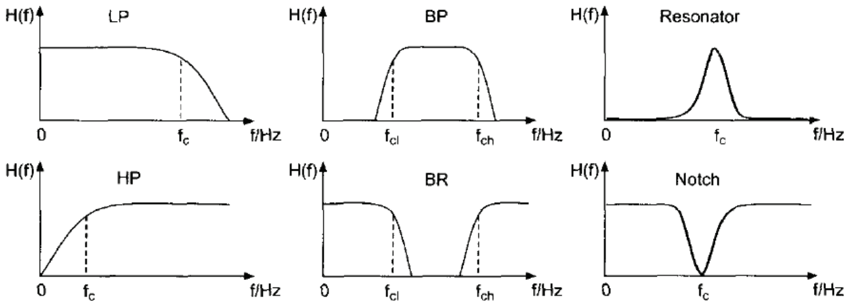
Equalizer

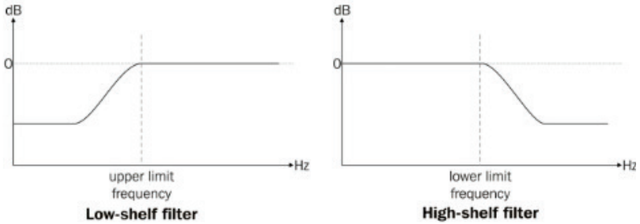
Equalizer (EQ)

- Function:
  - manipulation of the amplitude response of a sound or a transmission system
- Applications:
  - Tone control
  - flatten a non-ideal frequency response of a loudspeaker
  - loudness filtering
  - equalization of headphones
  - attenuation of the low frequency end in PA systems for speech
- LTI system = full characterization by frequency/impulse response

[Välimäki, Vesa & Reiss, J.D. 2016. All About Audio Equalization: Solutions and Frontiers. *Applied Sciences* 2016(6).]

Equalizer (EQ): Elementary filter types





- Low-pass and High-pass (LP, HP)
- Bandpass and Bandstop (BP, BS)
- Peaking/Bell and Notch
- Low shelf and High shelf
- All-pass = phase equalizer

$$|H(j\omega)| = 1$$

First order all-pass:

$$H(j\omega) = \frac{j\omega - \omega_0}{j\omega + \omega_0}$$

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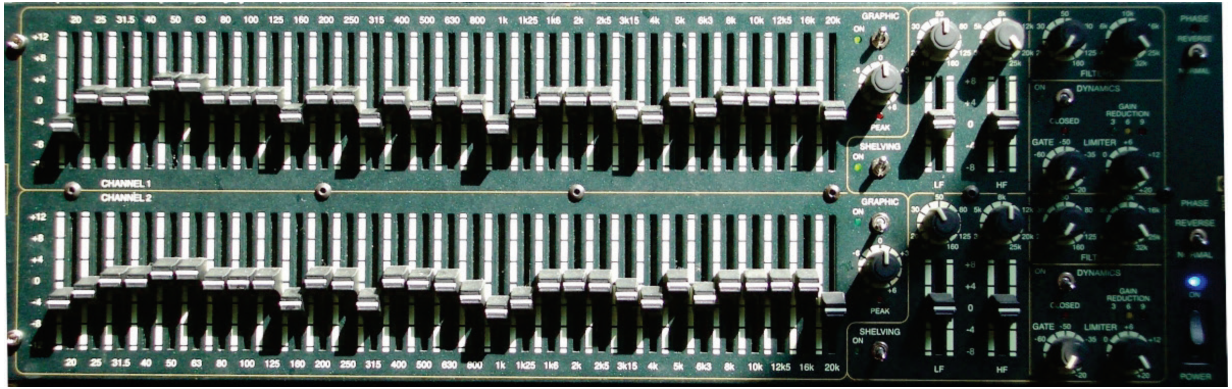
Equalizer (EQ): Graphical equalizer

- Parallel bandpass or cascaded bell filters of constant relative bandwidth
  - 1/3-octave filters (typ. 31 bands), Q=4
  - octave filters (typ. 10 bands), Q=1.4
- Adjustable amplication/attenuation in each band (typ. ±12 dB)
- Graphical representation of the amplitude response



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Equalizer (EQ): Adjustments in 1/3 octave bands

- Listening test: pink noise, amplification of the 800 Hz third octave band with normalization of total level
- → answers: *clearly -, just -, not audible*

1	A	B
2	A	B
3	A	B
4	A	B

Equalizer (EQ): Adjustments in 1/3 octave bands

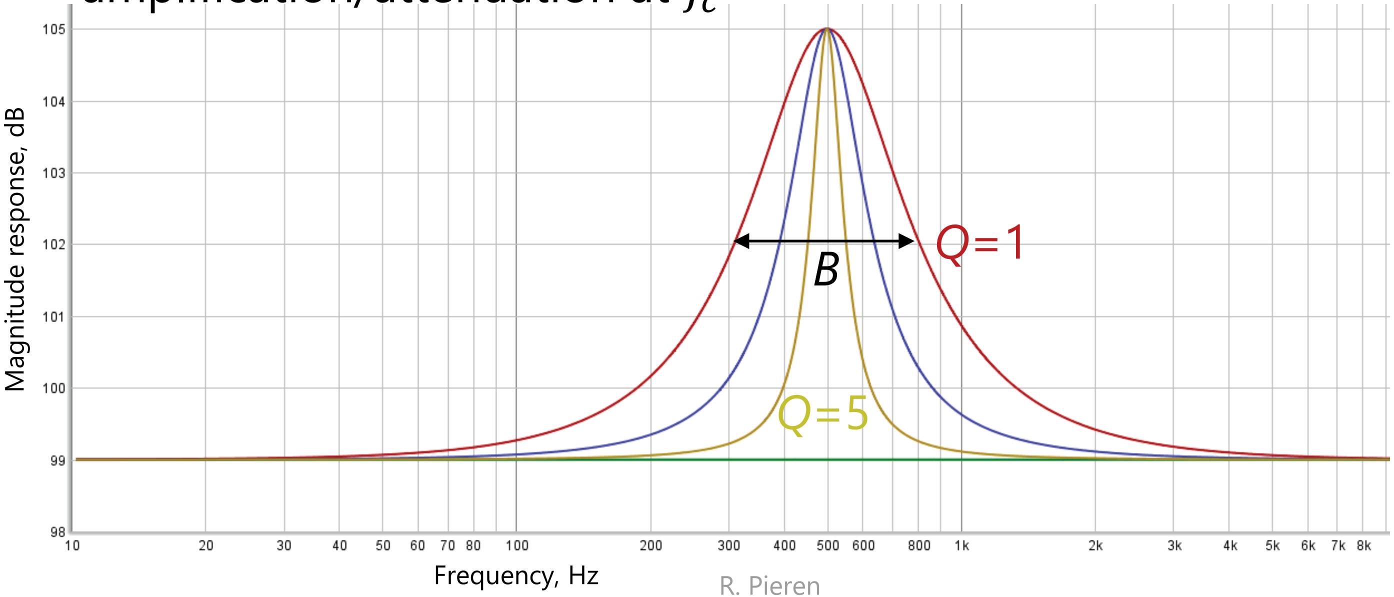
- Listening test: pink noise, amplification of the 800 Hz third octave band with normalization of total level
- → answers: *clearly -, just -, not audible*

1	A: flat	B: +9 dB
2	A: flat	B: flat
3	A: flat	B: +3 dB
4	A: flat	B: +6dB

Equalizer (EQ): Parametric equalizer

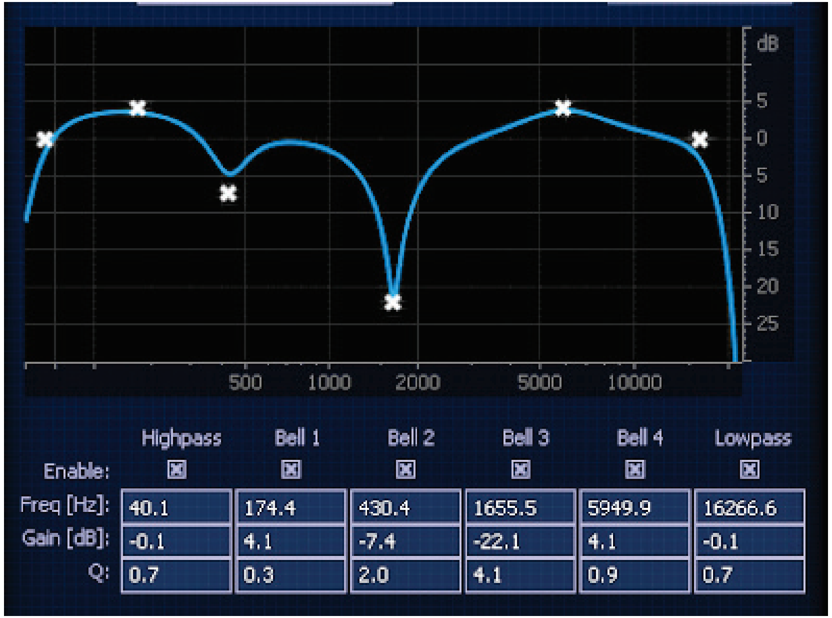
- Cascade of bell filters with variable
  - center frequency f_c
  - bandwidth B, or $Q = f_c/B$
  - amplification/attenuation at f_c

$$Q = \frac{2^{n/2}}{2^n - 1}, n\text{–octave bandwidth}$$



Equalizer (EQ): Parametric equalizer

- Example of software implementation
  - Bells + highpass and lowpass



Equalizer (EQ): Parametric equalizer

- Universal applicability
- Not that intuitive to reproduce a predefined amplitude response
- Can be easily made time-variable if realized by IIR filters

