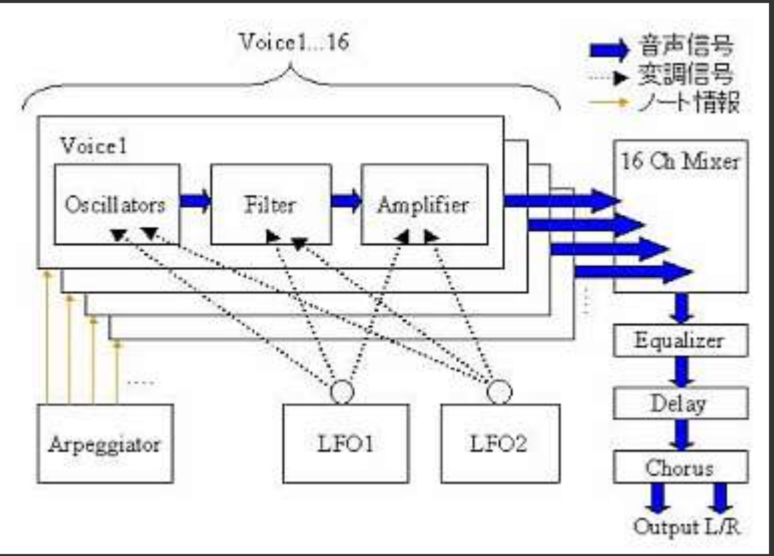
SOUND EFFECTS

in electronic musical instruments

Voice modes in synthesizers

- *Poly* – polyphonic mode, multiple voices.
- Default mode in modern EMIs.
- An instrument can generate *N* voices at the same time (usually 16 voices).
- If we press 16 keys concurrently, 16 sounds with different pitches will be generated.
- Polyphony allows us to play chords.
- For each voice, a separate synthesis system is needed.



Unison mode

- *Unisono* = multiple instruments play the same pitch.
- In the unison mode, two or more voices are generated for each note (each pressed key).
- Voices are slightly detuned from each other. Parameters:
  - *voice number* – how many voices per note,
  - *detune* – how much they differ in pitch,
  - *phase shift* – shifts the voices in time.
- Unison mode creates “fatter” sounds.
- Decreases the number of available voices for polyphony.
- *Clavia Nord Lead 2* uses two-voice unison mode by default.

Monophonic modes

- Mono mode simulates monophonic instruments:
  - only one voice is generated at a time,
  - usually for the latest note (pressed key),
  - a new sound starts with the attack phase.
- Legato mode – similar to *mono*, but:
  - new sound only alters the pitch of the currently playing sound, no new attack phase is created,
  - *portamento time* – controls the transition time between sounds,
  - creates smooth pitch transition between notes.

Arpeggiator

- *Arpeggio* – „breaking chords”: if multiple keys are pressed at the same time, notes are played one at the time, one after another. The sequence is looped.
- *Arpeggiator* – instrument module that creates arpeggio.
- Settings:
  - *gate* – duration of each note,
  - *tempo* (*beat*) – repetition rate,
  - sweeping mode: *up*, *down*, *up-down*, *random*
  - octave changes during repetition: none, +1, etc.
- *Arpeggiator* allows us to create an impression that a skilled musician plays very fast.

