

STOMPBOX DESIGN WORKSHOP

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FX Basics: Time Effects

Time-based effects are built upon the artificial introduction of delay and creation of echoes to be added to the original signal.

Emerged in the late 1940s and were created by loops of tape or other recording media; variable delay was achieved by changing write/read heads.

The idea behind time-based digital effects is to **temporarily store** a portion of the input signal **into a buffer of variable length**, and **recover it later** for mixing it with the original.

Ex: delay/echo, flanger, phaser, reverb



Delay / Echo

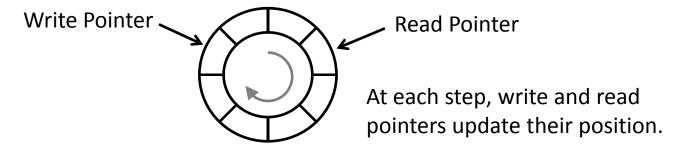
Produce the effect of an echo by creating a duplicate of the input signal and adding it with a slight time delay.

In order to present the simplest approach to digital delay, let's first introduce the concept of **delay line**:

→ At each seq. order (or time) *n*, it outputs the sample fed in at time *n-M*



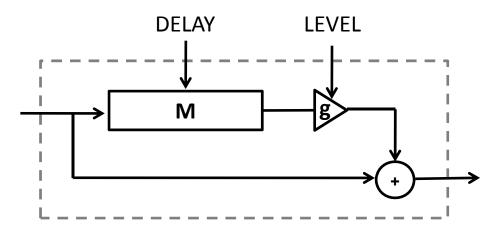
→ Usually implemented as fixed length buffer with write and read pointers 'spaced' M samples from each other:



Delay / Echo (ii)



The simplest, single-echo delay digital effect can be constructed with a variable length delay line plus a gain control:

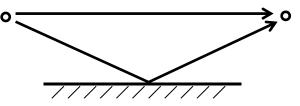


If the 'DELAY' control is set to be expressed in 'seconds', such value will have to be converted to 'number of samples'...

What if M needs to be non-integer?

FRACTIONAL DELAY!

Can be used to simulate a simple acoustical echo:



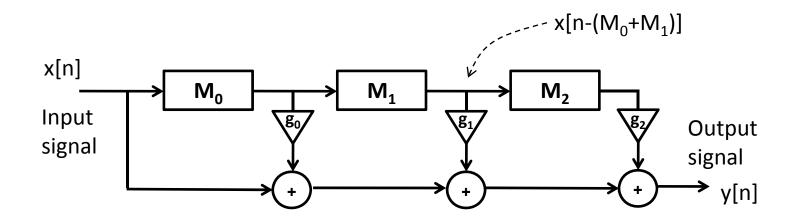


07_stomp_time_1.pd



Delay / Echo (iii)

By cascading several delay lines, one can obtain a **tapped delay** effect, which leads to a multiple echo:



pd~ 08_stomp_time_2.pd

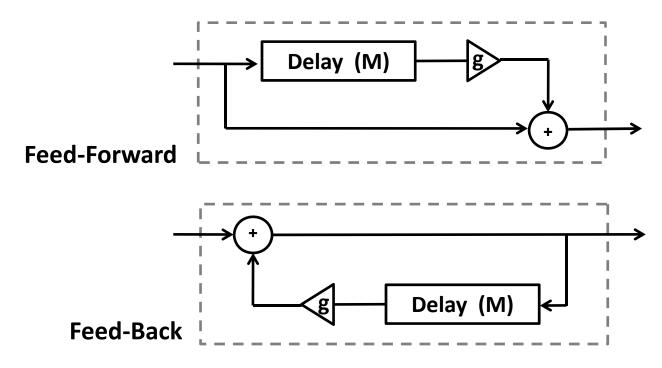
Before getting further with time-based effects: **COMB FILTERS**



Comb Filters

A comb filter adds a delayed version of a signal to itself.