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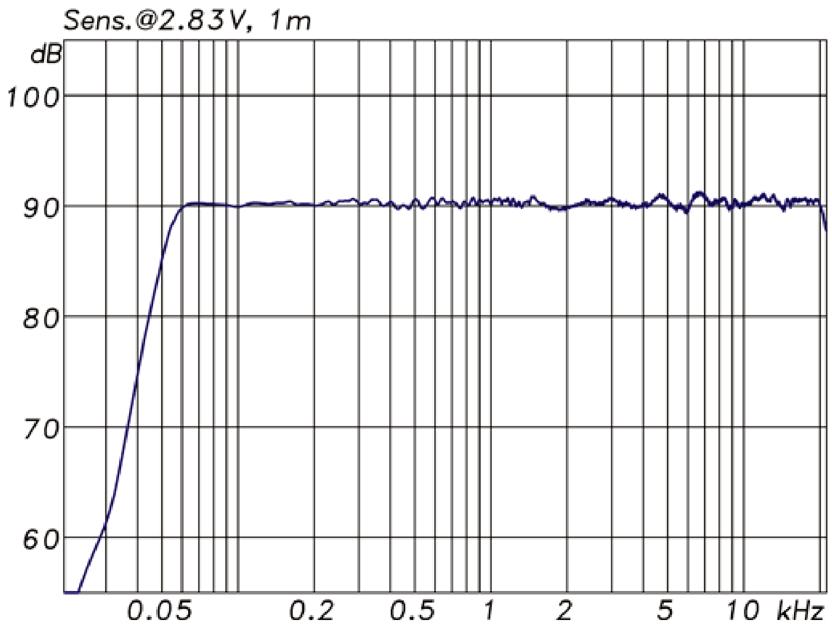
Equalizer (EQ): Example Low cut

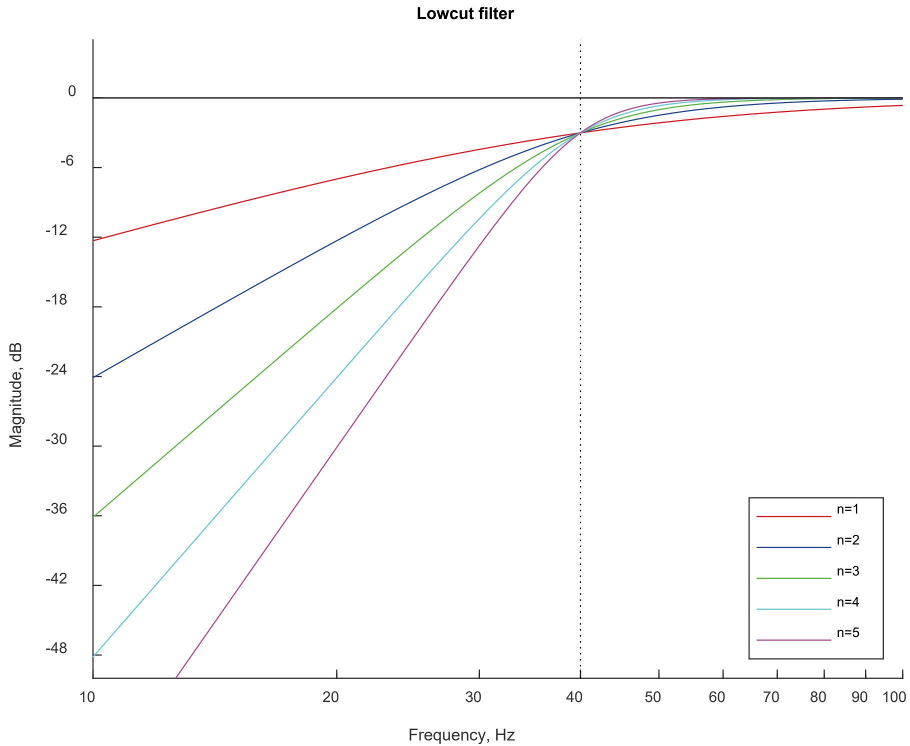
- Application
  - Assumption: Playback system limited by lower cutoff frequency of 50 Hz
  - Signal components below
    - unneccessary burden for amplifier
    - limiting signal dynamic range
  - → Low cut filter



2-way studio monitor Neumann KH 120

Equalizer (EQ): Example Low cut

- Often used type: Butterworth high-pass filter



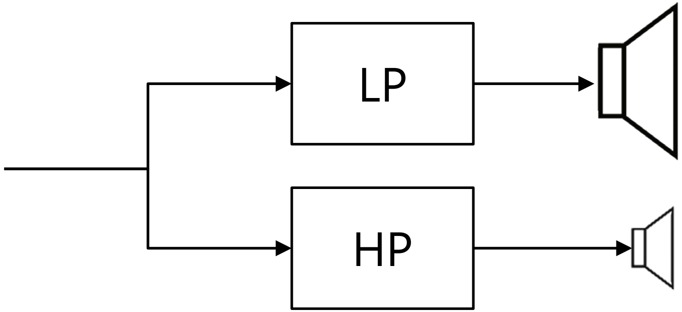
- Maximally flat in passband
- Slope = 6*n* dB/octave

n: filter order

f_c = 40 Hz

Equalizer (EQ): Example Crossover

- Application:
  - Subwoofers for low frequency content
  - Crossover filtering for subwoofers and satellites at ~100 Hz



- First-order filtering (Butterworth=Bessel):

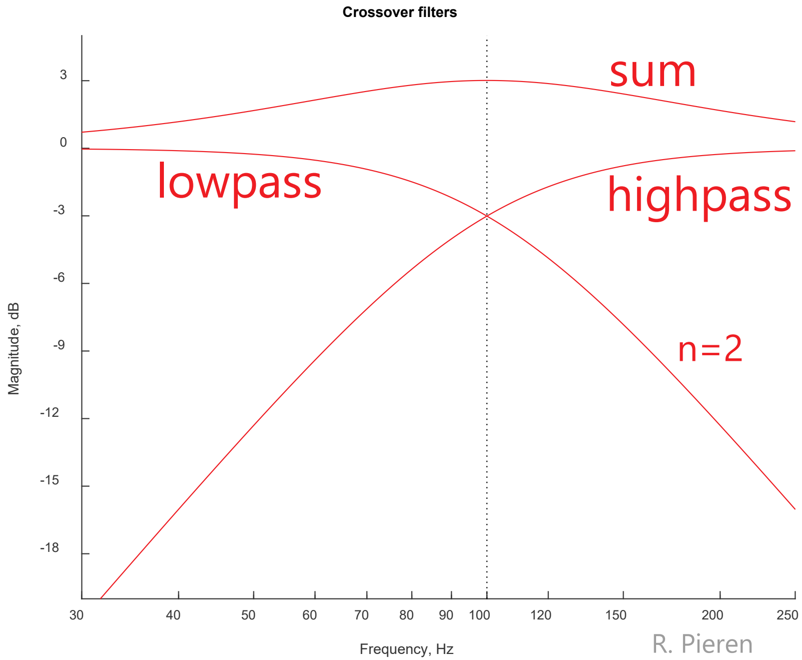
$$H_{LP} = \frac{\omega_0}{s+\omega_0}\,;\; H_{HP} = \frac{s}{s+\omega_0}\,;\; H_{tot} = H_{LP}+H_{HP} = \frac{s+\omega_0}{s+\omega_0} = 1$$

