CTP 431 Music and Audio Computing

Delay-based Audio Effect

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Introduction

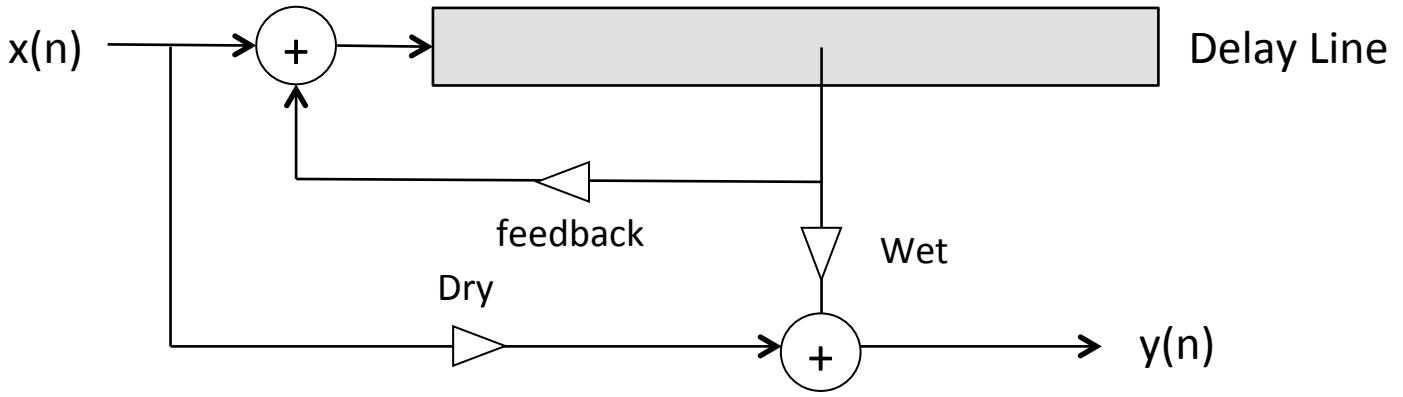
- Types of delay-based audio effect
  - Delay
  - Chorus
  - Flanger
  - Reverberation

Perception of Time Delay

- The 30 Hz transition
  - Given repeated click sound (e.g. impulse train):
    - If the rate is less than 30Hz, they are perceived as discrete events.
    - As the rate is above 30 Hz, they are perceive as a tone
  - Demo: http://auditoryneuroscience.com/?q=pitch/click_train

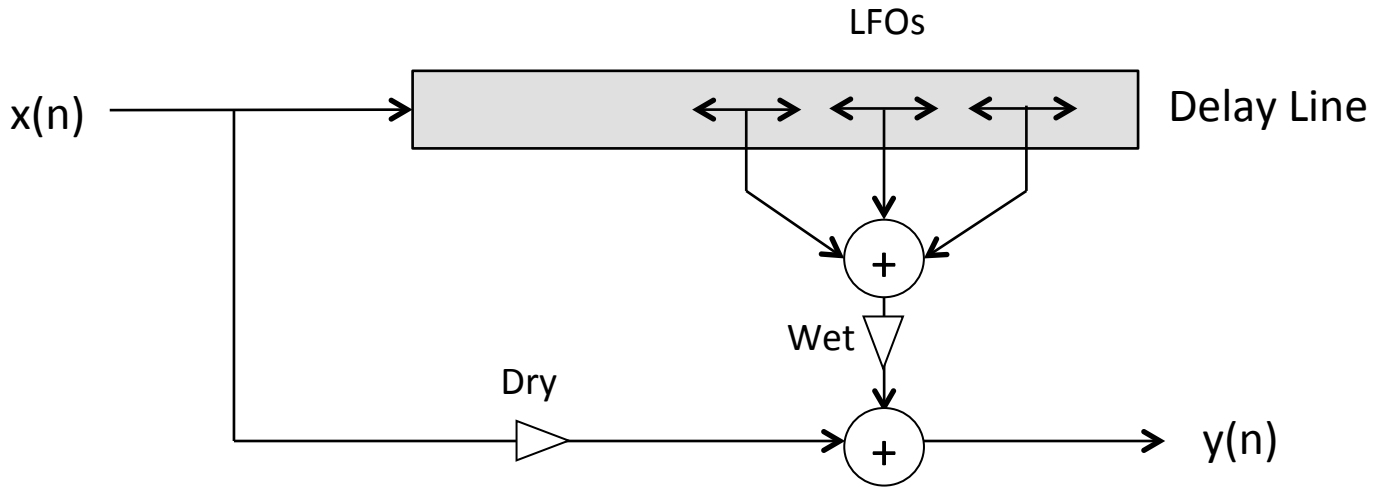
- Feedback comb filter: $y(n) = x(n) + r\, y(n-M)$
  - Models sound propagation and reflection with energy loss
  - If $M < fs/30$: generate a tone
    - E.g. Karplus-strong model of tone generation (https://en.wikipedia.org/wiki/Karplus%E2%80%93Strong_string_synthesis)
  - If $M > fs/30$: generate a looped delay
    - E.g. Delay effect