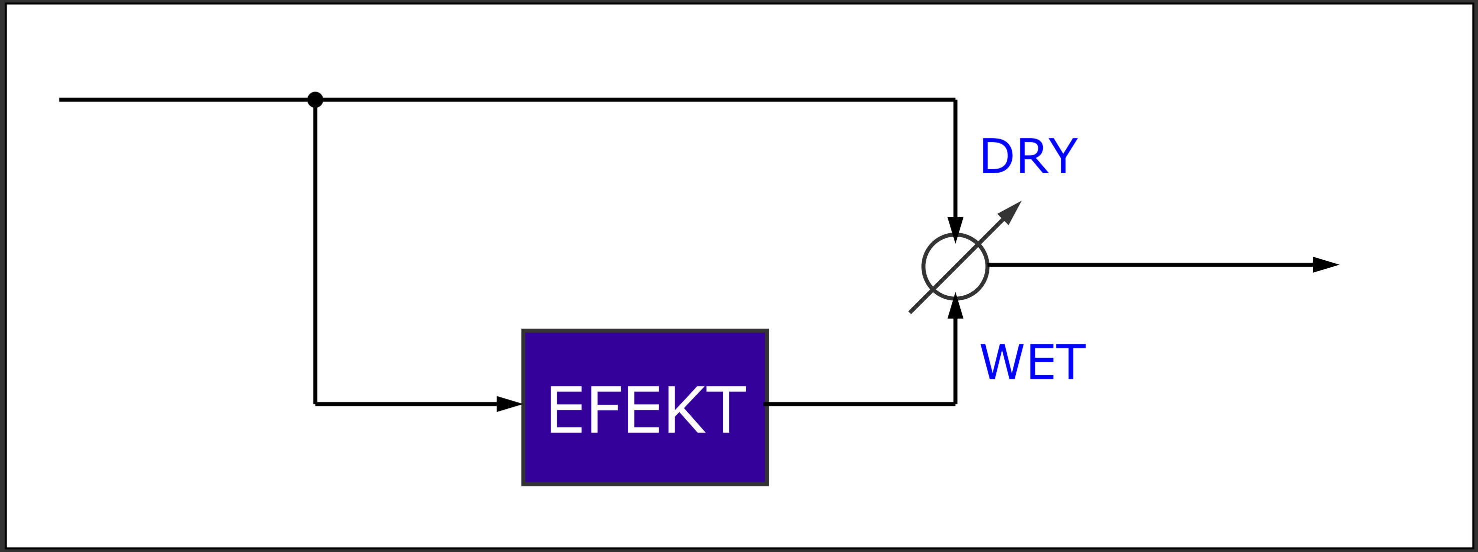
Sound effects in EMI

- Sound effects in EMIs are used to enhance the produced sounds.
- Effects are added at the final stage, before the output.
- Sound effects are not the part of the synthesis process.
- They are practically always used in synthesizers, in order to get a “fuller” sound.
- We need to adjust the parameters so that the sound is enhanced, but it’s easy to ruin the sound by overdoing the effect.

Dry and wet sounds

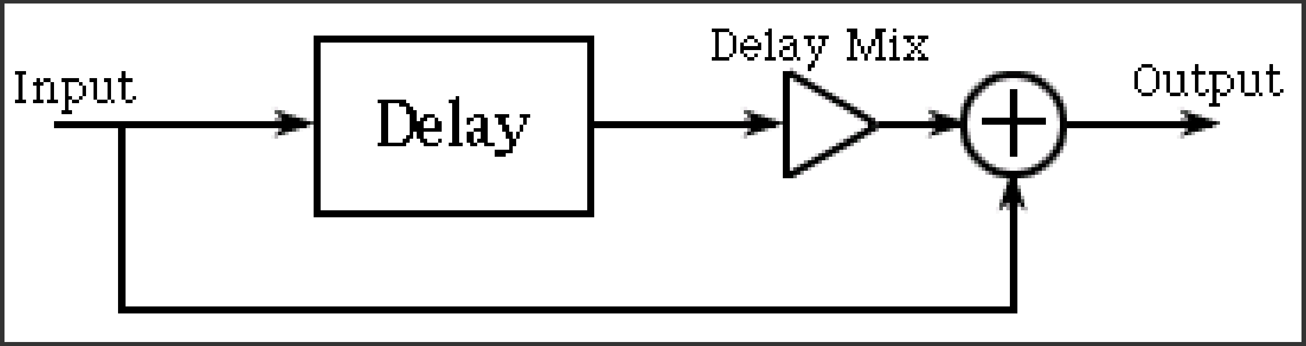
In many effects, two sounds are mixed with a regulated proportion, to control the strength of the effect:

- *wet* – a sound passed through the effect,
- *dry* – the original, unprocessed sound.



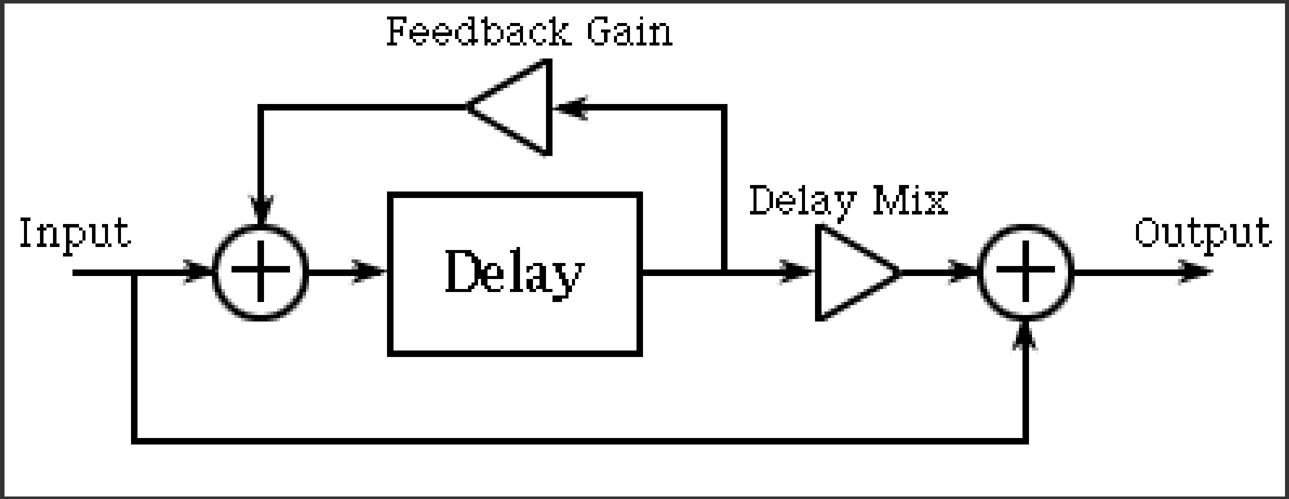
Delay / Echo

- The delay effects adds a delayed copy of the signal to the original one.
- A buffer is used to store the delayed signal samples.
- The length of the buffer determines the delay time.
- For small delay (ca. 10 ms), overlapped sounds are achieved.
- For longer delay (ca. 50 ms), an echo effect is created.