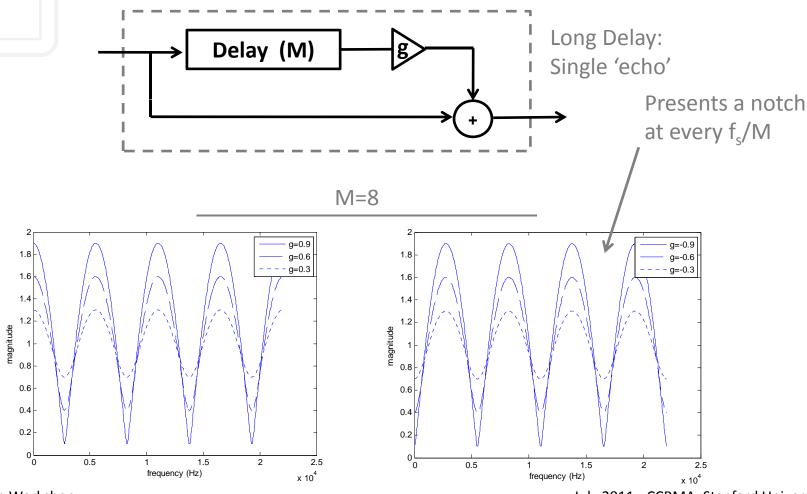
Ex: the single-echo effect presented before represents an instance of a comb filter.



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Comb Filters (ii)

Feed-Forward Comb Filter



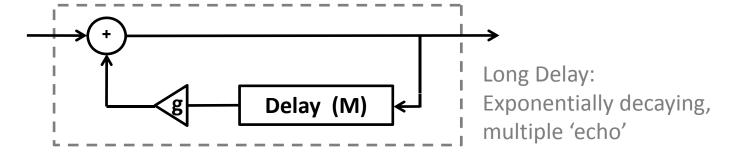
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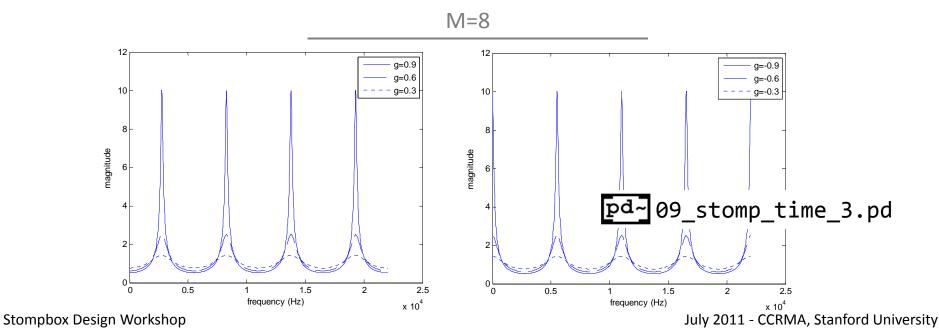
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Comb Filters (iii)

Feed-Back Comb Filter



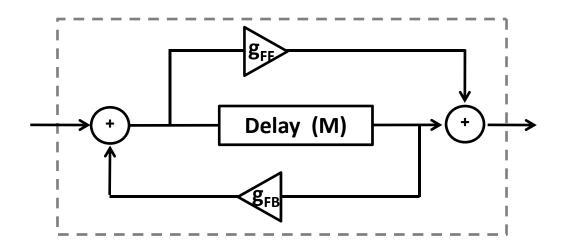


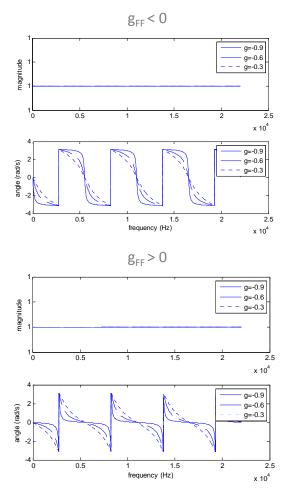


FX Basics: Time Effects

All-Pass Filter from Two Comb Filters

By cascading a Feed-Forward Comb Filter (FFCB) and a Feed-Back Comb Filter (FBCB), one obtains a particular All-Pass Filter whenever $\mathbf{g}_{\mathsf{FF}} = -\mathbf{g}_{\mathsf{FB}}$.





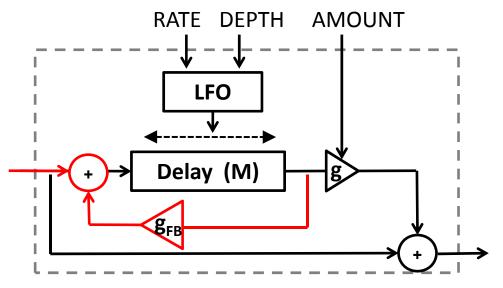
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Flanger

