Tampere University

Time-domain processing

SGN 14007

Lecture 4

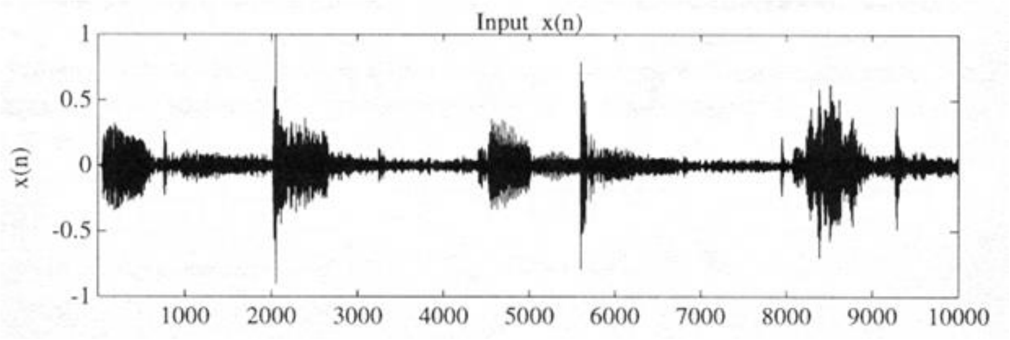
Annamaria Mesaros

Dynamic range control

Tampere University

What is dynamic range control?

- Sound pressure level of natural sounds varies over time
  - Movies, music: The loudness of sounds works as an effect and is part of the content
  - When recording, the distance to microphone may vary, which causes unwanted level variation
- Dynamic range:
  - Ratio of maximum to minimum signal levels over a period of time
  - Expressed in decibels, audio typically 40 dB – 120 dB
- Dynamic range control
  - Combination of level measurement and adaptive level adjustment

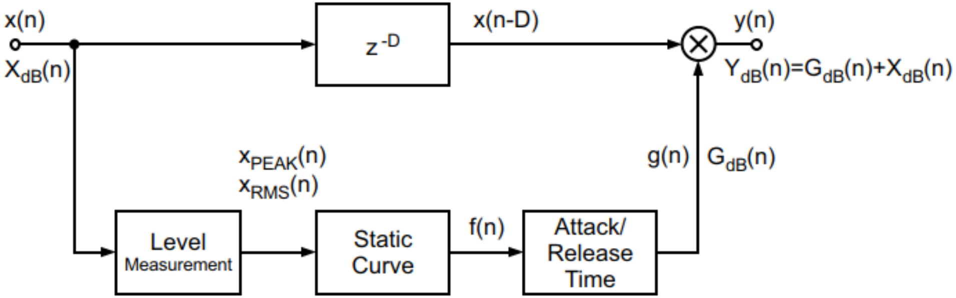


What is DRC useful for?

- When recording, it is desirable to:
    - Use the full amplitude range optimally
    - Also to protect AD converters from overloading (clipping)
- When playing back music or speech e.g. in a car, dynamic range variation has to be matched to background noise
    - Noise would not mask the quiet parts of the signal
    - Listening becomes easier / possible
    - Live music contains a dynamic range too wide for reproduction in the average home
- Suppress low-amplitude noise using noise gates
    - Only audio signals exceeding a certain level are passed through
- In various recording formats, the limited range of amplitudes should be optimally used

Tampere University

DRC block diagram



- Measure the level of input signal
- Multiply the delayed input signal by factor *g(n)*

$$y(n) = g(n) \cdot x(n - D)$$

- Delaying signal *x(n)* compared to the control signal *g(n)* allows ”predictive” level control (gain goes down before any ”big bangs”)
- Value of gain factor *g(n)* is calculated in two steps
  - *Static curve* defines desired output level corresponding to the input level
  - Temporal variation of *g(n)* is smoothed using a lowpass filter

Dynamic range compression

- Reduces the volume of loud sounds or amplifies quiet sounds

  → reducing/compressing dynamic range

- Downward compression – reduces loud sounds over a certain threshold; quiet sounds remain unaffected.

- Upward compression – increases the loudness of sounds below a certain threshold; leaves louder sounds unaffected.

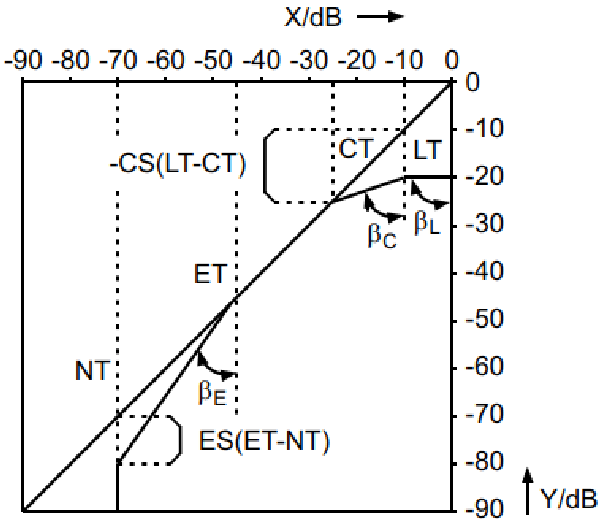
→ Both downward and upward compression reduce the dynamic range of an audio signal

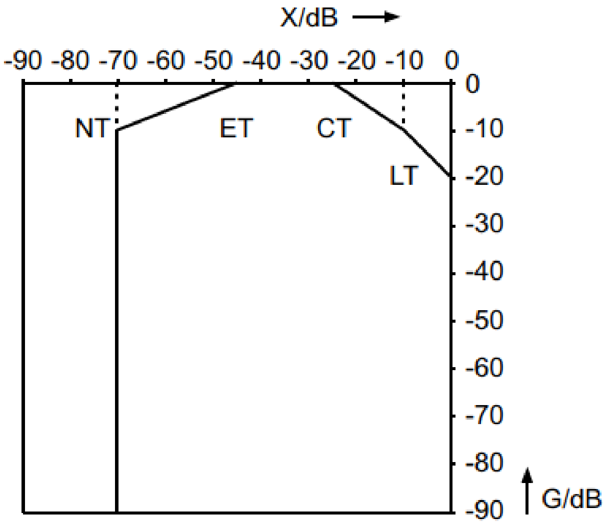
Tampere University

Static curve

- Defines the relationship between input level and weighting level G[dB] = f (X[dB])
  - Output level Y and weighting level G are functions of input level X
  - Thresholds: LT=limiter threshold, CT=compressor thr., ET=expander thr., NT=noise gate thr.