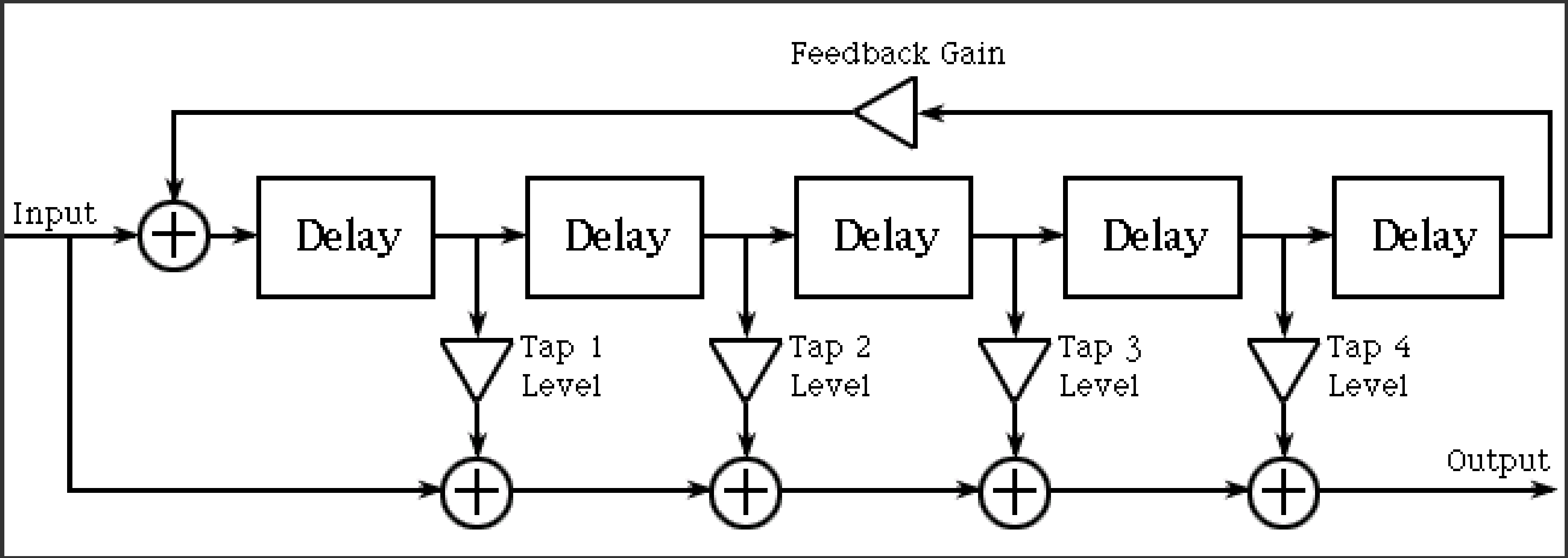
Delay with a feedback

- Usually, the delay effect has a feedback loop.
- It allows for creating multiple, repeated reflections.
- Amplitude of the reflections decreases in time.
- The feedback gain controls duration of the effect.
- The delay between reflections is constant.