→ sums up to 1 (ideal solution)

(90° phase shift between outputs)

Equalizer (EQ): Example Crossover

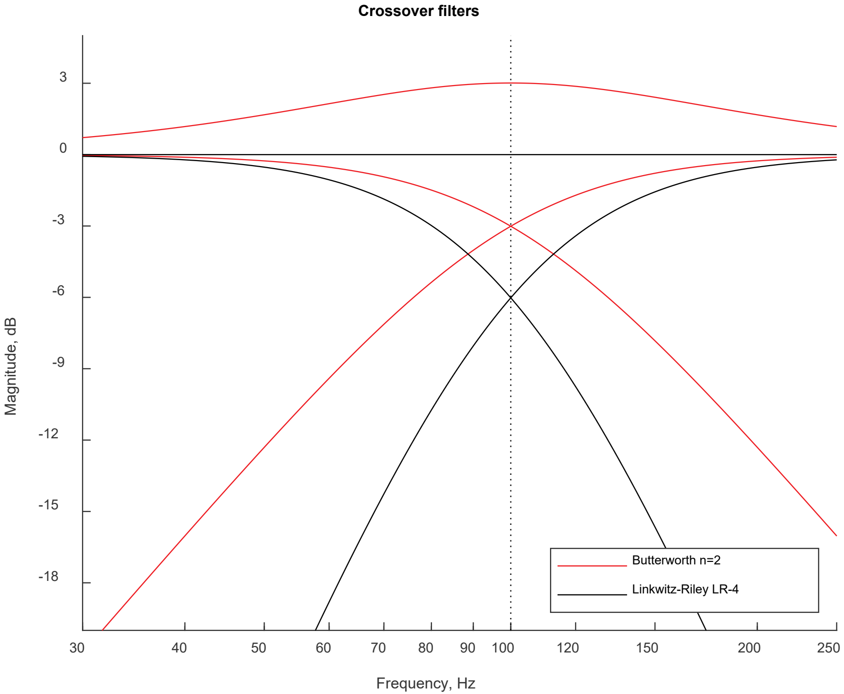
- Smaller transition region required → steeper filters → higher order filters
- Butterworth filters?



- Sum at crossover not 0 dB
- phase shifts between outputs (multiple of 180°)

Equalizer (EQ): Example Crossover

- Linkwitz–Riley (LR)-filter = 2 cascaded Butterworth



- Sum at crossover = 0dB
- Outputs in phase
- 4th order LR crossover = cascade of two 2nd order Butterworth → 24 dB/oct

[Linkwitz, S.H. 1976. Active crossover networks for noncoincident drivers. *Journal of the Audio Engineering Society* 24(1). https://www.ranecommercial.com/legacy/note160.html]

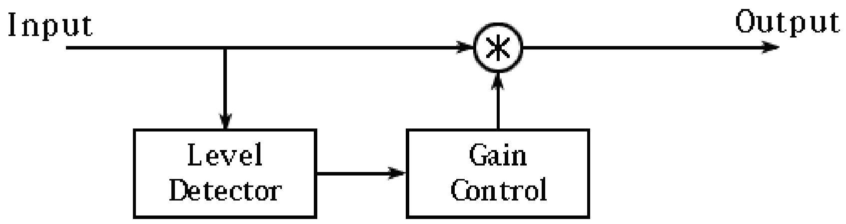
Dynamics

Compressor

Compressor/limiter

- Function:
  - reduction of signal dynamics → non-linear operation
- Applications for compressors:
  - increase of loudness with given maximal signal amplitude
    - music production
    - commercials
    - radio stations (listeners are often in noisy environment)
  - homogenization of unstable signal
  - artistic feature (e.g. audibility of breathing in a voice)
- Applications for limiters: avoid clipping in digital recordings

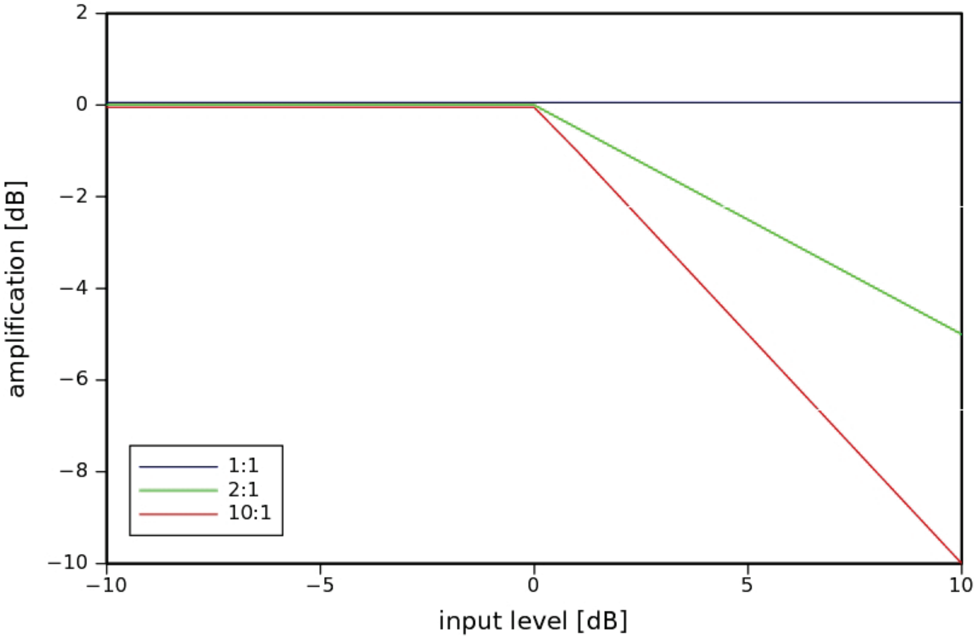
Compressor/limiter: Principle of operation



- Analog: Voltage Controlled Amplifier
- Steering voltage: *smoothed* input signal
- Reduced amplification above an adjustable threshold

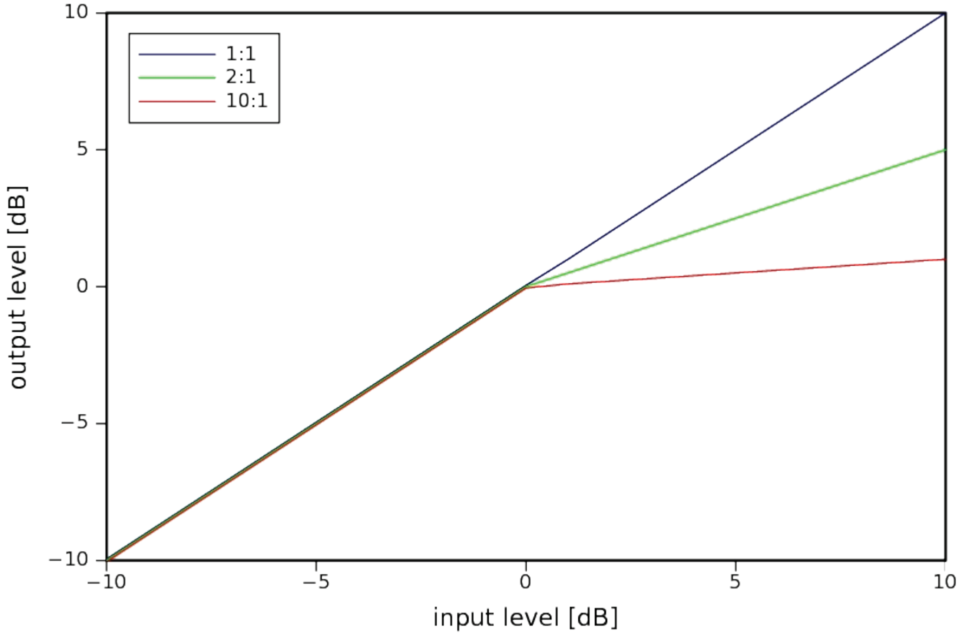
Compressor/limiter: Characteristics

- Parameters: Level *threshold* (here = 0dB) and *ratio*
- Input level → Amplification



Compressor/limiter: Characteristics

- Parameters: Level *threshold* (here = 0 dB) and *ratio*
- Input level → Output level



10:1
9 dB attenuation at 10 dB above threshold

- Limiter: ∞:1

Compressor/limiter: Time constants

- Amplification is steered by the level of the audio signal
- Determination of level needs an averaging process
- Different averaging types (*RMS*, *peak*)
- Typical use of two time constants:
  - *Attack* time: for signal increase (positive slopes)
  - *Release* time: for signal decrease (negative slopes)