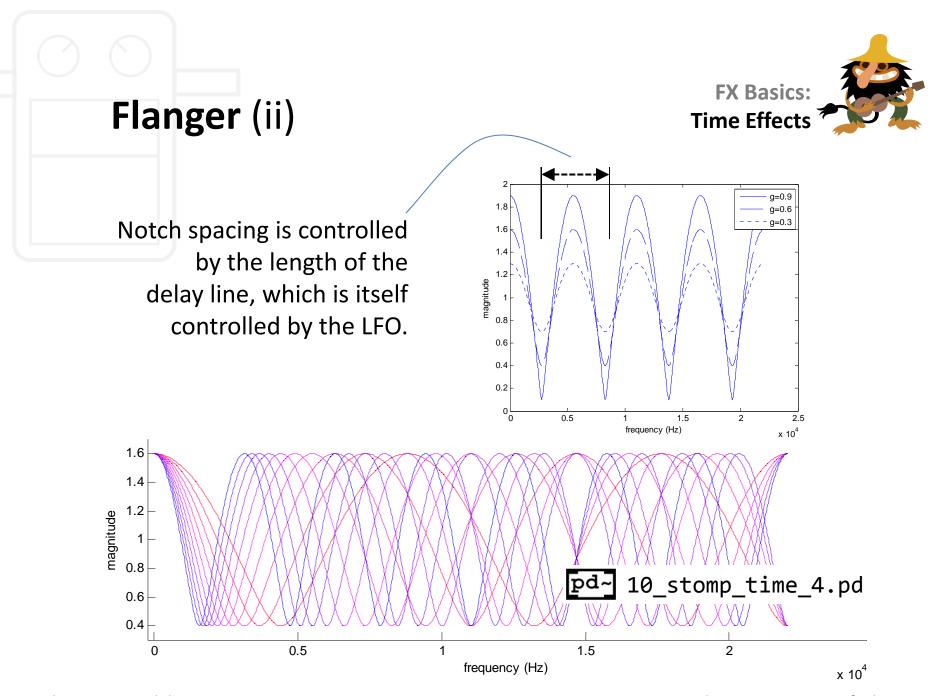
Available since the 1960s in recording studios, it was originated by using 2 tape machines (playing in unison) while pressing and releasing the flange of one of them, and thus introducing a changing, short delay between read signals before being mixed.

A simple flanger can be modeled as a **LFO-controlled**, **variable-delay FFCF**:



Harmonic series of notches in magnitude response; notches are uniformly spaced (at f_s/M).

Sometimes, a Feedback control can be added.

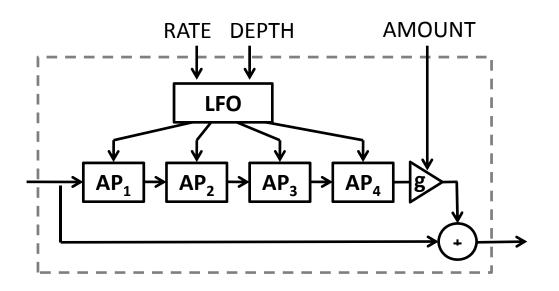




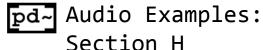
Phaser / Phase shifter

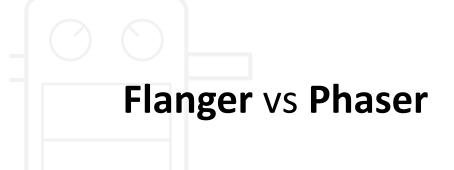
Closely related to the Flanger, it dates from the 1960s, too.

Also based on slightly delaying a signal and adding it to itself, substitutes the variable delay line of the Flanger by a **cascade of low-order All-Pass filters**.



Notches in magnitude response **are finite** (as a function of the number of stages.

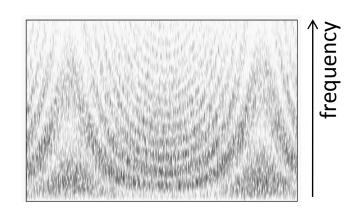




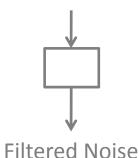


FLANGER

Infinite series of notches, uniformly spaced

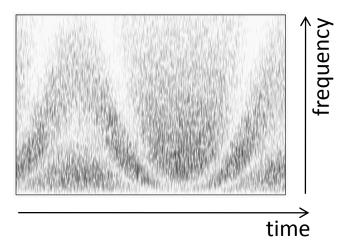


White Noise (Flat Spectrum)



PHASER

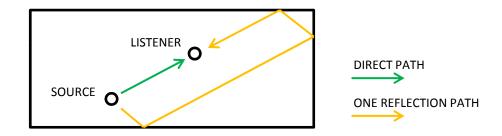
Finite series of notches, arbitrarily located





Reverb

In real spaces, reverberation arises from a complicated pattern of sound reflections off the walls and other objects.