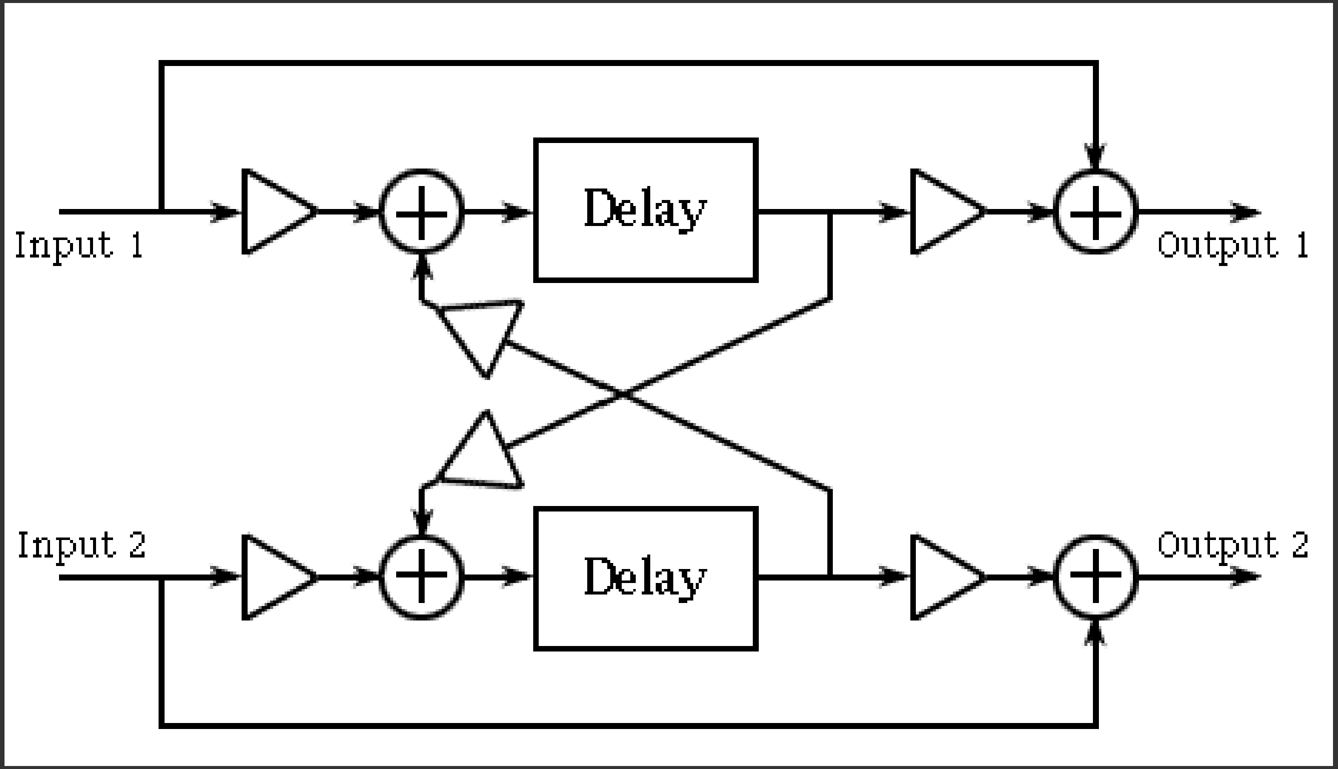
Multi-tap delay

- The buffer has many outputs (taps).
- By summing the outputs, we get multiple repetitions.
- Delays between the copies may be changed, they don’t have to be the same.
- The feedback gain adds repetition to the sound.



Ping-pong delay

- This effect operates on stereo channels.
- The delayed signal bounces between the channels.
- Effect of varying stereo panorama.



Tempo delay

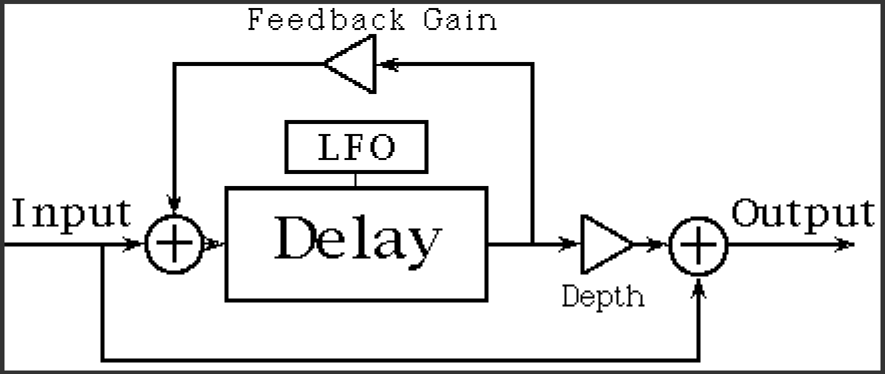
In EMIs, a tempo delay is normally used. The delay time is expressed in note duration values, relative to the beat.

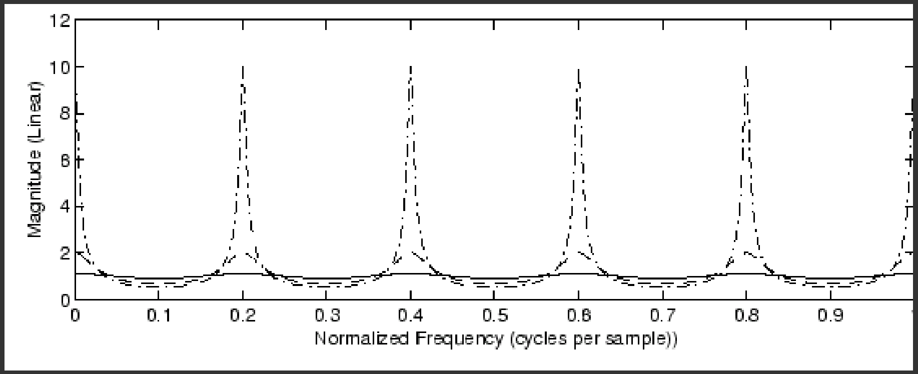
Typical parameters of the tempo delay effect:

- *delay time* – expressed as note duration, e.g. [4] means that the delay is equal to duration of a quarter note,
- *feedback* – gain of the feedback loop,
- *tone* – cut-off frequency of the low-pass filter in the feedback loop,
- *wet/dry* – the mixing proportion (strength of the effect).

