# FX Basics Time Effects

#### STOMPBOX DESIGN WORKSHOP

Esteban Maestre

CCRMA - Stanford University August 2013

## **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.

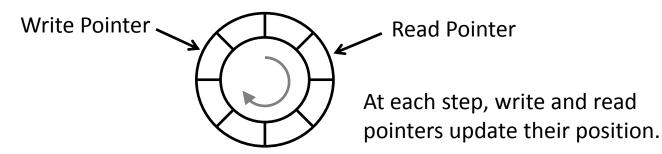
In order to present the simplest approach to digital delay, let's first introduce the concept of **delay line**:

 $\rightarrow$  At each seq. order (or time) *n*,

it outputs the sample fed in at time *n*-*M* 



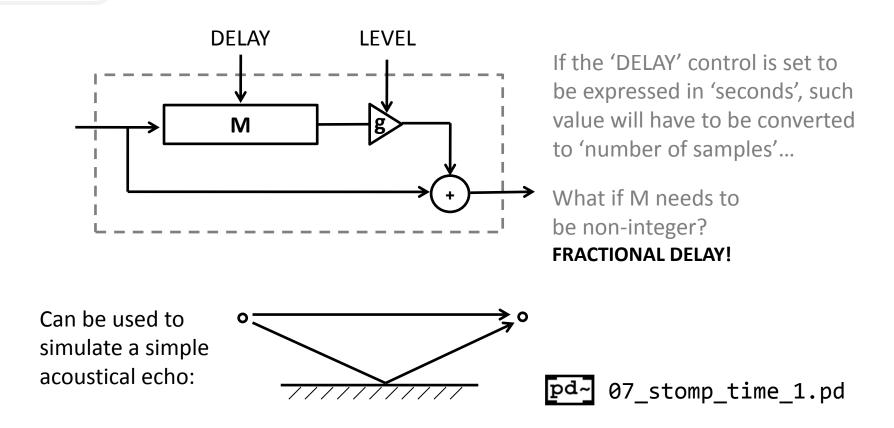
→ Usually implemented as fixed length buffer with write and read pointers 'spaced' M samples from each other:



Delay / Echo (ii)



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

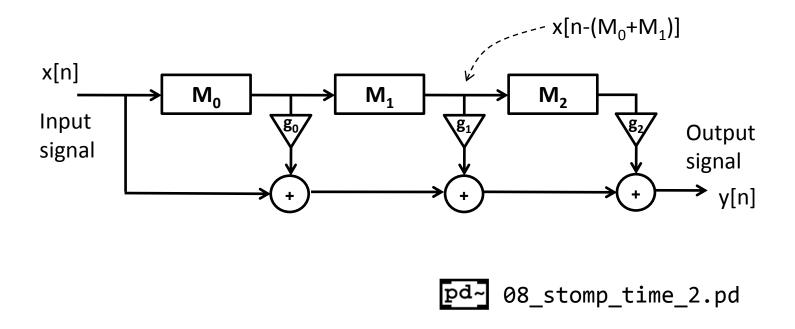


Delay / Echo (iii)



By cascading several delay lines,

one can obtain a **tapped delay** effect, which leads to a multiple echo:

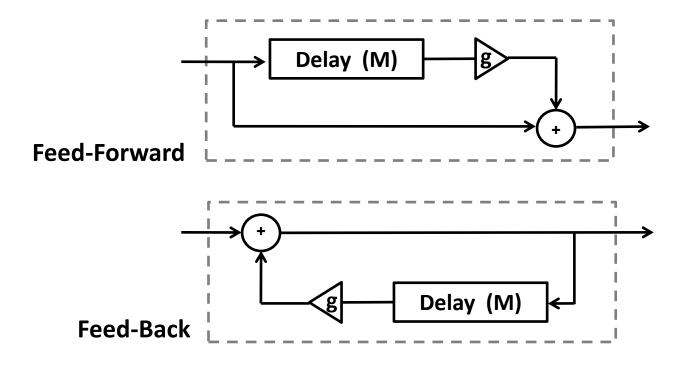


#### Before getting further with time-based effects: COMB FILTERS





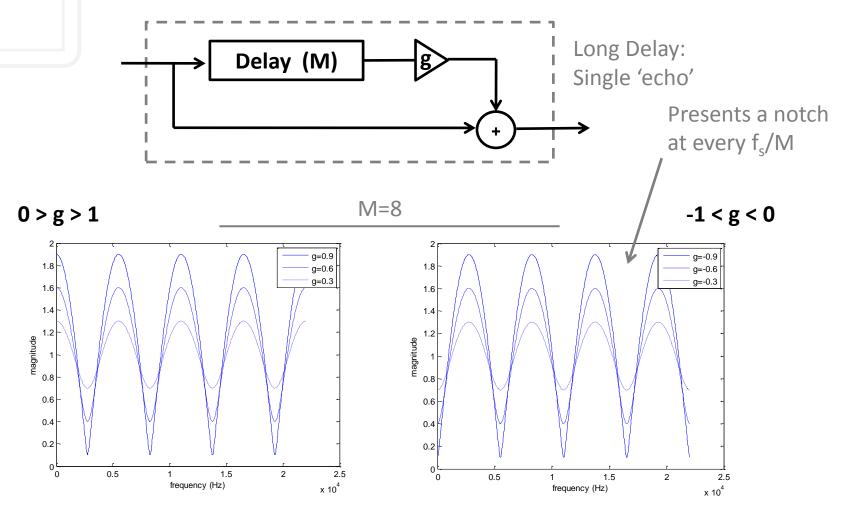
A comb filter adds a delayed version of a signal to itself. Ex: the single-echo effect presented before represents an instance of a comb filter.







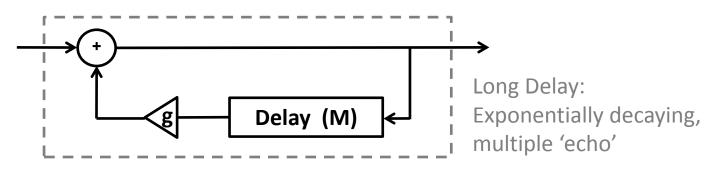
#### **Feed-Forward Comb Filter**

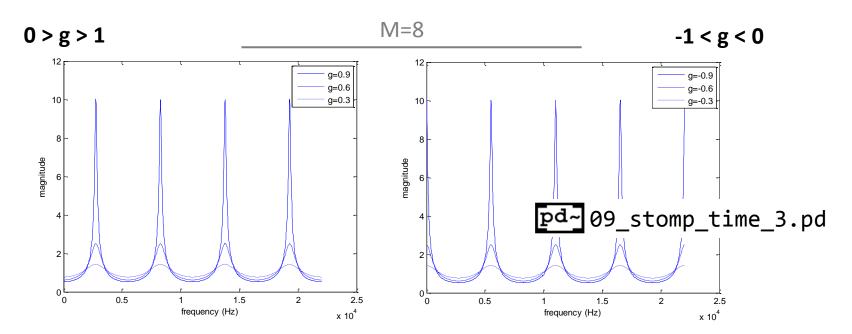




## **Comb Filters** (iii)

#### Feed-Back Comb Filter







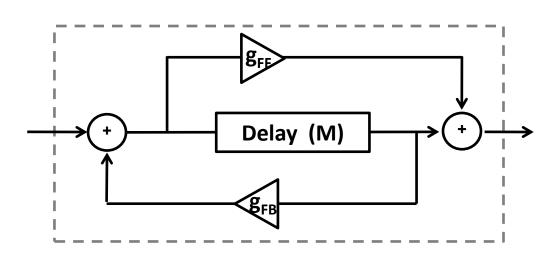
## **Comb Filters** (iv)

