Digital Audio Effects

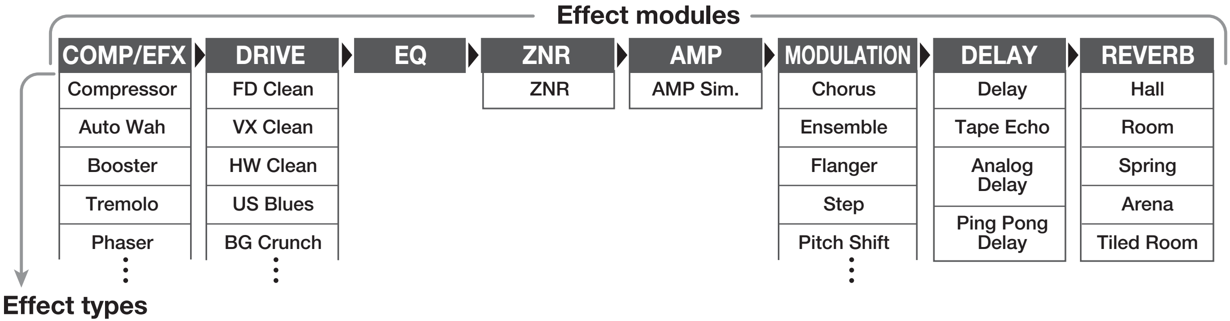
Having learned to make basic waveforms and basic filtering lets see how we can add some digital audio effects. These may be applied:

- As part of the audio creation/synthesis stage — to be subsequently filtered, (re)synthesised
- At the end of the *audio chain* — as part of the production/mastering phase.
- Effects can be applied in different orders and sometimes in a *parallel* audio chain.
- The order of applying the same effects can have drastic differences in the output audio.
- Selection of effects and the ordering is a matter for the sound you wish to create. There is no absolute rule for the ordering.

Typical Guitar (and other) Effects Pipeline

Some ordering is *standard* for some audio processing, *E.g*:
Compression → Distortion → EQ → Noise Redux → Amp Sim → Modulation → Delay → Reverb

Common for some guitar effects pedal:



Note: Other Effects Units allow for a completely **reconfigurable** effects pipeline. *E.g.* Boss GT-8

Classifying Effects

Audio effects can be classified by the way do their processing:

Basic Filtering — Lowpass, Highpass filter etc,, Equaliser

Time Varying Filters — Wah-wah, Phaser

Delays — Vibrato, Flanger, Chorus, Echo

Modulators — Ring modulation, Tremolo, Vibrato

Non-linear Processing — Compression, Limiters, Distortion, Exciters/Enhancers

Spacial Effects — Panning, Reverb, Surround Sound

Basic Digital Audio Filtering Effects: Equalisers

Filters by definition **remove/attenuate** audio from the spectrum above or below some cut-off frequency.

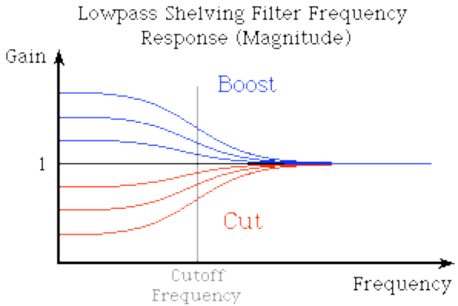
- For many audio applications this a little too restrictive

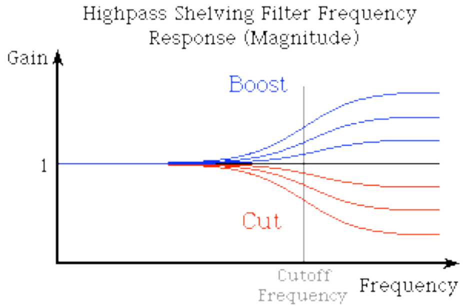
Equalisers, by contrast, **enhance/diminish** certain frequency bands whilst leaving others **unchanged**:

- Built using a series of *shelving* and *peak* filters
- First or second-order filters usually employed.