Chorus / Flanger

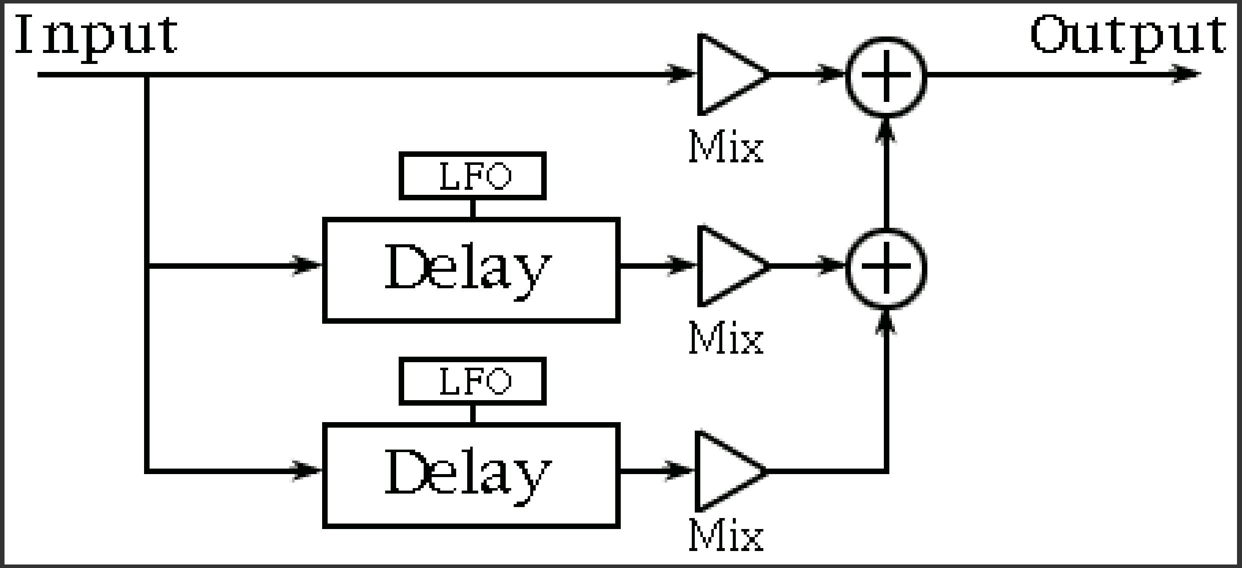
- The *chorus* and *flanger* are similar to *delay*, but the delay time is modulated by an LFO.
- A standard delay is a comb filter.
- Modulation moves the spectral peaks on the frequency axis, introducing cyclic timbre changes.
- Fractional delay buffers are needed.





Multi-voice chorus

A chorus with multiple voices requires several chorus blocks in parallel, with different delay and LFO parameters.
Their outputs are summed.



Chorus / Flanger

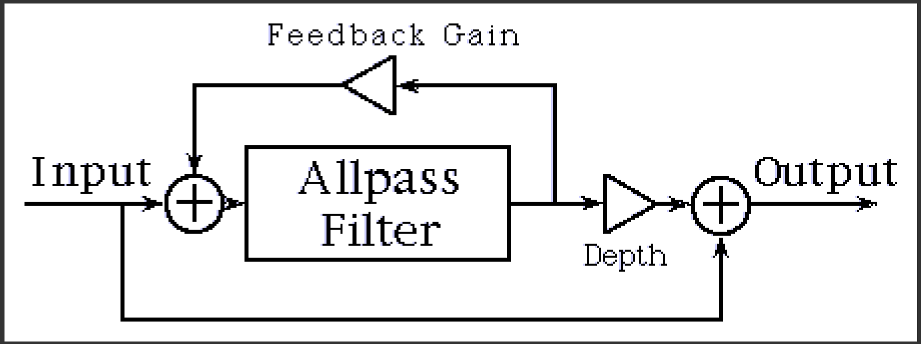
Parameters of a chorus/flanger module:

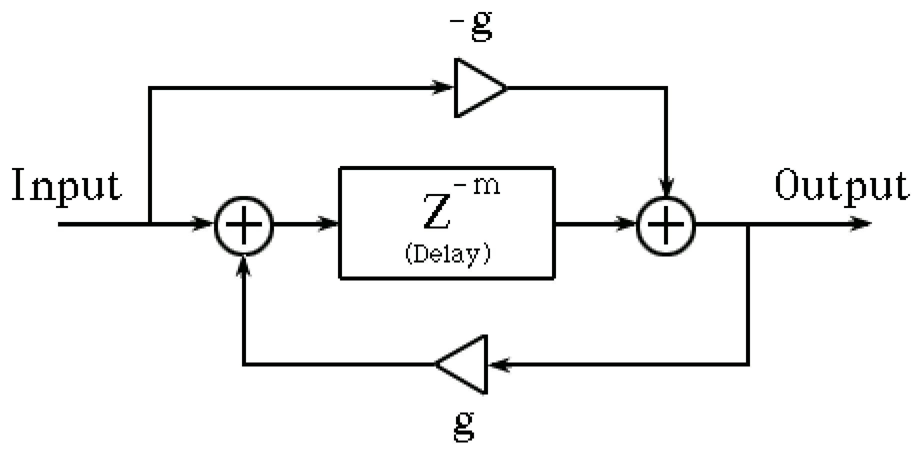
- *delay time* – delay between copies:
  - – small values (< 10 ms): *flanger*,
  - – higher values (> 20 ms): *chorus*,
- *feedback* – gain of the feedback loop,
- *rate* – LFO frequency, rate of timbre changes,
- *depth* – LFO amplitude, range of timbre changes,
- *level* or *wet/dry* – mixing proportion (strength of the effect).