- Output level is Y [dB] = X [dB]+f (X [dB]) = X [dB]+G [dB]





Operation regions on the static curve

- **Limiter** limits the output level when the input level exceeds the limiter threshold LT
  - All input levels above the threshold lead to a constant output level
- **Compressor** maps a change of input level to a smaller change in the output level
- **Expander** increases changes in the input level to larger changes in the output level
- **Noise gate** is used to suppress low-level signals and to reduce noise
- Threshold values in different parts of the static curve determine the lower limit for limiter and compressor and upper limit for expander and noise gate

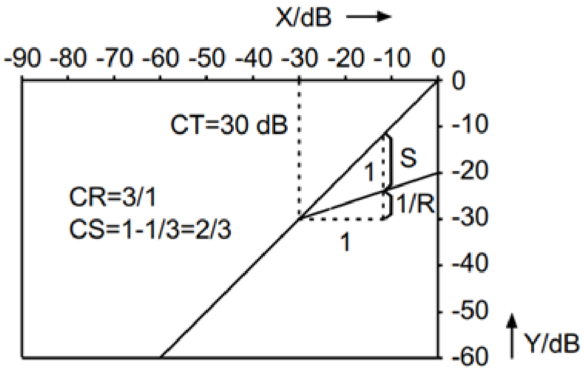
Compression ratio

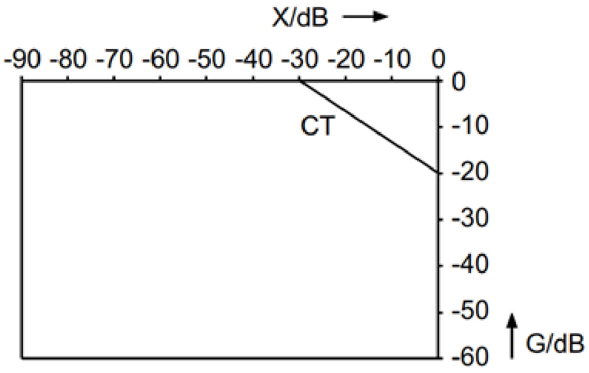
Compression ratio is visible in the logarithmic representation of the static curve: **ratio of input level change** ΔP_I **to output level change** ΔP_O

$$R = \Delta P_I / \Delta P_O$$

Typical compression ratios:

- R = (limiter), R > 1 (compressor), 0 < R < 1 (expander), R = 0 (noise gate at threshold)





For those who want more details, not important for exam

Compression ratio

- From figure: line equation $Y = CT + (1/R)\cdot(X - CT)$ and compression ratio $R = (X - CT)/(Y - CT)$
  - CT is input threshold (dB), where compression starts
- Switching from logarithmic to linear representation we obtain

$$R = \frac{\log_{10}(x/c_T)}{\log_{10}(y/c_T)}$$

  where *x* and *y* are linear levels, and c_T is linear compression threshold
- From here we can solve linear output level *y* as a function of the input level *x*

$$y/c_T = 10^{(1/R)\cdot\log_{10}(x/c_T)} = (x/c_T)^{1/R}$$

$$y = c_T^{\,1-1/R}\cdot x^{1/R}$$

- The control factor *g*(*n*) can be calculated as

$$g(n) = y/x = (x/c_T)^{1/R-1}$$

Thinking break (2 min)

Calculate compression ratios for expander and compressor segments on the figure from slide 7

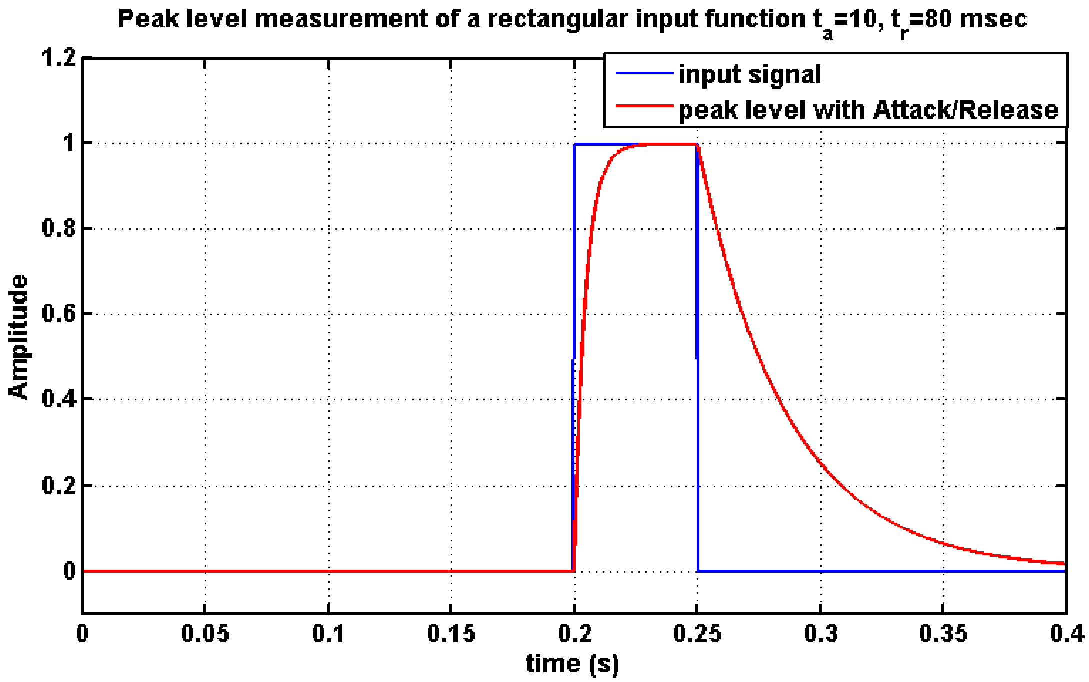
Dynamic behavior

- Besides static curve based level control, the **dynamic behaviour** (time-varying behaviour) of the control factor (attack and release time) plays a significant role in sound quality
- The rapidity of input signal level measurement also affects the rapidity of the overall dynamic range control

- There are two widely-used methods for **level measurement**: PEAK and RMS
- Functionally interchangeable (same task)
- However PEAK is typically used in the limiter, whereas RMS is used in compressor, expander and noise gate