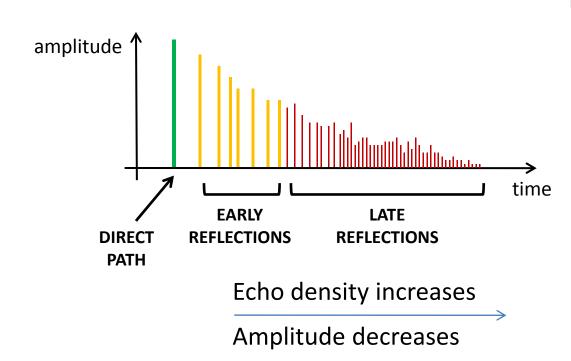
Artificial reverberation represents a very challenging problem, presenting a very **high computational cost** when modeled **from** a purely **physical perspective** (too many computations needed to simulate sound propagation in a 3D space).

However, it is possible to construct **efficient** artificial reverberation **models using delay lines** as basic building blocks.



Reverb (ii)

The profile of a reverberation can be **modeled** as sequence of delayed copies (echoes) of the source sound:



RELEVANT MEASURES

Arrival time of first reflection

Should be below 40-50ms, or it may be perceived as echo.

Reverberation time (T_{60}) :

Time needed to drop 60dB. Larger, less absorbent spaces present a higher T_{60} value.

Echo density increase rate

Linked to T_{60} , should show a behavior inversely related to space size.

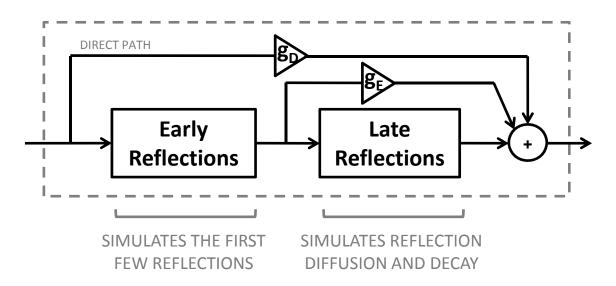


Reverb (iii)

One can find many strategies for constructing, via delay lines, artificial reverberators that result perceptually satisfactory.

It is **not straightforward** to design delay line-based reverberators so that target measures can be met.

A common approach is to use **2 different stages**, each one in charge of representing the two differentiated observed behaviors:

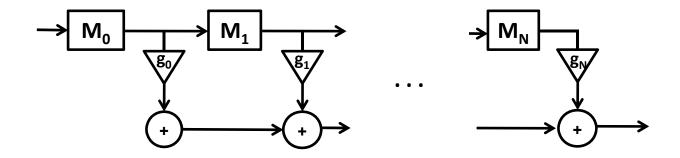


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Reverb (iv)

EARLY REFLECTIONS

One can use a tapped delay line line (one tap per reflection) with tuned delays \mathbf{M}_n and gains \mathbf{g}_n .



It is suggested that none of the taps' delay exceeds **40-50ms**, since it is acknowledged as the threshold for echo perception.

An idea is to control delays and gains with a 'shared' parameter.

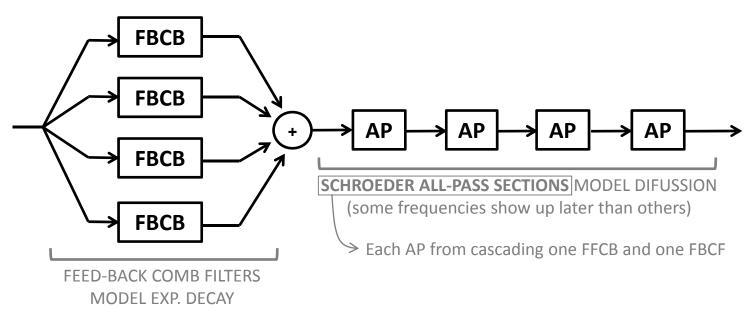


Reverb (v)

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LATE REFLECTIONS (including DIFFUSION)

Different variations over structures based on cascading AP sections with particular settings (*Schroeder All-Pass Sections*):



Delay line lengths must be set to be **mutually prime**, so **pd** 'Freeverb' smooth decay and echo density increase are ensured.

http://www.bagger288.com/temp/aboutThisReverberationBusiness.pdf
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