Sometimes the effect is multi-stage (1, 2 or 4 stages).

Phaser

- Phaser is similar to the flanger, but an all-pass filter replaces the delay buffer.
- Only the signal phase is modulated.
- The amplitude spectrum is unchanged (no comb).
- Usually, multiple sections (1 to 6) are used.

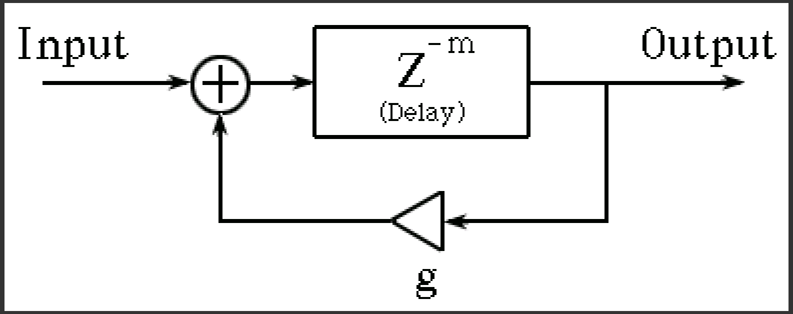




All-pass filter

Reverb

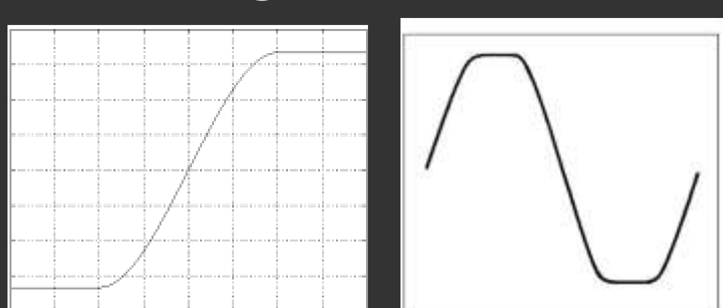
- In theory, a reverb effect should simulate sound reflection in a room. Delay between reflections decreases with time. This is difficult to simulate.
- In EMIs, the term “reverb” is often used incorrectly for:
  – a standard delay effect, or
  – a delay with an all-pass filter,
  – optionally with a feedback loop:



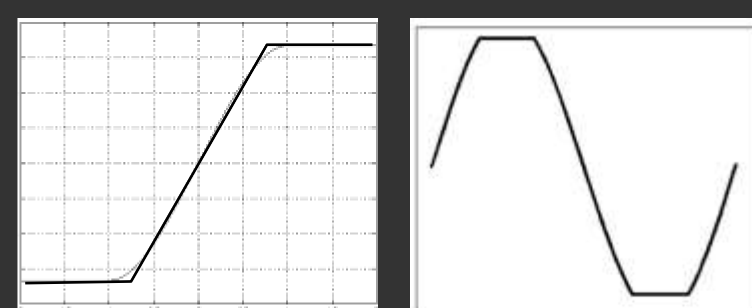
Distortion

- The distortion effect passes the sound through a module with a nonlinear input-output function.
- Harmonic distortion is introduced to the sound.
- Analog distortion uses soft clipping, the sound is distorted gently,
- Digital distortion uses hard clipping which makes the sound very harsh.