Peak measurement example

For a rectangular pulse



PEAK algorithm

For those who want more details, not important for exam

PEAK algorithm (produces a level measurement $x_{peak}(n)$):

- Coefficient AT determines the attack time, coefficient RT the release time

if $|x(n)| > x_{\text{peak}}(n-1)$

$x_{\text{peak}}(n) = (1- AT)\, x_{\text{peak}}(n-1) + AT \cdot |x(n)|$

else

$x_{\text{peak}}(n) = (1 - RT) \cdot x_{\text{peak}}(n - 1)$

end

$AT = 1 - exp(-2.2\, T_s / (t_a/1000))$, where t_a=attack time (ms), $T_s=1/F_s$ is samp. interval

$RT = 1 - exp(-2.2\, T_s / (t_r/1000))$, where t_r = release time (ms)

Typically t_a = 0.02ms…10ms and t_r = 1ms…5000ms

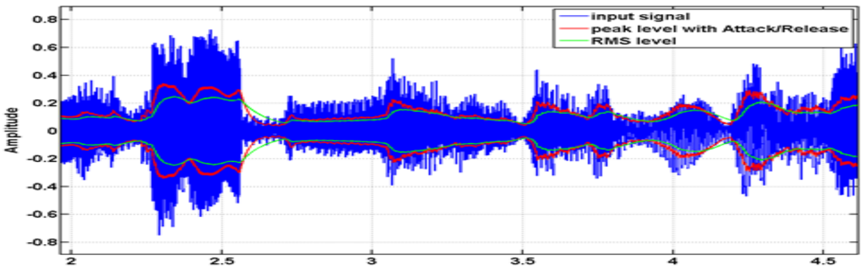
RMS algorithm

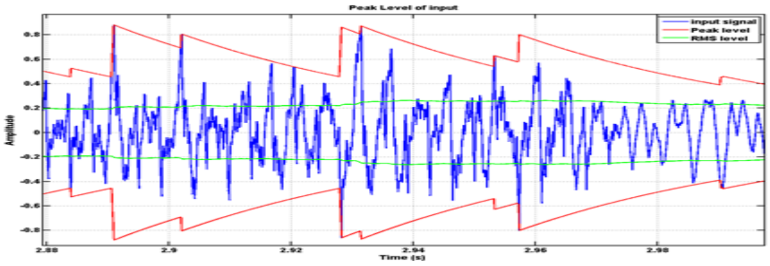
For those who want more details, not important for exam

- *RMS algorithm* (produces level measurement $x_{rms}(n)$):
  - Square of the input, averaging with first-order lowpass filter
  - Temporal length of the averaging is determined by coefficient *TAV*

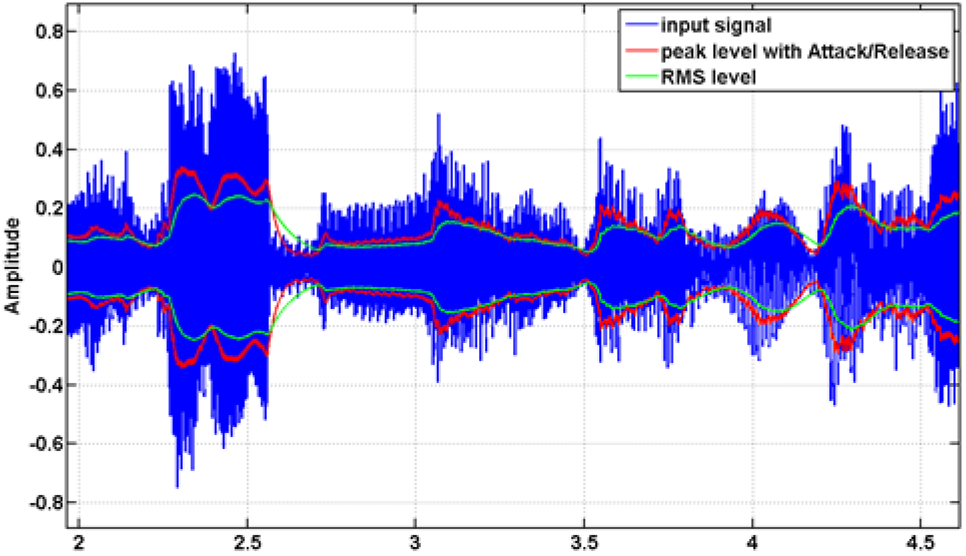
$$x^2_{rms}(n) = (1 - TAV) \cdot x^2_{rms}(n-1) + TAV \cdot [x(n)]^2$$

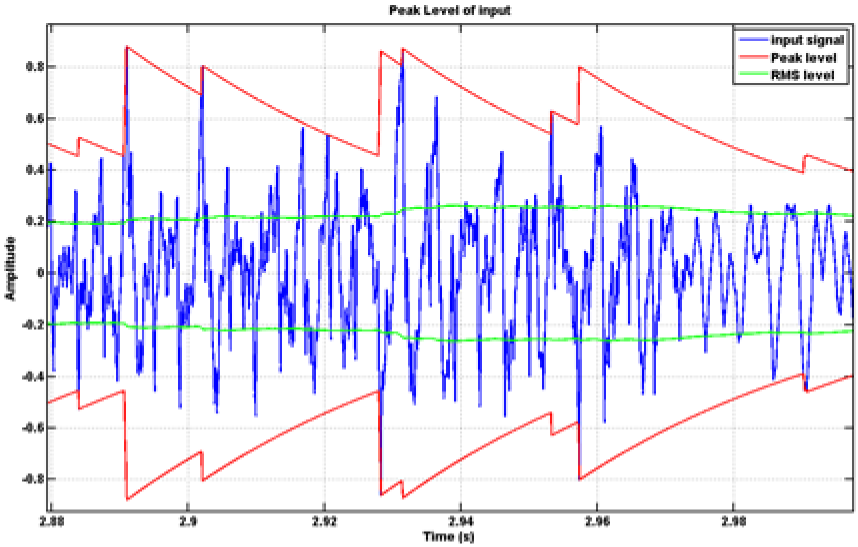
  - $TAV = 1\text{-}exp(-2.2\ Ts / (t_M / 1000))$
    - t_M is the averaging time in milliseconds.
    - In the figure t_M=*100 ms* (green line), Ts=sampling interval
    - For peak (red line): on the left, t_a=10ms, on the right t_a=0.02ms