Compressor/limiter: Time constants

- Choice of constants is critical
  - ! Attack too small → distortion of transients
  - ! Release too small → pumping: modulation of signal by dominant components
  - ! Attack/release too large → signal dynamics is not properly reduced due to delayed reaction
- Typical values (highly dependent on audio material)
  - Attack: order of milliseconds
  - Release: 10 – 3000 ms

Compressor/limiter: Example

- Original sample:

- Compression starting 25 dB below full-scale
  → approx. 10 dB higher signal power at identical peak value:

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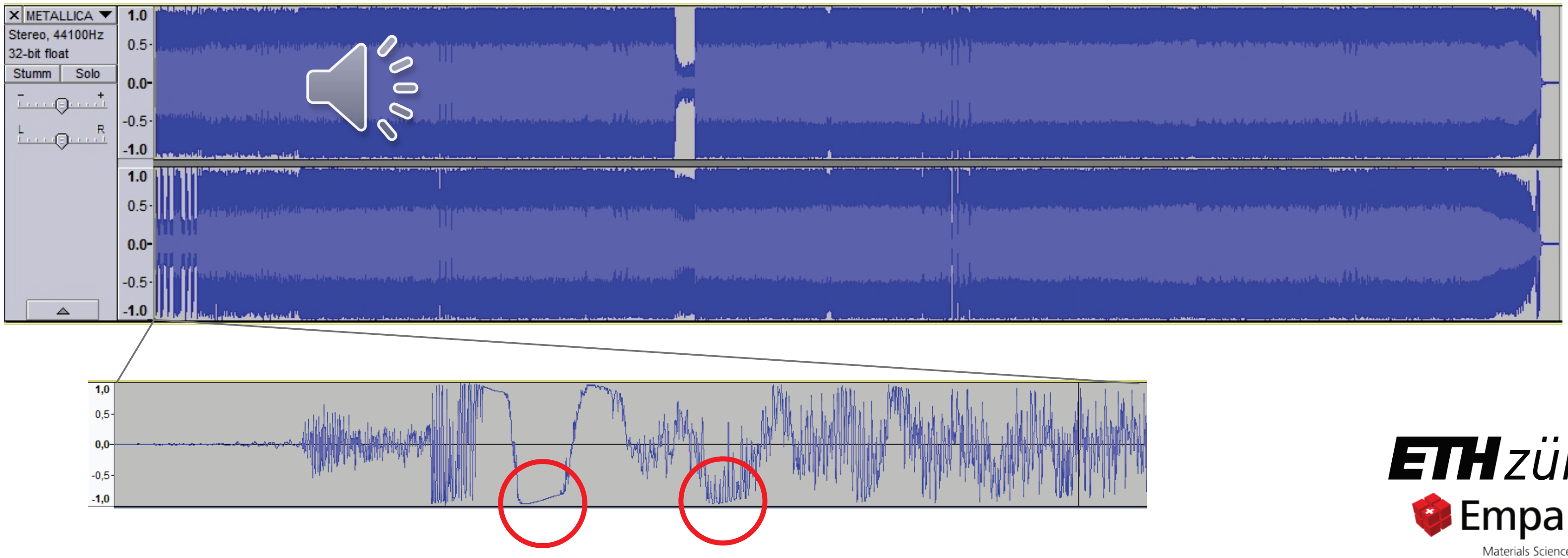
Compressor/limiter: Finalizer

- *Loudness war*
  - producers push perceived loudness to maximum (radio, TV, music)
  - increasing concern regarding audio quality
  - loss in dynamic range
  - unofficial dynamic range data base:

    http://www.dr.loudness-war.info/

Compressor/limiter: Finalizer

- Loudness war example:
  - CD version of *Metallica: Death Magnetic: My Acopalypse*, 2008



Compressor/limiter: Loudness normalization

- End of the loudness war? → Loudness normalization
  - Evaluate loudness level of each track → adjust playback gain for each track to unified loudness level
  - Applied in today's online portals, e.g. Spotify or YouTube
  - Loudness L_{KG} measured in LKFS/LUFS (Loudness Units relative to Full Scale) (equal to dB)

[ITU 2015. Recommendation ITU-R BS.1770-4 - Algorithms to measure audio programme loudness and true-peak audio level, International Telecommunication Union (ITU), Geneva, Switzerland.]

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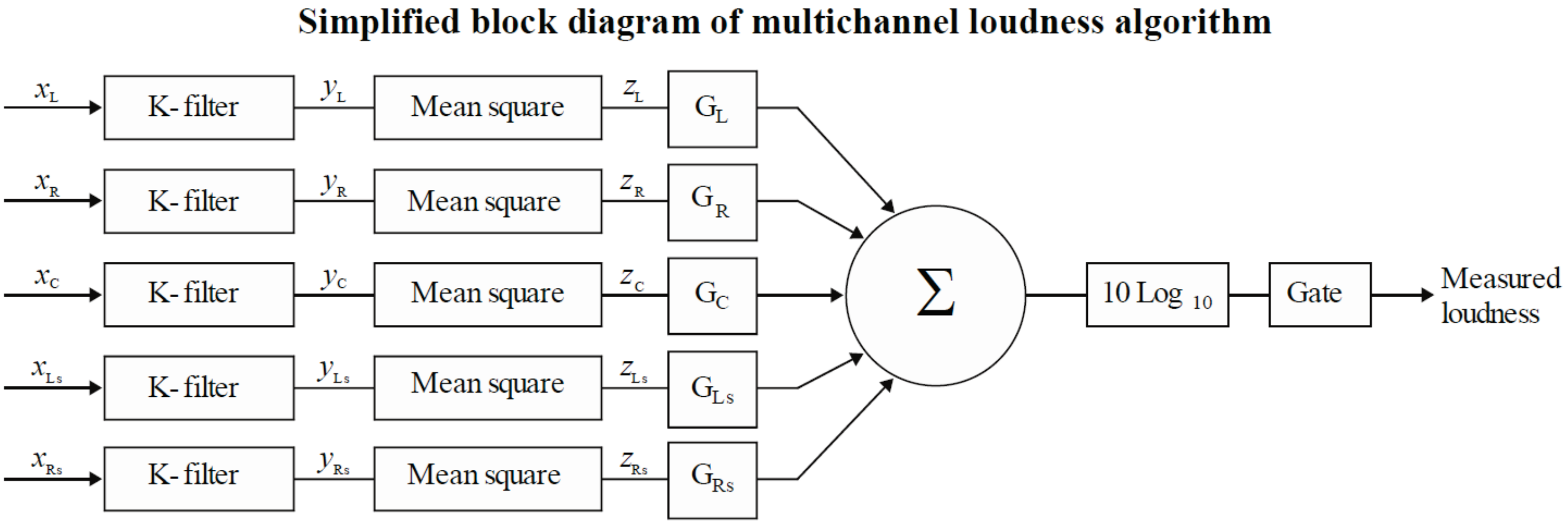
Compressor/limiter: Loudness normalization



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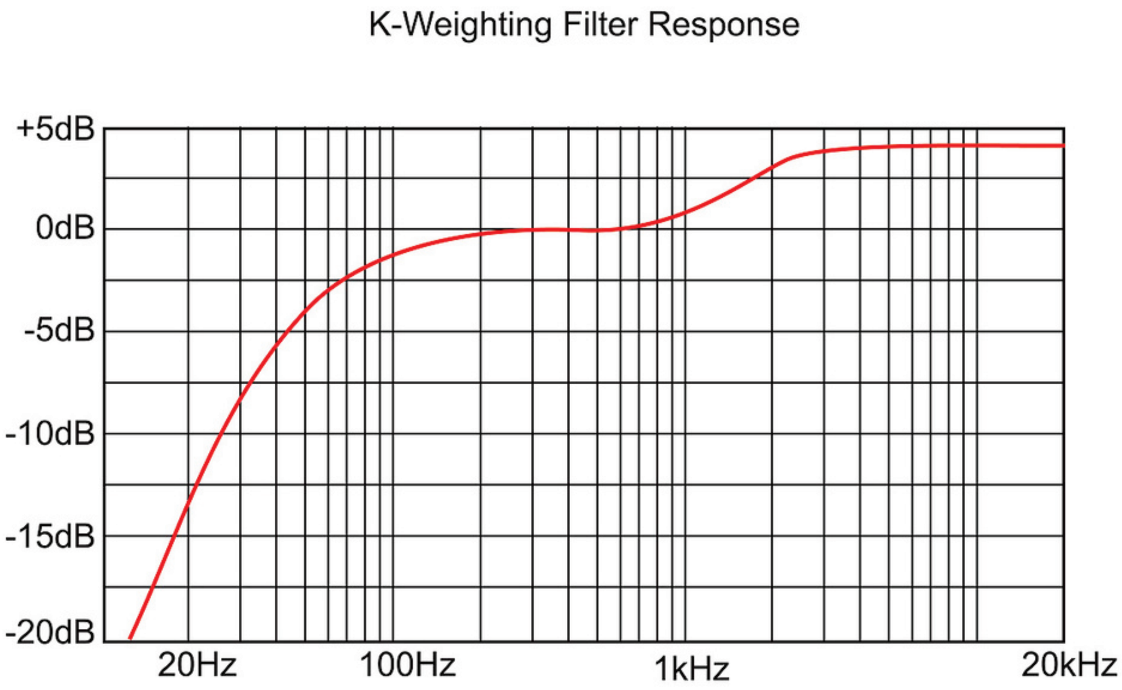
Compressor/limiter: Loudness normalization