Analog distortion



Digital distortion

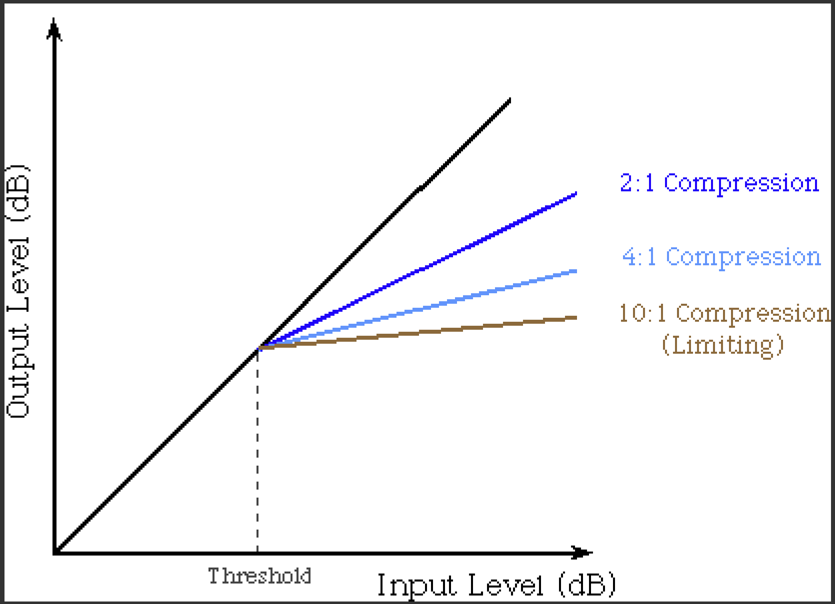


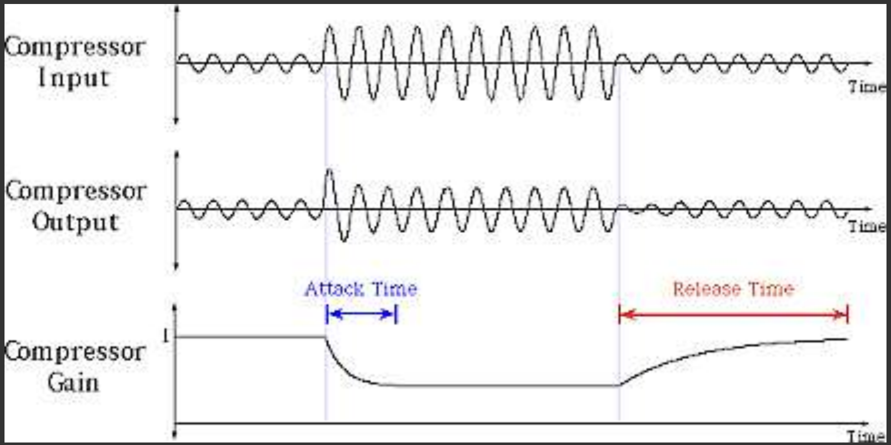
Other effects

- Ring modulator multiplies the signal by a sine with a regulated frequency. A rich, inharmonic sound is created.
- Decimator (Bit Crusher) decreases the bit resolution by removing the least significant bits, creating a digital distortion, simulating e.g. 8-bit sound chips.
  - *Sample rate* changes the sampling frequency and introduces aliasing.
  - *Depth* sets the bit resolution.

Compressor

- A compressor attenuates louder sound components.
- *Level* controls the "knee" position.
- *Depth* controls the slope of the compressed section.
- *Attack time* controls the reaction time of the compressor.





Equalizer and panning

Panning – sets the balance of left/right stereo channels:

- L/R – a proportion of channels,
- it may be modulated with the LFO.

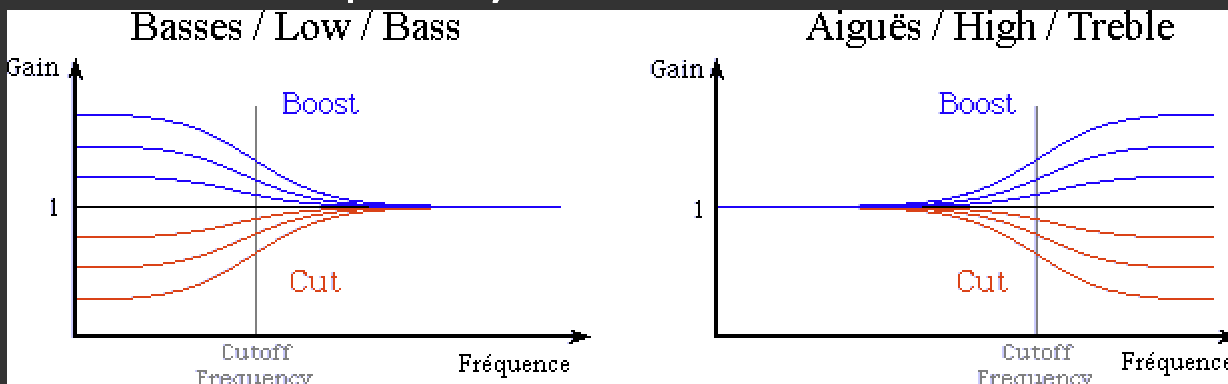
Equalizer – a peak filter that processes the output signal. Parameters:

- *frequency* – the center frequency of the passband,
- *Q* – quality, the width of the passband,
- *level* – the amount of gain / attenuation in the passband.

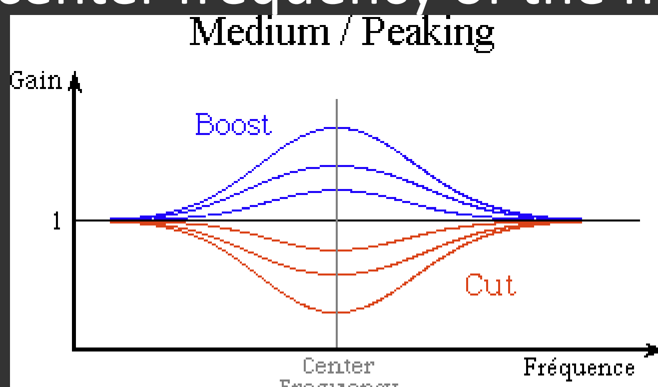
Tonal equalizer

An equalizer in EMIs often consists of two filters.