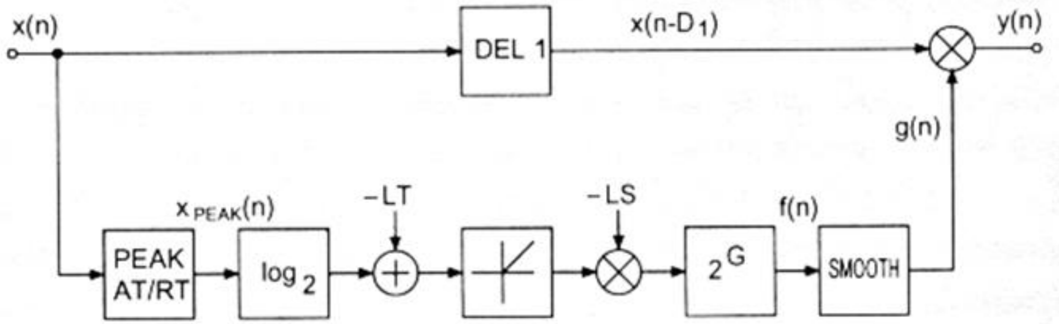
Peak and RMS measurement examples



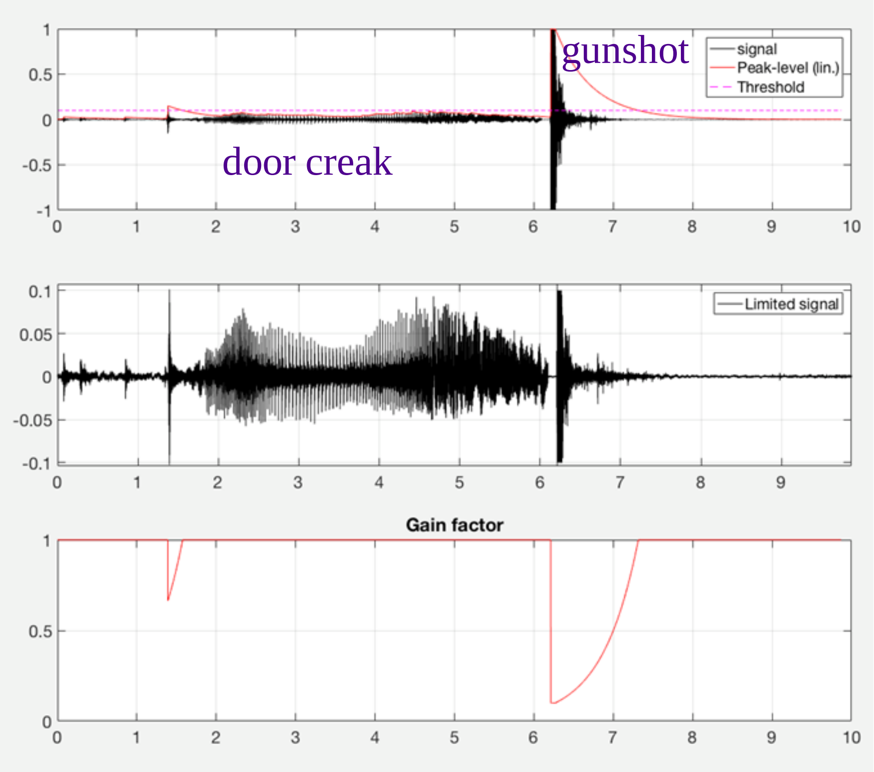


Example of a limiter

1. Level $x_{peak}(n)$ of input signal $x(n)$ is measured using the PEAK algorithm
2. Value $\log_2[x_{peak}(n)]$ is compared to limiter threshold LT
3. If the value is above the threshold (difference is positive)
    a. Difference is multiplied by the negative slope –LS of the limiter
    b. Take antilogarithm 2^G
    c. The resulting control factor $f(n)$ is smoothed using first-order lowpass filter SMOOTH
4. If the value is not above the threshold, factor $f(n)$ is set to value 1.
5. Delayed input signal $x(n – D_1)$ is multiplied with the smoothed control factor $g(n)$ to give the output $y(n)$