Noise gate

Noise gate

- Function
  - Attenuation of low level signals
- Application
  - Suppression of noise during signal pauses

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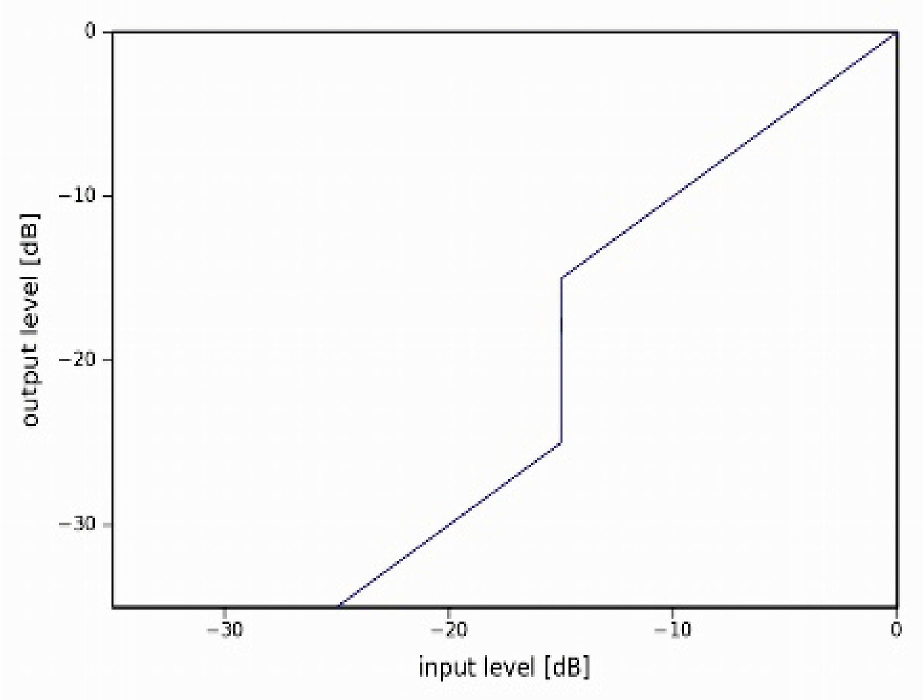
Noise gate

- Parameters: Level threshold + step



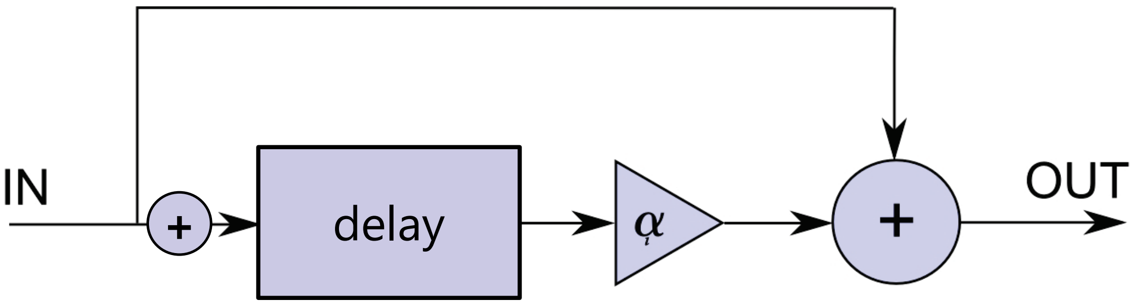
Advanced dynamics and EQ effects

- Various more advanced (combined) effects
  - Side-chain compressor
  - Multiband compressor
  - Dynamic EQ
  - Exciter
  - Wah-wah
  - …

Delays

Echo

- Applications:
  - Repetition of sound event
  - Simulation of an early reflection added to direct sound
  - Broadening and softening a sound



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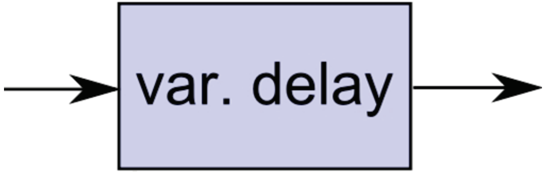
Vibrato

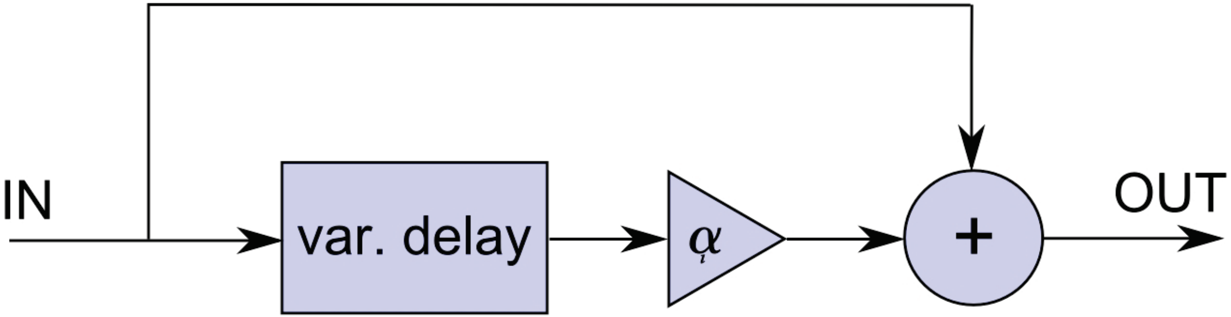
- Typical characteristic in singing voice, violin or guitar
- Time-variable pitch
  - Modulation frequency of a few Hz
  - Modulation depth: below semitone = below 6%
- Implementation by variable delay steered by oscillator, typically sinusoid
- Time-dependence → no LTI system



Flanger

Flanger: Principle

- summation of signal and a delayed copy
- slow variation (0.1 Hz) of the delay (couple of milliseconds)



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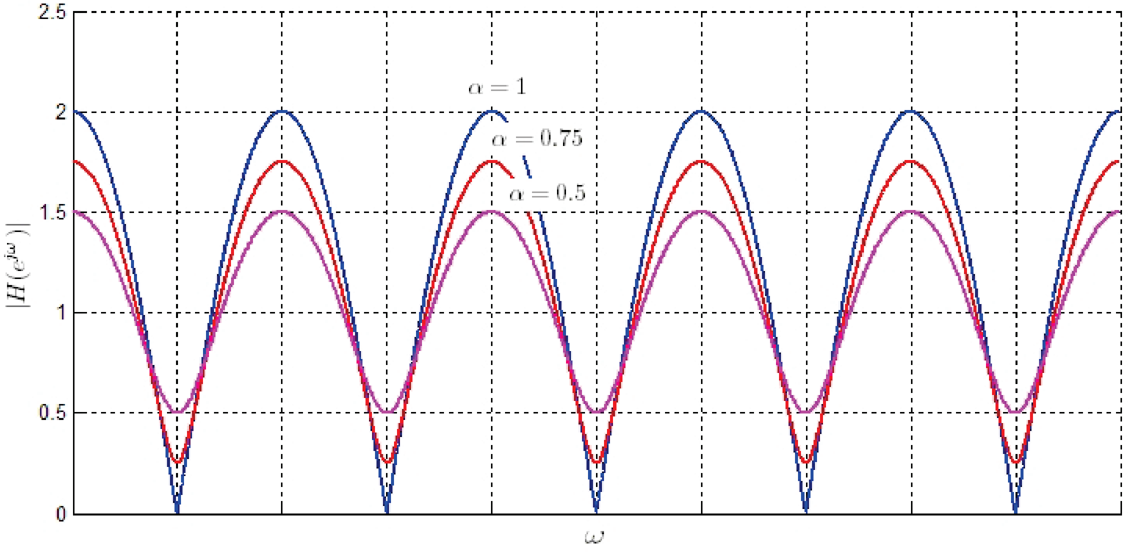
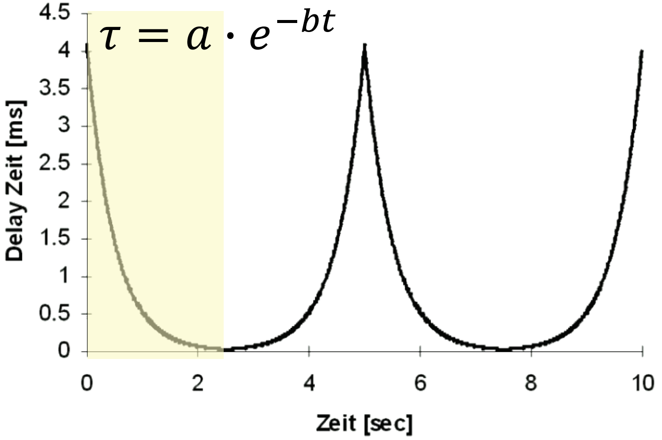
Flanger: Comb filter