- The first one is a shelving filter. The *tone* parameter sets the cut-off frequency.



- The second one is a peak filter. The *frequency* parameter sets the center frequency of the filter.

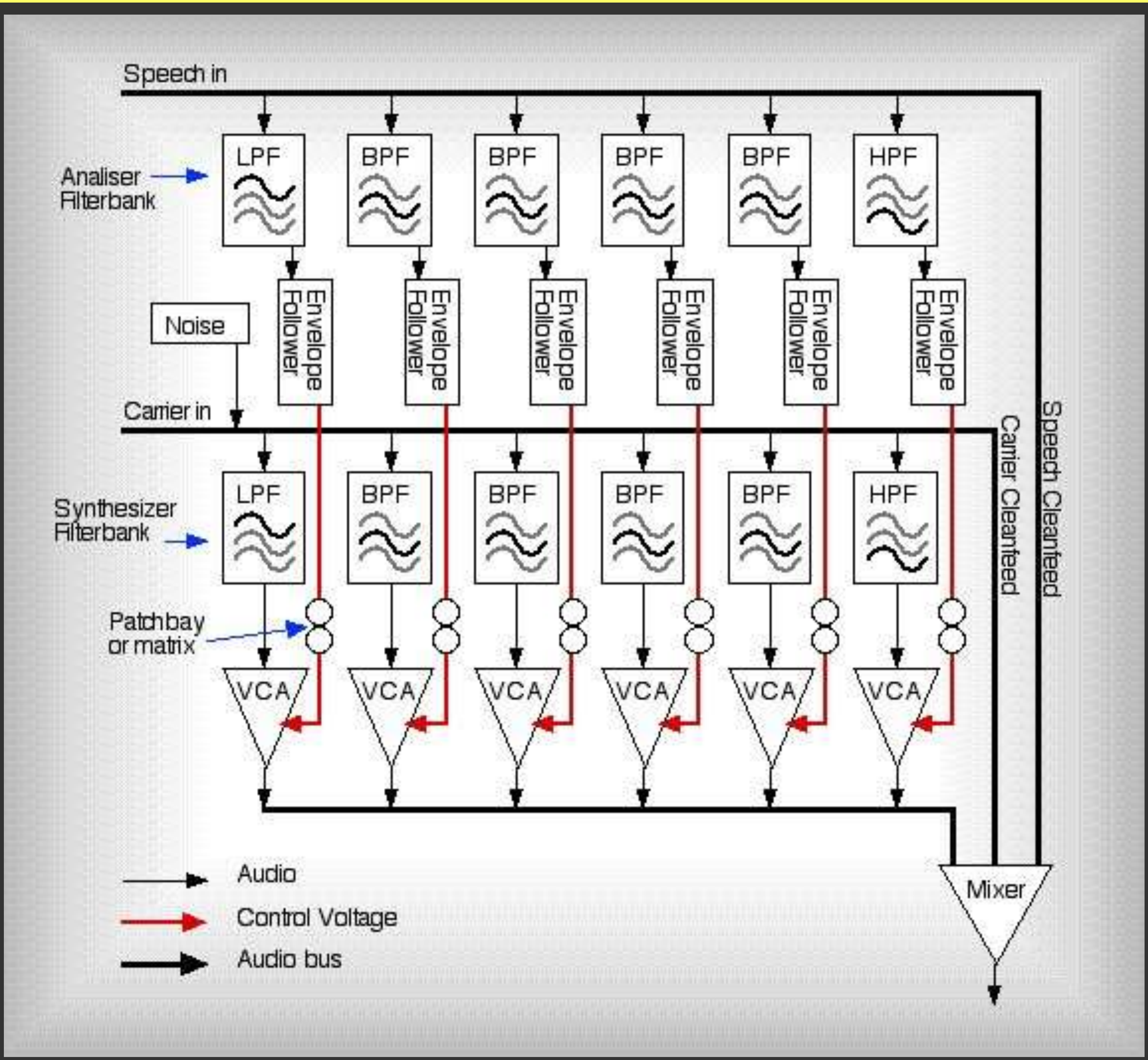


Vocoder

Vocoder (voice coder) is used as a sound effect.

- The signal is divided into frequency bands.
- A loudness envelope (a function of loudness changes) is extracted from each band.
- These envelopes are then used to modulate the amplitude of a carrier signal in the same bands.
- A typical example:
  - envelopes are extracted from the vocal,
  - they are used to modulate a synthesizer sound,
  - a “singing synthesizer” effect is obtained.

Block diagram of a vocoder



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