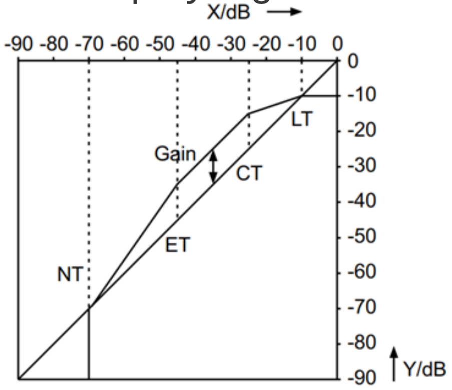


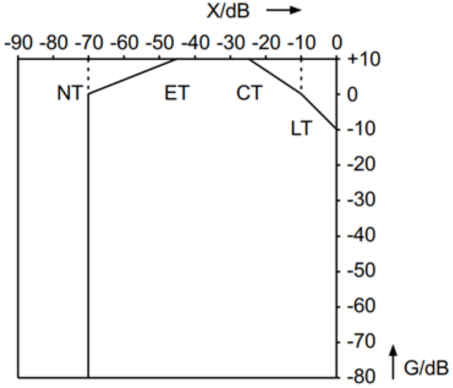
Example of a limiter use



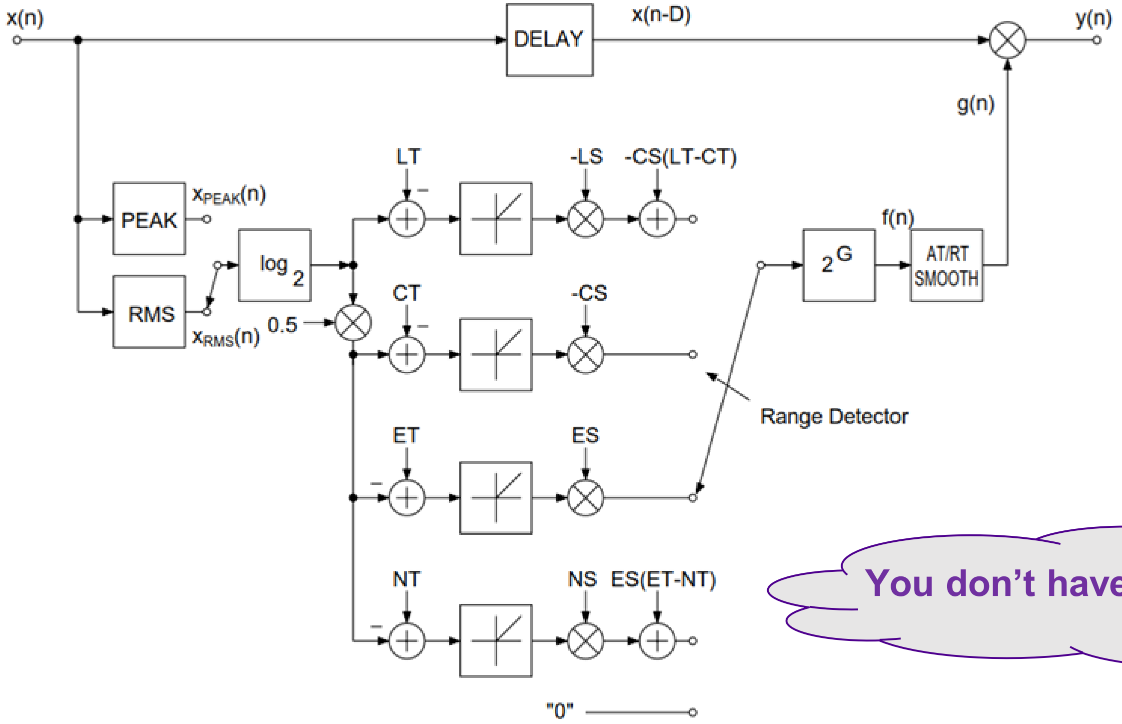
Combination system

- The basic structure of compressor, expander, and noise gate is similar to the limiter
- Practical implementation is a cascade, where each part implements one of the basic operations (limiter / compressor / expander)
  - Appropriate parameters can be chosen for each stage
  - Limiter using PEAK measurement, compressor/expander/noise gate using RMS
  - If LT threshold is crossed, use limiter to calculate characteristic curve
  - If LT is not crossed, log of RMS is used with one of three possible paths, depending on thresholds
- By limiting the maximum level, the dynamic range is reduced → the overall static curve can be shifted up by a gain factor





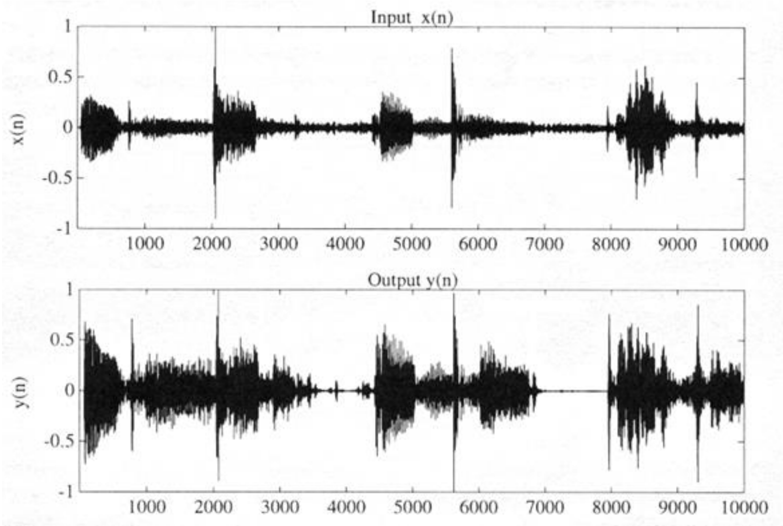
Combination system



You don’t have to know this diagram!

Combination system

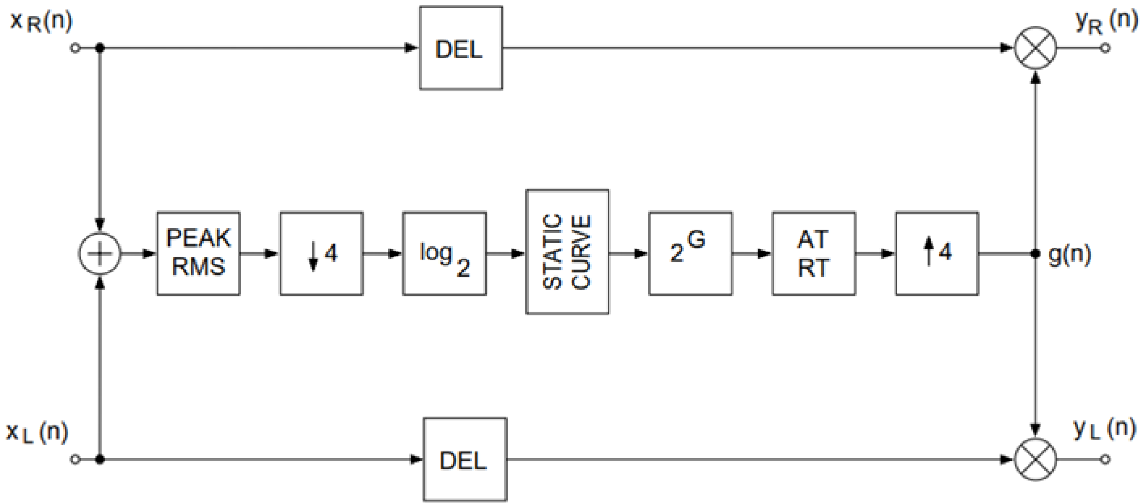
Signals with high amplitude are compressed, ones with low amplitude are expanded



Tampere University

Stereo signals

- For stereo signals, a common control factor g(n) is needed
  - That way the stereo balance is not displaced
  - Input signal level is measured from channel average
- Figure: dynamic range control of a stereo signal



Use of dynamic range control

- Music – make music sound louder without increasing its peak amplitude → music in public places
- Broadcasting – TV commercials heavily compressed to achieve near-maximum perceived loudness while staying within permissible limits (peak loudness by law)
- Voice – De-essing: dynamic range control at sibilants (/s/ phoneme) frequency range (about 5 kHz)
- Hearing aids – use a compressor to bring the audio volume into the listener's hearing range
- “Active sound protection" earmuffs/earplugs - let sounds at ordinary volumes be heard normally while attenuating louder sounds; possibly amplify softer sounds