Delay



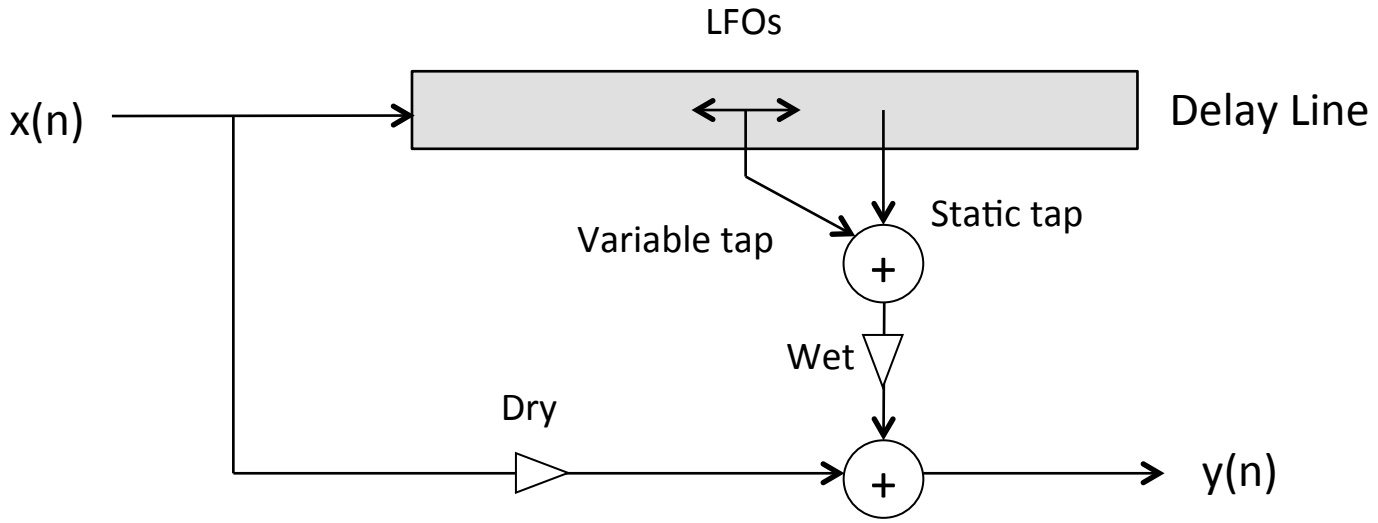
- Delay effect
  - Generate repetitive loop delay
  - Feedback coefficient controls the amount of delayed input
  - Can be extended to stereo signals such that the delay output is "ping-ponged" between the left and right channels
  - The delay length is often synchronized with music tempo
  - The delayline is implemented as a "circular buffer"

Chorus



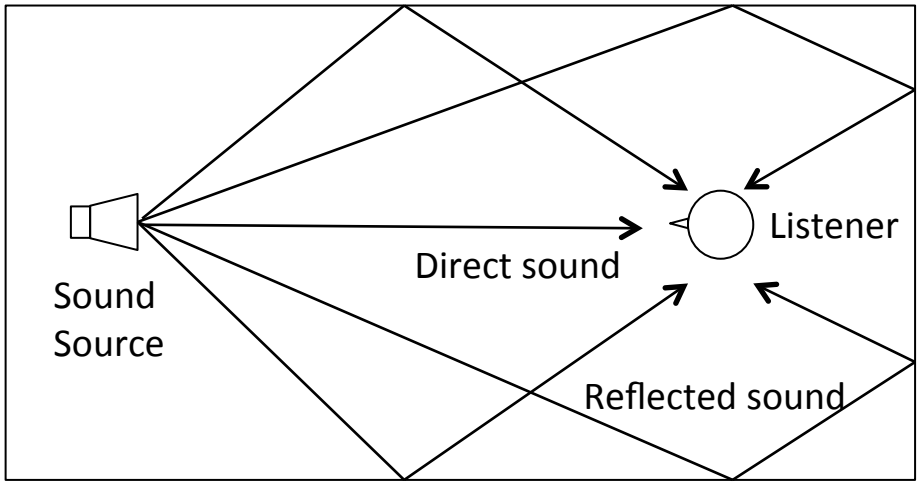
- Chorus effect
  - Gives the illusion of multiple voices playing in unison
  - By summing detuned copies of the input
  - Low frequency oscillators are used to modulate the position of output tops → This causes the pitch of the input (resampling!)