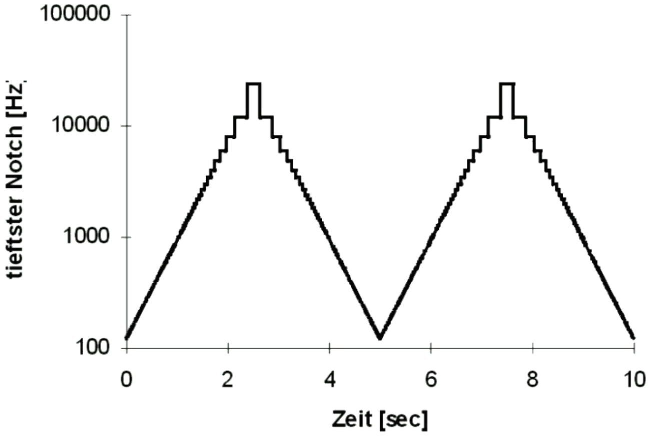
- Comb filter for delay τ:
  - Maxima: $f = 0, \frac{1}{\tau}, \frac{2}{\tau}, \frac{3}{\tau}$ Minima: $f = \frac{0.5}{\tau}, \frac{1.5}{\tau}, \frac{2.5}{\tau}$



Flanger: Comb filter

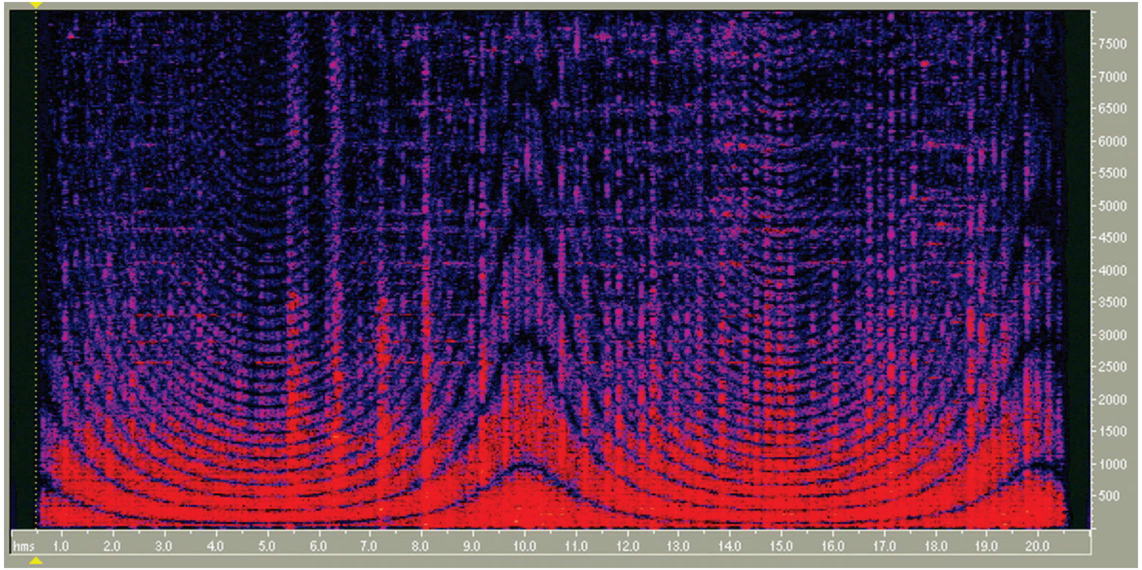
- time dependency of the delay variation function
  - Different variants
  - e.g. sinusoidal: uneven perceived variation of the maxima/minima frequencies due to logarithmic frequency perception
  - in general → non-sinusoidal time function





Flanger: Example

- Original sample:
- Flanger with delay: 0.5-5ms, variation: sinusoidal with 0.1Hz:



Chorus

- Application: Virtual multiplication of an instrument or voice
- Parallel arrangement of independent flangers

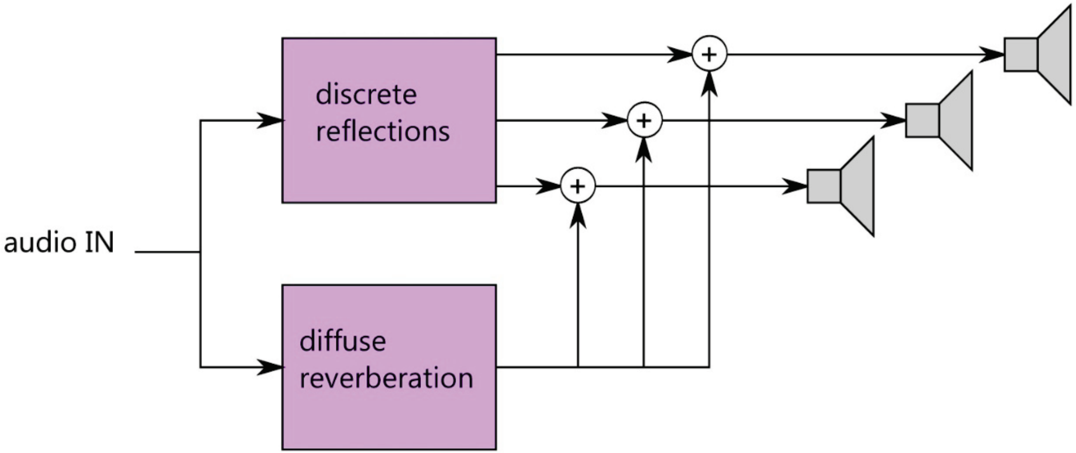
- Original:
- With chorus:

Phaser

- Similar to flanger but summation of signal and *phase-shifted* copy
- Slow variation of the phase shift
- Result = Spectral - non-harmonic - amplifications and attenuations
- Realization by cascade of time-varying all-pass filters

Reverberation

Reverberation: Room simulation



Reverberation: Realization of diffuse reverberation

- Analog
  - Spring reverb: mechnical springs
  - Plate reverb: vibration of metal plate
  - Echo chambers: reverberant acoustical space → KKL
- Digital
  - Feed-back structures (IIR)
  - Convolution with impulse responses (FIR)

Artificial reverberation

- Many reverb algorithms
- Original work by M. Schroeder

[Schroeder, M.R. 1962. Natural sounding artificial reverberation. *Journal of the Audio Engineering Society*, 10(3).]

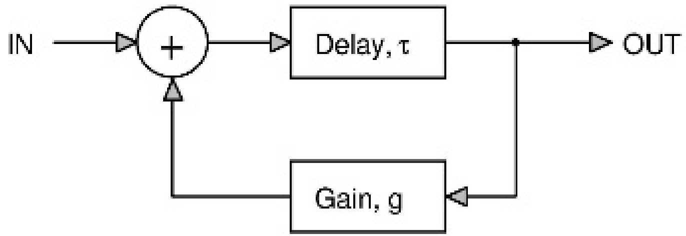
[Välimäki, Vesa et al. 2012. Fifty Year of Artificial Reverberation. *IEEE Transactions on Audio, Speech and Language Processing 20(5)*.