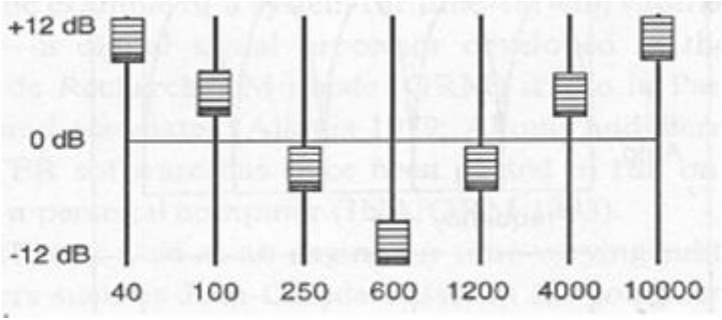
Equalizers

Equalization

- **Equalization** – adjusting the balance between frequency components within a signal

- **Equalizer** – the circuit or equipment used to achieve it
  - Strengthen (boost) or weaken (cut) the energy of specific frequency bands; “frequency-specific volume knobs”
  - Found both in consumer products as well as in professional use
  - Consumer products (e.g. car radio or amplifier) often employ simple bass and treble control
  - Recording studios and professional audio often use more sophisticated equalizers capable of detailed adjustments
- Correcting/adjusting frequency response

Equalizer



- Broad definition: all linear filters at the disposal of a listener or sound engineer

Digital linear filters

Remember digital linear filters:

$$y[n] = \sum_{k=0}^{L} x[n-k]b_k - \sum_{k=1}^{M} y[n-k]a_k$$

$$y[n] = x[n]b_0 + x[n-1]b_1 + \ldots - y[n-1]a_1 - y[n-2]a_2 \ldots$$

- y[n] is the (current) output
- x[n] is the current input at time n
- b_k, a_k are coefficients that control the frequency response:
  - b_k controls input
  - a_k controls feedback

Digital linear filters

- The response of the filter is H(z)

$$H(z) = \frac{Y(z)}{X(z)} = \frac{b_0 + b_1 z^{-1} + .. + b_L z^{-L}}{1 + a_1 z^{-1} + \cdots + a_M z^{-M}}$$

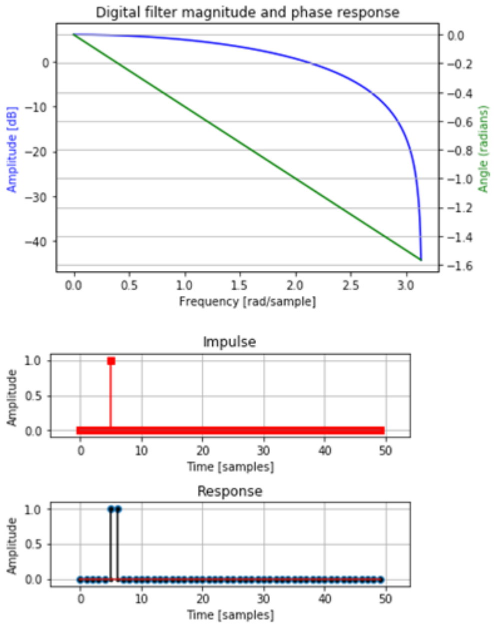
- The Z-transform of x[n] is

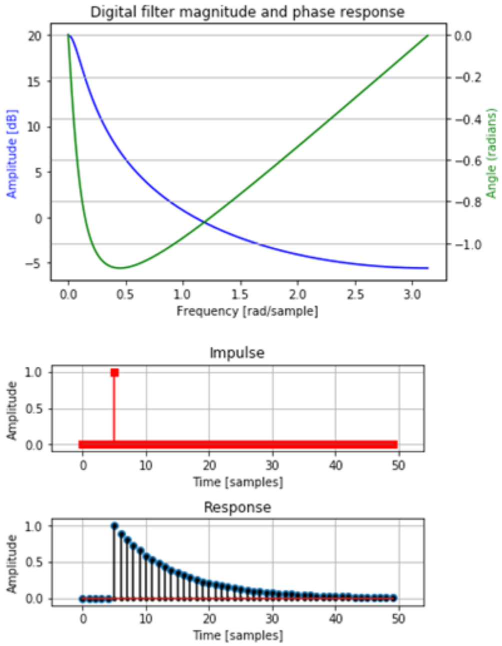
$$X(z) = \sum_{n=-\infty}^{\infty} x[n] z^{-n}$$

- If a_k=0 for all k, the filter is finite impulse response - FIR, nonrecursive
- Otherwise the filter is infinite impulse response - IIR, recursive

Digital linear filters

Example: 1st order lowpass FIR (left) and IIR (right) filters

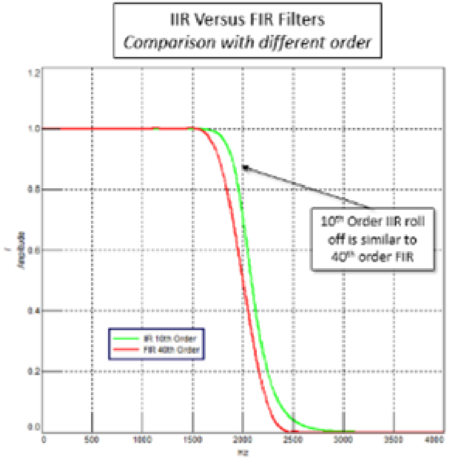




IIR or FIR for audio processing?

Computational efficiency:

IIR filters are faster from a computational point of view than FIR filters: a narrow transition band is achieved using a small number of filter coefficients (multiplication operations)



Images from https://community.plm.automation.siemens.com/t5/Testing-Knowledge-Base/Introduction-to-Filters-FIR-versus-IIR/ta-p/520959

IIR or FIR for audio processing?

Delay: Filtered signal is a delayed version of the original at the passband

- FIR filter has equal time delay at all frequencies, while the IIR filter time delay varies with frequency.
    - Usually the biggest time delay in the IIR filter is at the cut off frequency of the filter.
    - In the case of audio that is not always important, since the human ear is not very sensitive to frequency component phases
    - Magnitude response is perceptually more important

  → supports choosing IIR filters

    - However, phases at low frequencies affect the stereo image

IIR or FIR for audio processing?

Stability:

- an IIR filter can sometimes be unstable; FIR filter formulation is always stable

- FIR filters allow more accurate control of the filter response:
  - When designing filterbanks, FIR filters enable so-called perfect reconstruction, meaning that analysis/synthesis filterbank does not distort the signal if no processing is done at the subbands (Simplifies the design of an audio codec, for example)

- IIR filters provide easier way of varying the filter response in real time (for example varying the cut-off frequency)
  - Parametric filter structures

IIR or FIR for audio processing?