Flanger



- Flanger effect
  - Originally generated by summing the output of two un-locked tape machines while varying their sync (used to be called “reel-flanging”)
  - Emulated by summing one static tap and variable tap in the delay line
    - Feed-forward combine filter where harmonic notches vary over frequency.
  - LFO is often synchronized with music tempo

Reverberation



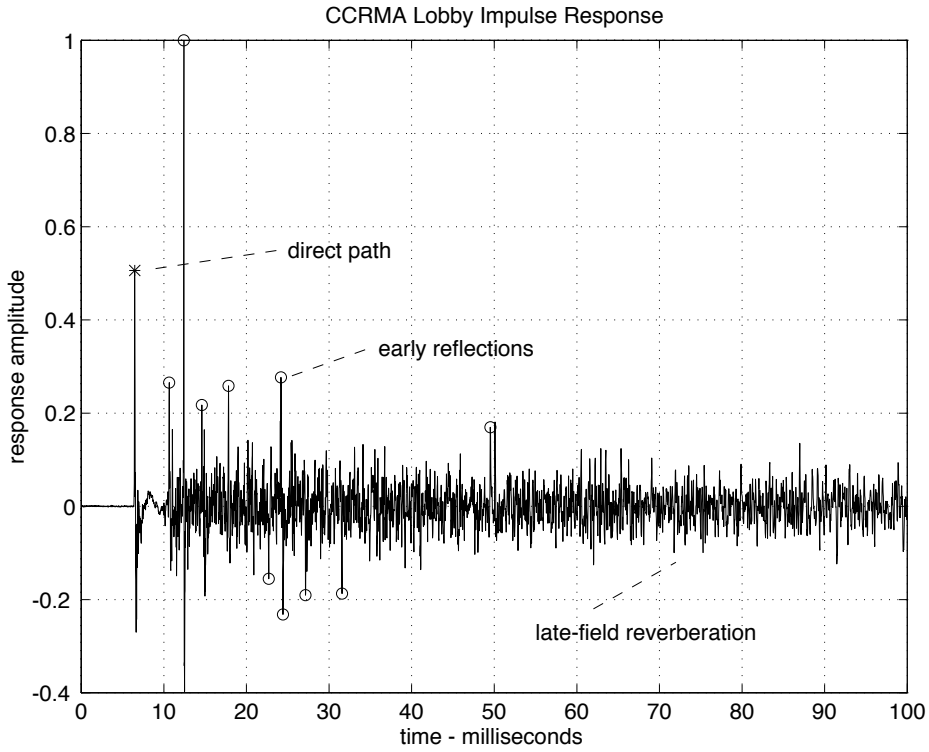
- Natural acoustic phenomenon that occurs when sound sources are played in a room
  - Thousands of echoes are generated as sound sources are reflected against wall, ceiling and floors
  - Reflected sounds are delayed, attenuated and low-pass filtered: high-frequency component decay faster
  - The patterns of myriads of echoes are determined by the volume and geometry of room and materials on the surfaces

Reverberation

- Room reverberation is characterized by its impulse response (IR)
  - E.g. when a balloon pop is used as a sound source

- The room IR is composed of three parts
  - Direct path
  - Early reflections
  - Late-field reverberation: high echo density