Smith III, Julius O. 2010. Physical Audio Signal Processing: for Virtual Musical Instruments and Digital Audio Effects. W3K Publishing.]

https://valhalladsp.com/2009/05/30/schroeder-reverbs-the-forgotten-algorithm/
https://ccrma.stanford.edu/~jos/pasp/Artificial_Reverberation.html

Artificial reverberator element

- Simplest feedback structure



- Condition for stability: $g < 1$
- Physical concept: Plane wave travelling between two parallel walls with slight absorption → flutter echo

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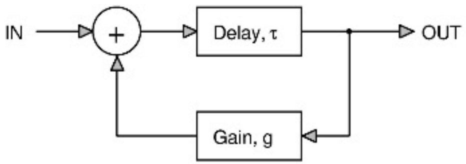
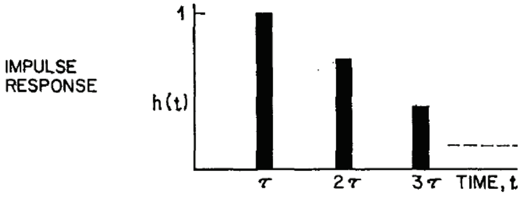
Artificial reverberator element





- Impulse response:

$$h(t) = \delta(t-\tau) + g\delta(t-2\tau) + g^2\delta(t-3\tau) + ..$$

- Transfer function:

$$H(\omega) = e^{-j\omega\tau} + ge^{-j\omega 2\tau} + g^2 e^{-j\omega 3\tau} + ..$$

Artificial reverberator element

$$H(\omega) = e^{-j\omega\tau} + ge^{-j\omega 2\tau} + g^2 e^{-j\omega 3\tau} + ..$$

is a geometric series →

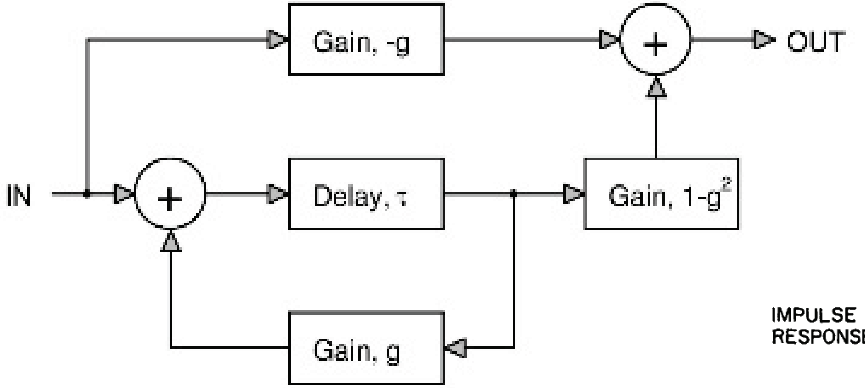
$$H(\omega) = \frac{e^{-j\omega\tau}}{1 - ge^{-j\omega\tau}}$$

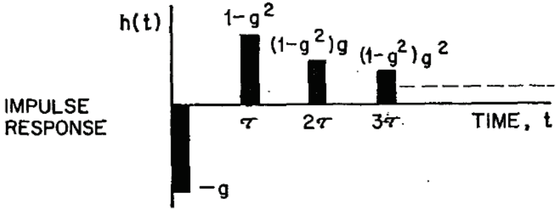
with magnitude response:

$$|H(\omega)| = \frac{1}{\sqrt{1 + g^2 - 2g\cos\omega\tau}}$$

→ comb filter function! «feedback comb filter»

Artificial reverberator element: All-pass





- Feedback + feed-forward comb filter = All-pass
- Flat amplitude response
- Reverberation time: $RT = \dfrac{60\tau}{-20\log(g)}$
- Issue: For g→1 direct sound path dominates
- Use of multiple elements → Schroeder

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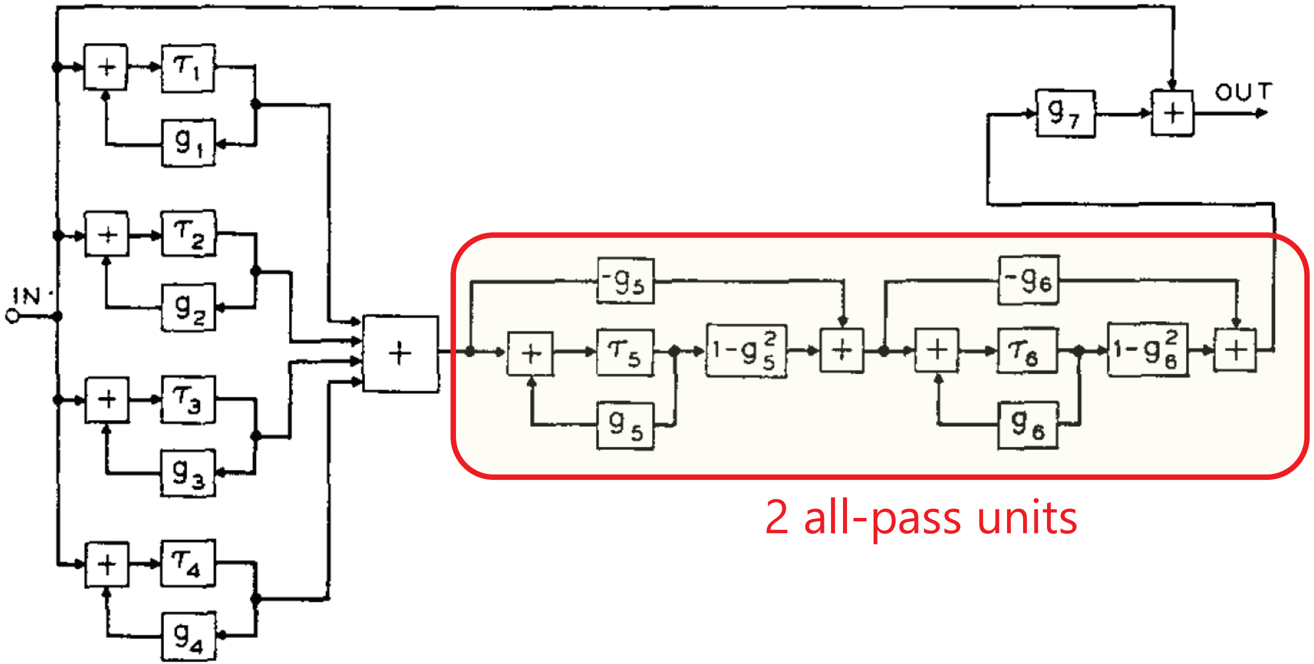
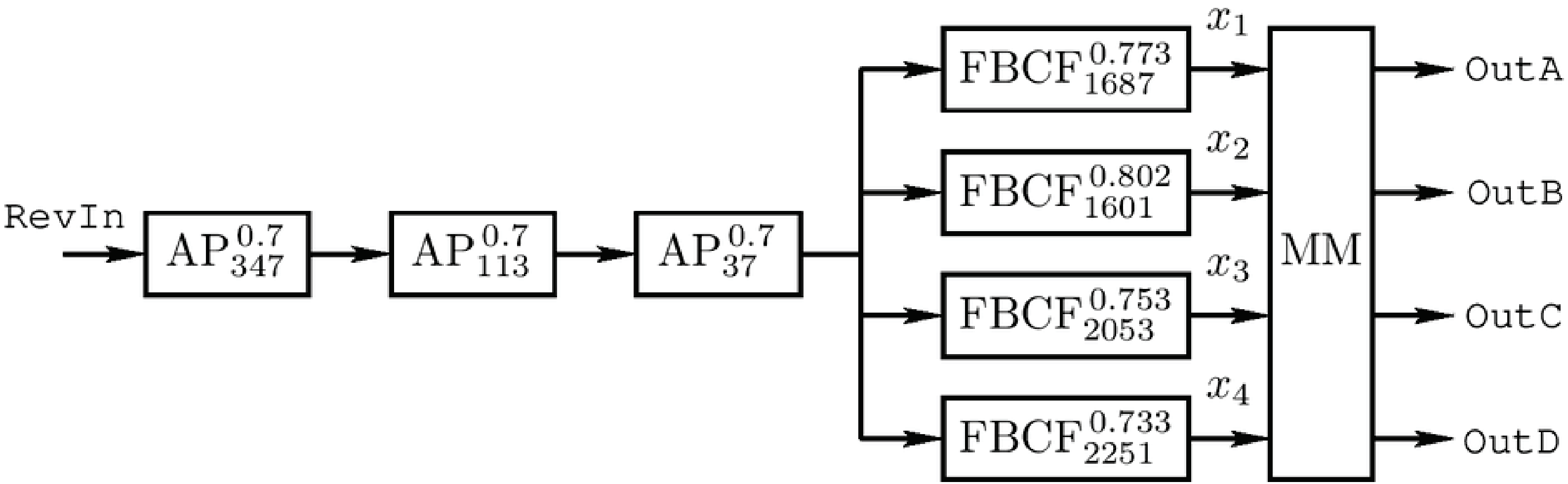
Schroeder reverberator 1962