	IIR	FIR
Computational speed	Fast / low order	Slow / high order
Phase / Delay	Not constant	Constant
Stability	Sometimes	Always

→ Choice FIR / IIR depends on the application

Thinking break (2 min)

Shelving equalizers

- High and low pass filters – useful for **removing** unwanted signal above or below a set frequency

- **Shelving filters** – **reduce or increase** signals above or below a set frequency:
  - Tone controls (bass and treble) found in consumer audio equipment such as home stereos, and on guitar amplifiers
  - Most usually these implement a first order response and provide an adjustable boost or cut to frequencies above or lower than a certain point.

- **Idea**: Change the gain of some part of the spectrum while leaving other parts untouched
  - Obvious application to equalization: manipulating the system frequency response only at a given frequency range

First order shelving filter design

For those who want more details, not important for exam

- A low-pass shelving filter can be expressed as

$$H_{lp}(z) = C_{lp}\left(\frac{1-b_1 z^{-1}}{1-a_1 z^{-1}}\right)$$

$$C_{lp} = \frac{1+k\mu}{1+k}, b_1 = \frac{1-k\mu}{1+k\mu}, a_1 = \frac{1-k}{1+k}, k = \frac{4}{1+\mu}\tan\left(\frac{\Omega_c}{2}\right), \mu = 10^{G/20}$$

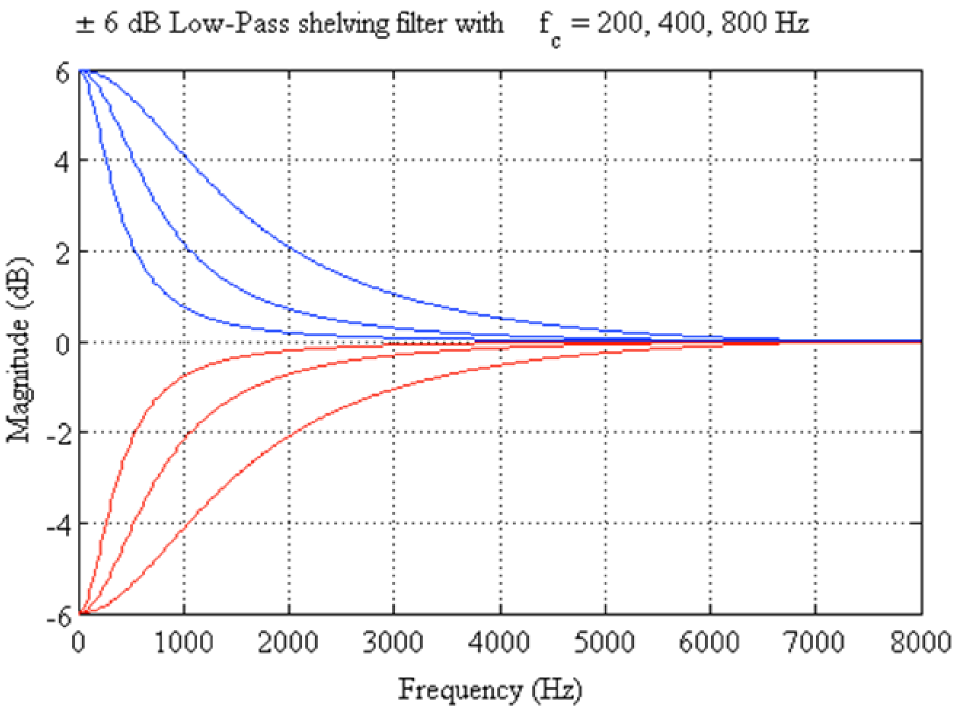
- A high-pass shelving filter

$$H_{hp}(z) = C_{hp}\left(\frac{1-b_1 z^{-1}}{1-a_1 z^{-1}}\right)$$

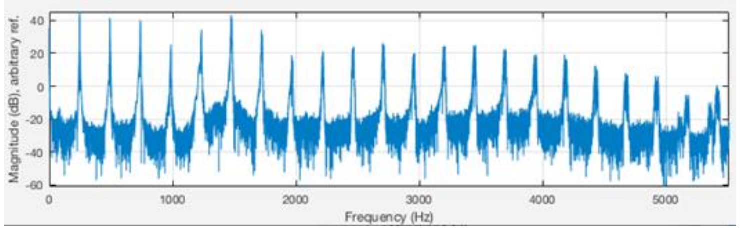
$$C_{hp} = \frac{\mu+p}{1+p}, b_1 = \frac{\mu-p}{\mu+p}, a_1 = \frac{1-p}{1+p}, p = \left(\frac{1+\mu}{4}\right)\tan\left(\frac{\Omega_c}{2}\right), \mu = 10^{G/20}$$

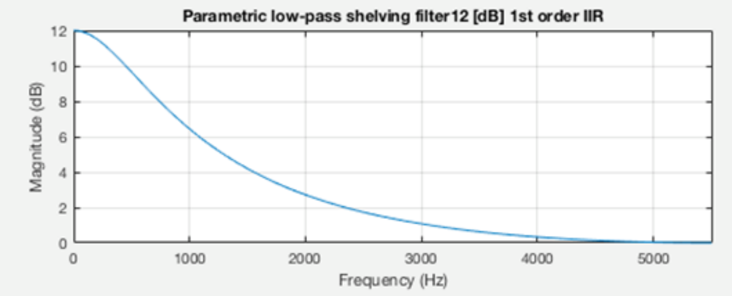
where G is gain (dB) and Ω_c is normalized cutoff frequency.