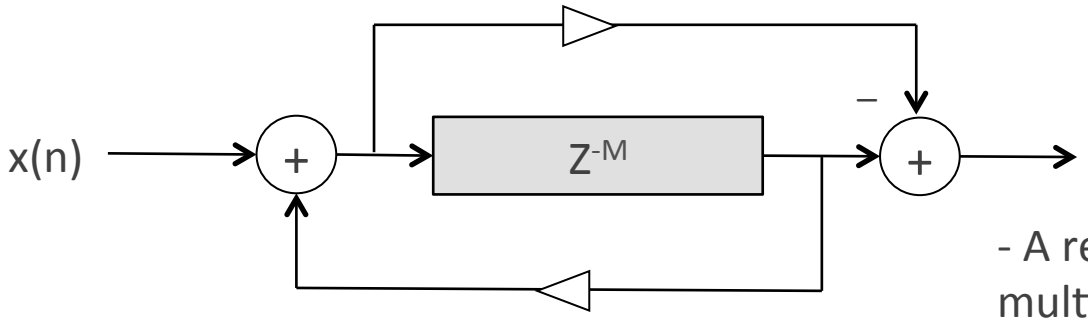
- RT60
  - The time that it takes the reverberation to decay by 60 dB from its peak amplitude



Artificial Reverberation

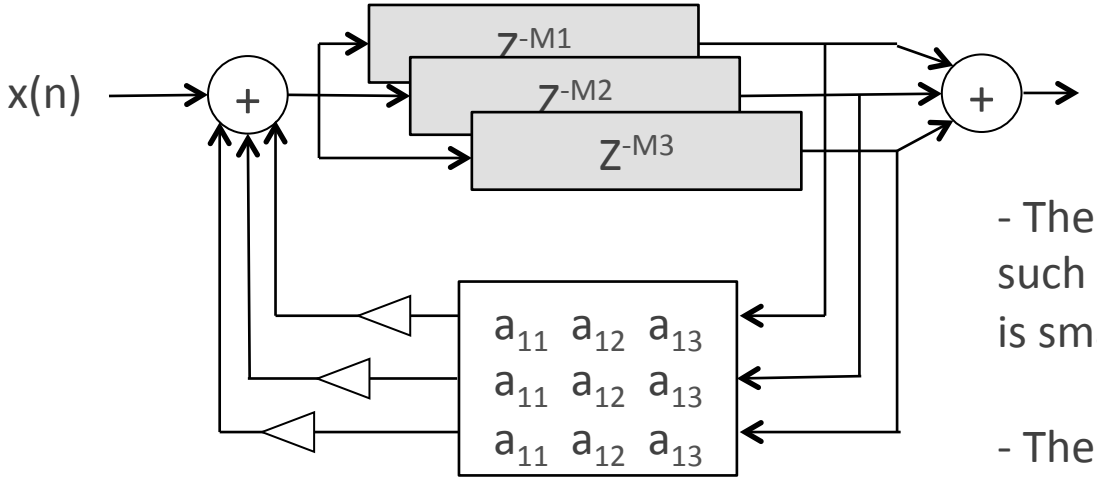
- Mechanical reverb
  - Use metal plate and spring
  - Plate reverb: https://www.youtube.com/watch?v=XJ5OFpvX5Vs
- Delayline-based reverb
  - Early reflections: feed-forward delayline
  - Late-field reverb: allpass/comb filter, feedback delay networks (FDN)
  - “Programmable” reverberation
- Convolution reverb
  - Measure the impulse response of a room
  - Do convolution input with the measured IR

Delay-based Reverb



- A reverb is constructed by cascading multiple AP or FFCF units

AllPass filter / Comb filter (when one tap is absent)



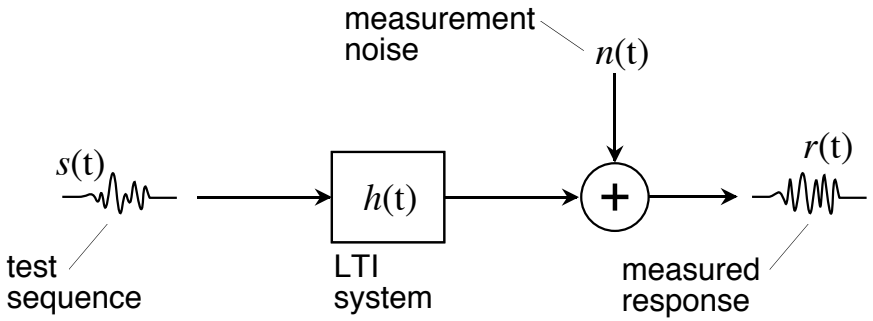
- The lengths of delaylines are chosen such that their greatest common factors is small (e.g. prime numbers)