#### $g_{FF} < 0$ q = -0.9g=-0.6 magnitude g=-0.3 0.5 1.5 2 2.5 x 10<sup>4</sup> g=-0.9 g=-0.6 ngle (rad/s) g=-0.3 0.5 1.5 2 2.5 frequency (Hz) x 10<sup>4</sup> $g_{FF} > 0$ g=-0.9 g=-0.6 magnitude g=-0.3 0.5 1.5 2 2.5 x 10<sup>4</sup> g=-0.9 g=-0.6 ngle (rad/s) g=-0.3 0.5 1.5 2.5 0 2 1 frequency (Hz)

x 10<sup>4</sup>

#### **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  $\mathbf{g}_{FF} = -\mathbf{g}_{FB}$ .

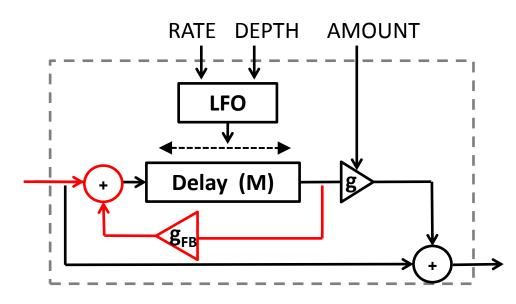


## Flanger



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**:



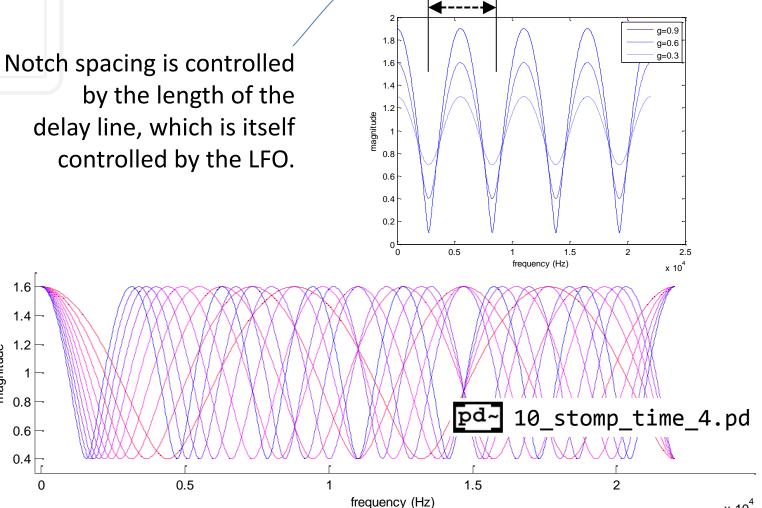
Harmonic series of notches in magnitude response; notches are uniformly spaced (at  $f_s/M$ ).

Sometimes, a Feedback control can be added.



## Flanger (ii)

magnitude

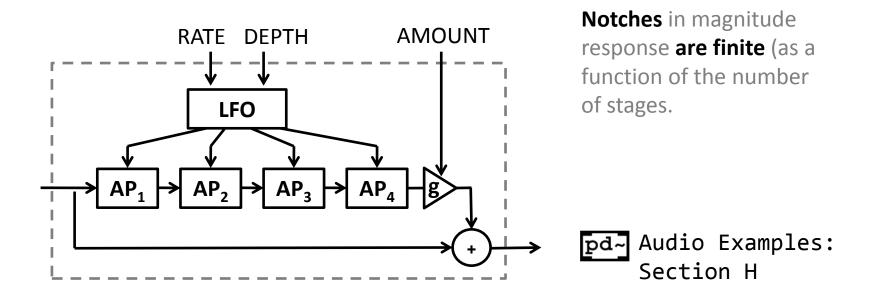


x 10<sup>4</sup>

## Phaser / Phase shifter

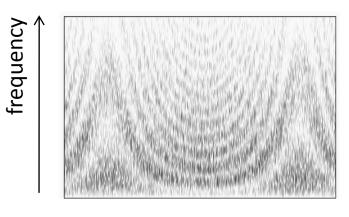


Closely related to the Flanger, it dates from the 1960s, too. Also based on slightly delaying a signal and adding it to itself, substitutes the variable delay line of the Flanger by a **cascade of low-order All-Pass filters**.