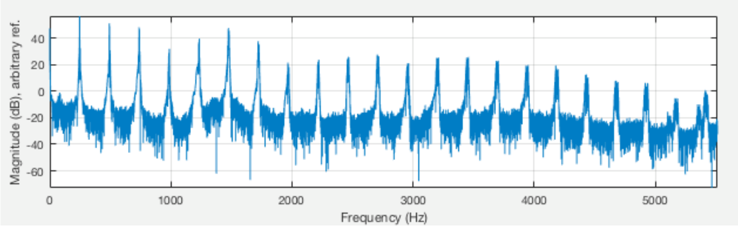
6 dB Low-Pass shelving filter responses



Example: low-pass shelving oboe



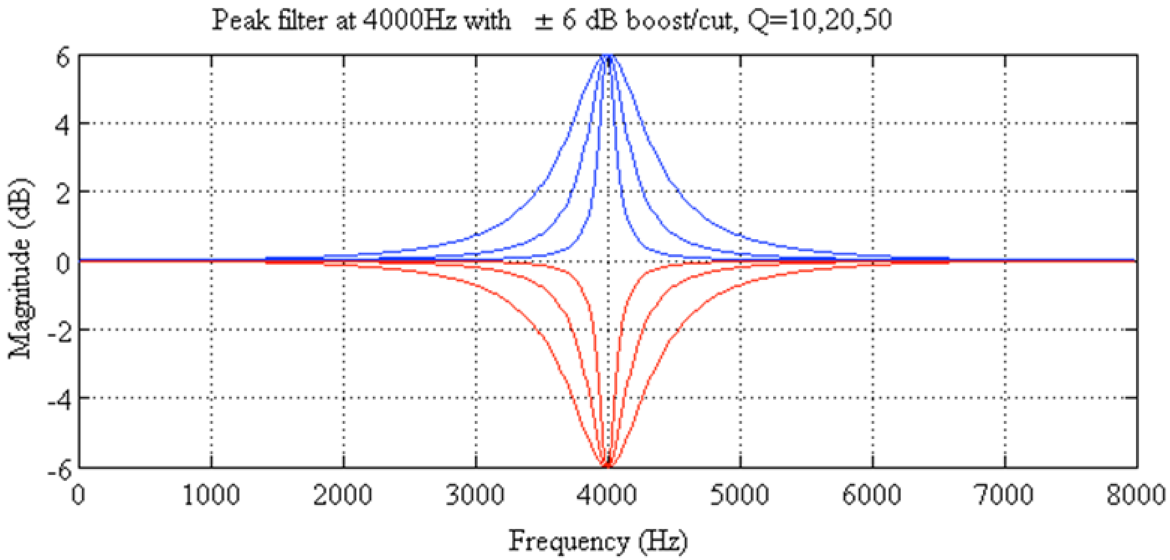




Tampere University

Peak filter

Another type of filter for boosting/cutting desired frequencies



Peak filter

For those who want more details, not important for exam

- The transfer function of a peak equalizer

$$H_{pk}(z) = C_{pk}\left(\frac{1+b_1 z^{-1}+b_2 z^{-2}}{1+a_1 z^{-1}+a_2 z^{-2}}\right), C_{pk} = \frac{1+k_q\mu}{1+k_q}, k_q = \left(\frac{4}{1+\mu}\right)\tan\left(\frac{\Omega_c}{2Q}\right)$$

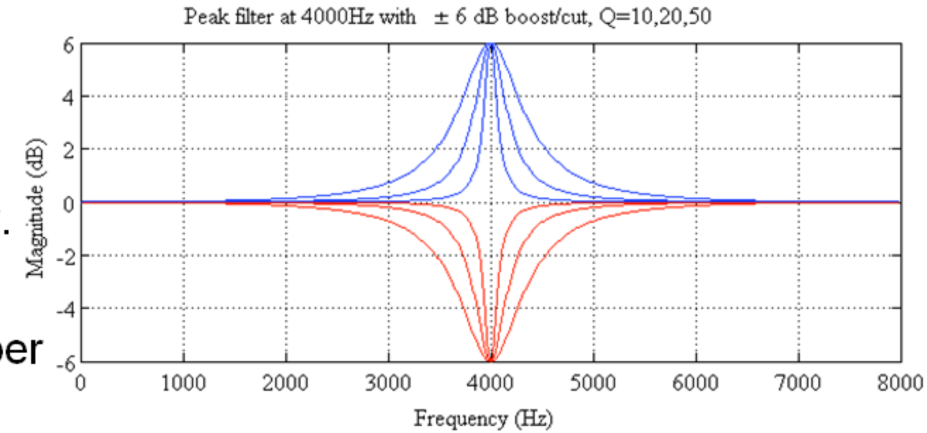
$$b_1 = \frac{-2\cos(\Omega_c)}{1+k_q\mu}, b_2 = \frac{1-k_q\mu}{1+k_q\mu}, a_1 = \frac{-2\cos(\Omega_c)}{1+k_q}, a_2 = \frac{1-k_q}{1+k_q},$$

Q - quality factor

G - gain (dB)

Ω_c - normalized center frequency.

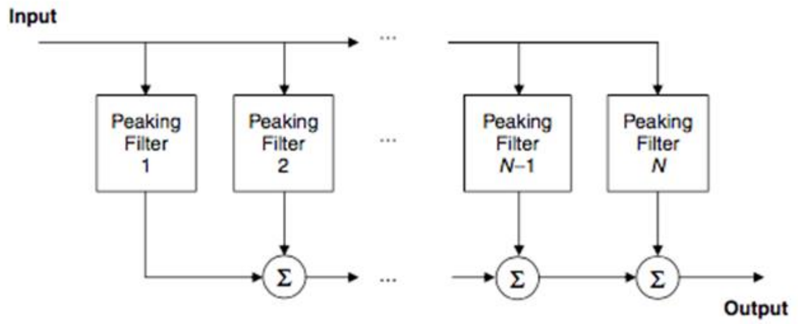
- high Q -> sharper



Tampere University

Graphic equalizers

- A cascaded setup of peaking filters to design an audio graphic equalizer



- Audio spectrum is divided into several frequency bands (given center frequency)
- Each band’s gain (G) is controlled by the user

Analog equalizer

Parametric equalizer

- Multi-band variable equalizers which allow users to control the three primary parameters: amplitude, center frequency and bandwidth
- Capable of much more precise adjustments to sound than other equalizers; commonly used in sound recording and live sound reinforcement

Creative use of equalizers

- Instead of fixing a problem, creativeness can be expressed with equalizers
- **Fullness**: adding +4 to +6dB 100-300Hz range to emphasize weak instruments (e.g. acoustic guitar, harp)
- **Crispness**: for percussion instruments by adding HF shelving boost above 1-2 kHz.
- Emphasize articulation transients of instruments. E.g. boost frequencies of finger movements on string instruments.

Summary

Dynamic Range Control:

- What is DRC and where it is needed
- The basic idea of dynamic range compression
- Level measurement and static curve define gain value
- Different operation modes: Noise gate, Expander, Compressor, Limiter

Equalizers:

- What is equalization and how it differs from filtering
- What types of equalization operations exist
- Parametric equalizers and their use