- The mixing matrix is chosen to be unitary (orthonormal)

Feedback Delay Networks

Convolution Reverb

- Measuring impulse responses

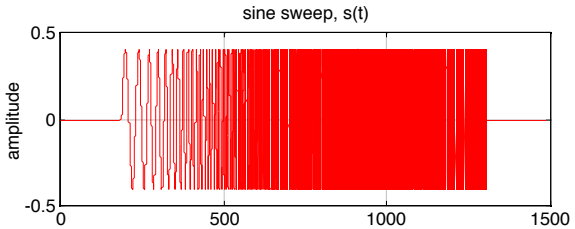


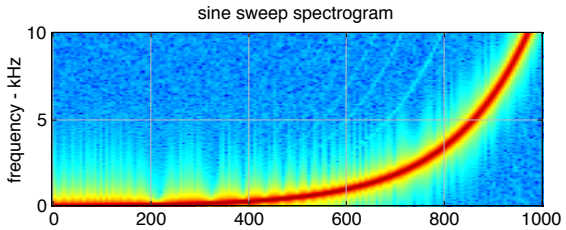
$$r(t) = s(t) * h(t) + n(t)$$

  - If the input is a unit impulse, SNR is low
  - Instead, we use specially designed input signals
    - Golay code, allpass chirp or sine sweep: their magnitude responses are all flat but the signals are spread over time
  - The impulse response is obtained using its inverse signal or inverse discrete Fourier transform

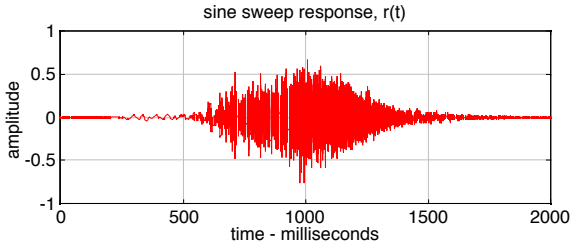
Convolution Reverb

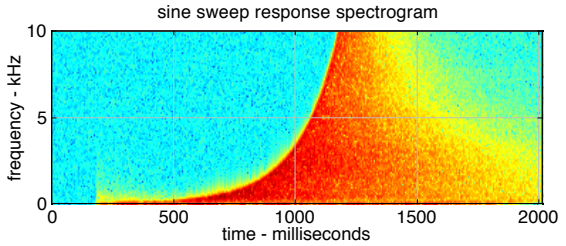
$s(t)$



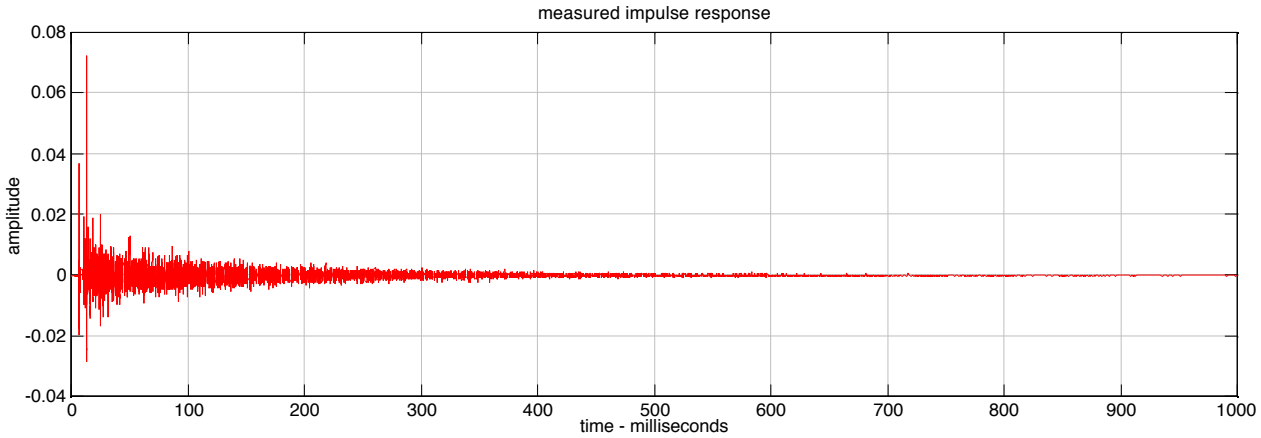


$r(t)$





$\hat{h}(t)$



(J. Abel)