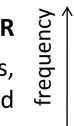


## Flanger vs Phaser



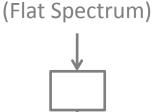
## FLANGER

Infinite series of notches, uniformly spaced



PHASER

Finite series of notches, arbitrarily located



White Noise

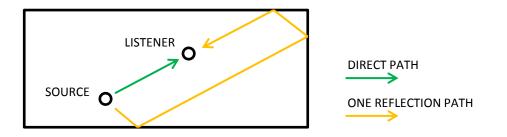
Filtered Noise

time





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



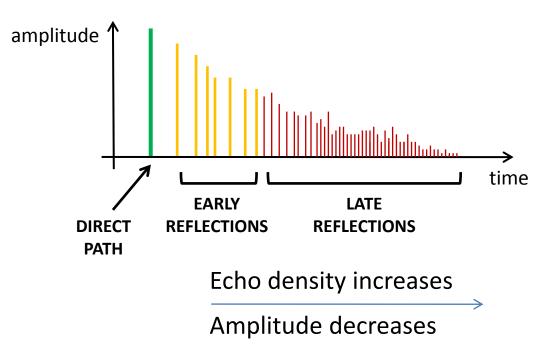
Artificial reverberation represents a very challenging problem, presenting a very **high computational cost** when modeled **from** a purely **physical perspective** (too many computations needed to simulate sound propagation in a 3D space).

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.

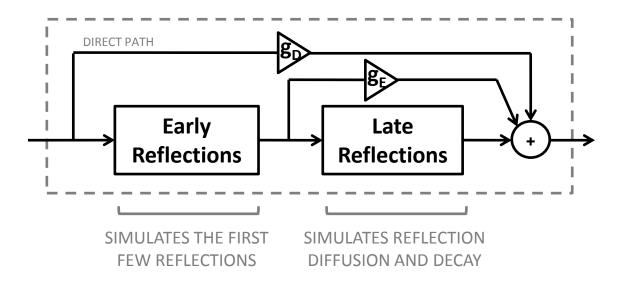
## Reverb (iii)



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:

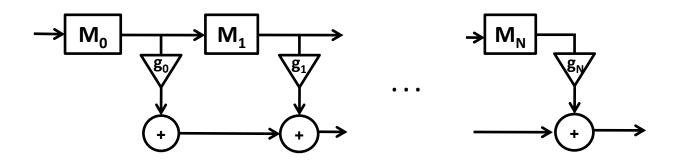






#### **EARLY REFLECTIONS**

One can use a tapped delay line line (one tap per reflection) with tuned delays  $M_n$  and gains  $g_n$ .



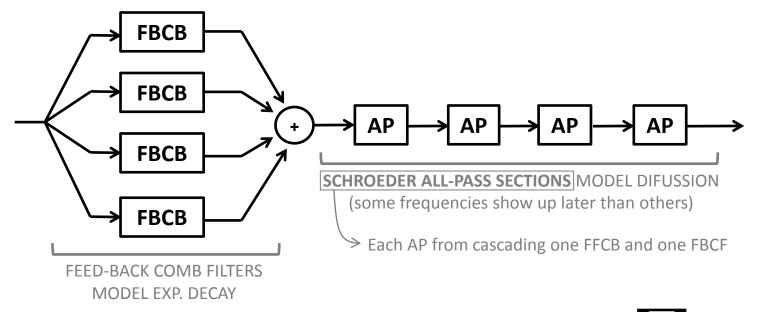
It is suggested that none of the taps' delay exceeds **40-50ms**, since it is acknowledged as the threshold for echo perception. An idea is to control delays and gains with a 'shared' parameter.





### **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.