2 all-pass units

4 feedback comb filter units

Schroeder type reverberators



- FBCF: Feedback comb filter
- AP: All-pass = feedback + feedforward comb filter
- MM: Matrix mixing

Convolutional Reverb

- Reverberation is linear effect
- Simulation by linear convolution with room impulse response (RIR)
- RIR:
  - Truncation to get finite length
  - may be quite long: up to 15 seconds
  - much shorter than the input signal
- Digital implementation: FIR filter
- Efficient frequency domain algorithm using FFT/iFFT and overlap-add method

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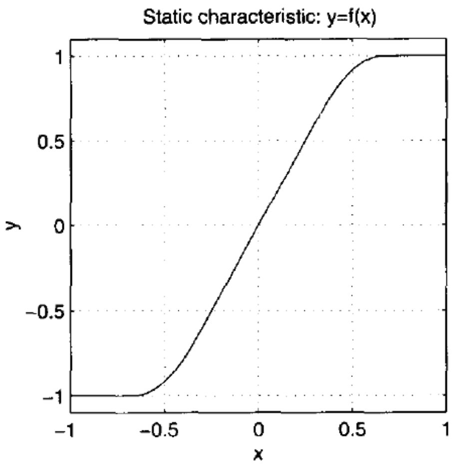
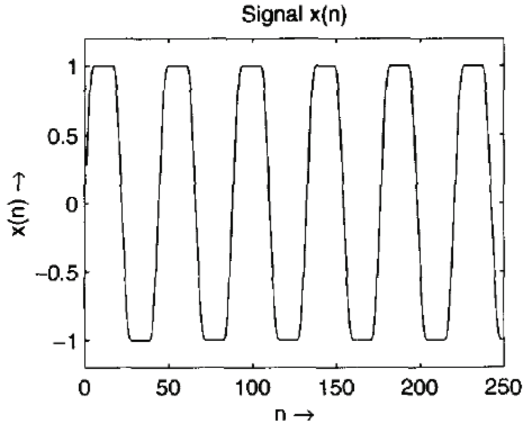
Other effects

Overdrive, distortion



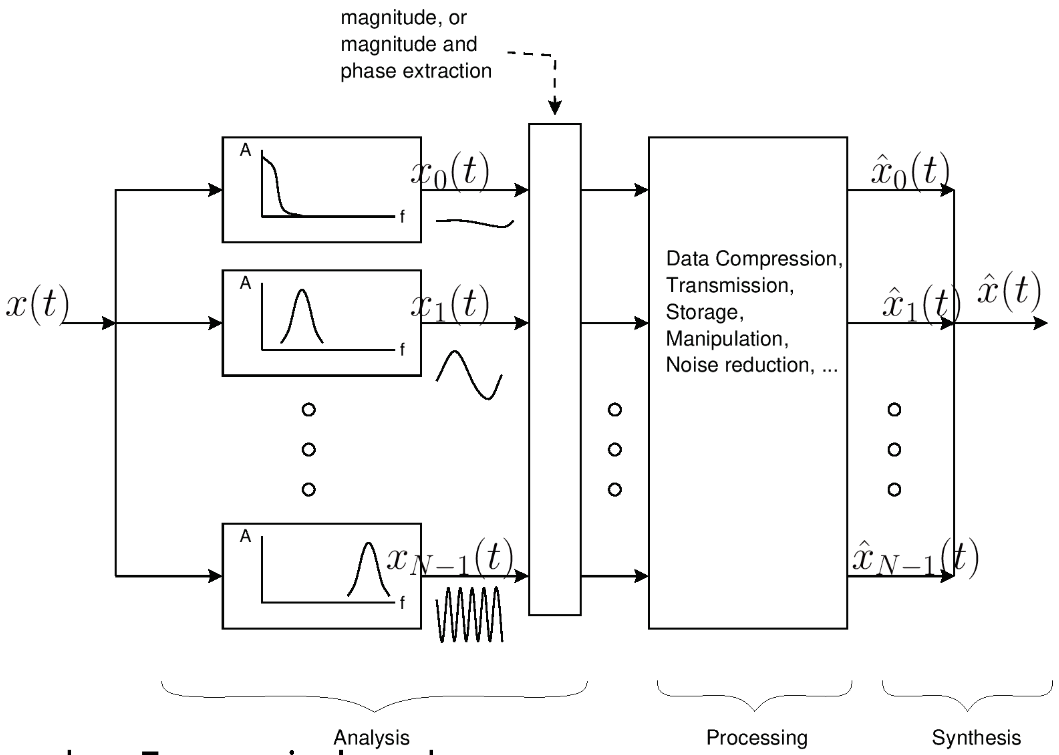


- Symmetrical clipping: creation of odd harmonics only
- Asymmetrical clipping: creation of even and odd harmonic

[Zölzer, Udo. et al. 2002. DAFX – Digital Audio Effects. John Wiley & Sons.]

Vocoder

Vocoder = Voice coder: Principle



Vocoder Example:
Daft Punk – Get Lucky

- Channel vocoder: Energy in bands
- Phase vocoder: Considers phase through STFT

Pitch-shift and time scaling

Pitch-shift and time scaling

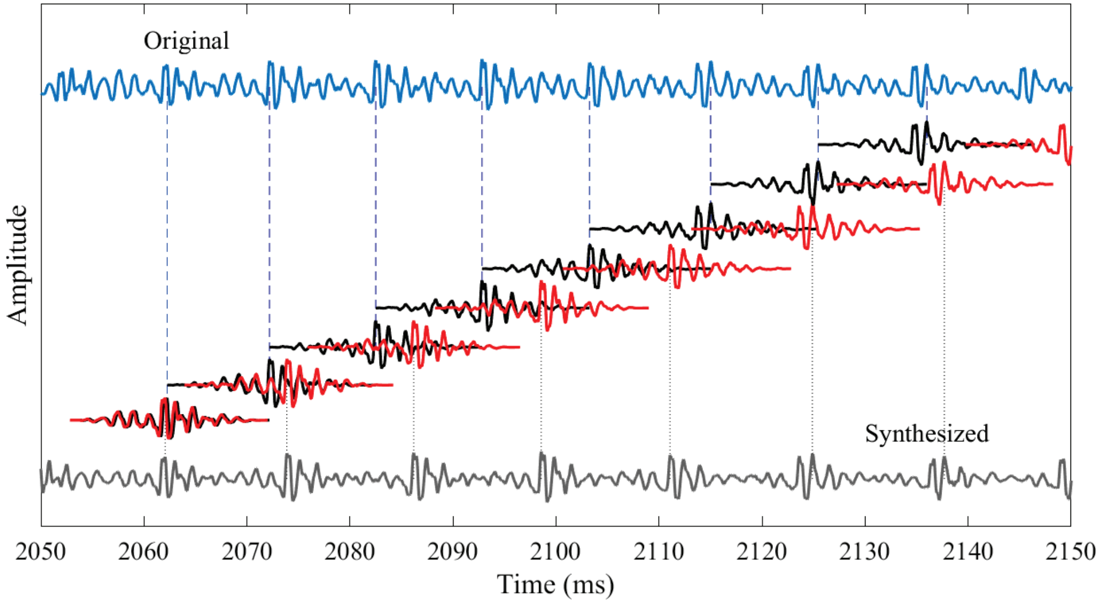
- Goal
  - Pitch variation without scaling time axis
  - Time axis scaling without pitch shift
- Examples:

	Original	Stretching factor 2
Violin		
Drums		

from Průša 2017, https://ltfat.github.io/notes/050/

Pitch-shift and time scaling : PSOLA

- Time-domain approach: Pitch-Synchronous OverLap and Add
- Processing: Pitch estimation, segmentation, windowing, assembling



[https://wiki.aalto.fi/display/ITSP/Introduction+to+Speech+Processing]

Pitch-shift and time scaling : Phase vocoder

- Frequency-domain method using the Short-Time Fourier Transform (STFT)
- Usual problem: Artefacts like phasiness or loss of presence
- Many algorithms for consideration of phase coherence across time and frequency → phase locking methods

[Laroche, J. & Dolson, M. 1999. Improved Phase Vocoder Time-Scale Modification of Audio. IEEE Transactions on Speech and Audio Processing, 7(3).]