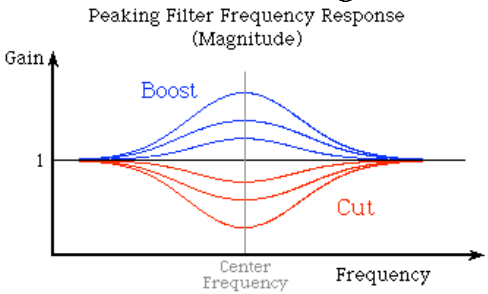
Shelving and Peak Filters

Shelving Filter — Boost or cut the low or high frequency bands with a cut-off frequency, F_c and gain G





Peak Filter — Boost or cut mid-frequency bands with a cut-off frequency, F_c, a bandwidth, f_b and gain G



Shelving Filters

A first-order shelving filter may be described by the transfer function:

$$H(z) = 1 + \frac{H_0}{2}(1 \pm A(z)) \quad \text{where } LF/HF +/-$$

where $A(z)$ is a first-order **allpass** filter — passes all frequencies but modifies phase:

$$A(z) = \frac{z^{-1} + a_{B/C}}{1 + a_{B/C} z^{-1}} \quad \text{B=Boost, C=Cut}$$

which leads the following algorithm/difference equation:

$$\begin{aligned} y_1(n) &= a_{B/C}x(n) + x(n-1) - a_{B/C}y_1(n-1) \\ y(n) &= \frac{H_0}{2}(x(n) \pm y_1(n)) + x(n) \end{aligned}$$

Shelving Filters (Cont.)

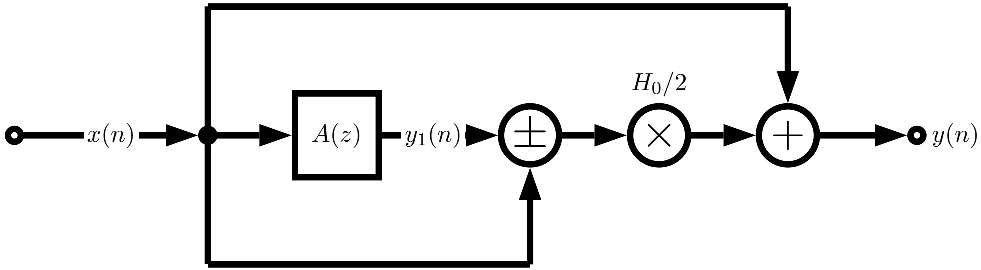
The gain, G, in dB can be adjusted accordingly:

$$H_0 = V_0 - 1 \;\text{ where } V_0 = 10^{G/20}$$

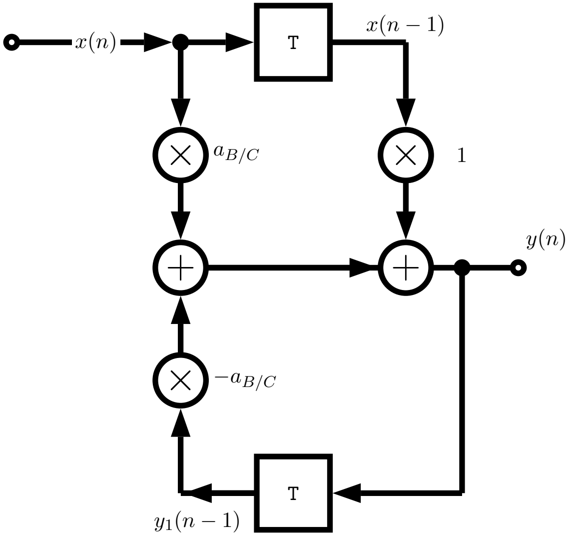
and the cut-off frequency for **boost**, a_B, or **cut**, a_C are given by:

$$\begin{aligned} a_B &= \frac{tan(2\pi f_c/f_s) - 1}{tan(2\pi f_c/f_s) + 1} \\ a_C &= \frac{tan(2\pi f_c/f_s) - V_0}{tan(2\pi f_c/f_s) - V_0} \end{aligned}$$

Shelving Filters Signal Flow Graph



where $A(z)$ is given by:



Peak Filters

A first-order shelving filter may be described by the transfer function:

$$H(z) = 1 + \frac{H_0}{2}(1 - A_2(z))$$

where $A_2(z)$ is a second-order **allpass** filter:

$$A(z) = \frac{-a_B + (d - da_B)z^{-1} + z^{-2}}{1 + (d - da_B)z^{-1} + a_B z^{-2}}$$

which leads the following algorithm/difference equation:

$$\begin{aligned} y_1(n) &= 1a_{B/C}x(n) + d(1 - a_{B/C})x(n-1) + x(n-2) \\ &\quad -d(1 - a_{B/C})y_1(n-1) + a_{B/C}y_1(n-2) \\ y(n) &= \frac{H_0}{2}(x(n) - y_1(n)) + x(n) \end{aligned}$$

Back
Close

Peak Filters (Cont.)

The center/cut-off frequency, d, is given by:

$$d = -cos(2\pi f_c/f_s)$$

The H_0 by relation to the gain, G, as before:

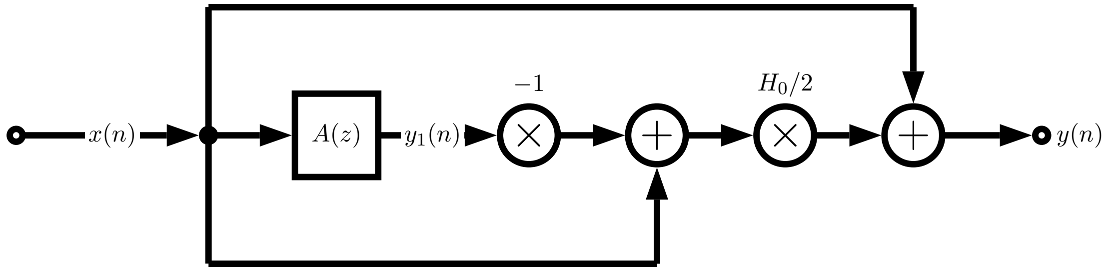
$$H_0 = V_0 - 1 \text{ where } V_0 = 10^{G/20}$$

and the bandwidth, f_b is given by the limits for **boost**, a_B, or **cut**, a_C are given by:

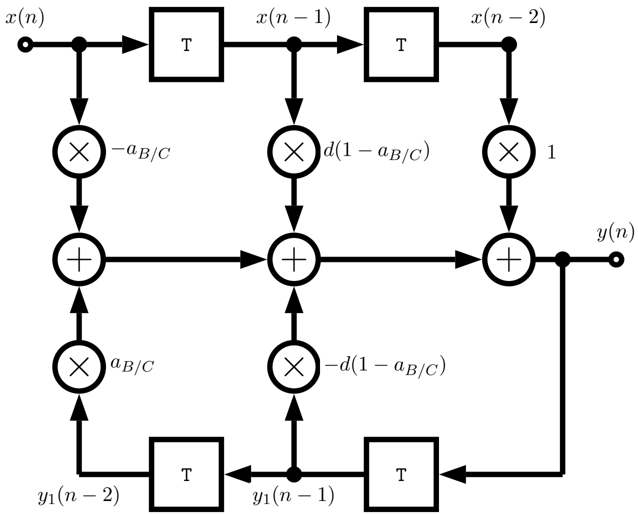
$$a_B = \frac{tan(2\pi f_b/f_s) - 1}{tan(2\pi f_b/f_s) + 1}$$

$$a_C = \frac{tan(2\pi f_b/f_s) - V_0}{tan(2\pi f_b/f_s) - V_0}$$

Peak Filters Signal Flow Graph



where $A(z)$ is given by:



Shelving Filter EQ MATLAB Example

The following function, shelving.m performs a shelving filter:

```
function [b, a]  = shelving(G, fc, fs, Q, type)
%
% Derive coefficients for a shelving filter with a given amplitude
% and cutoff frequency.  All coefficients are calculated as
% described in Zolzer's DAFX book (p. 50 -55).
%
% Usage:     [B,A] = shelving(G, Fc, Fs, Q, type);
%
%            G is the logrithmic gain (in dB)
%            FC is the center frequency
%            Fs is the sampling rate
%            Q adjusts the slope be replacing the sqrt(2) term
%            type is a character string defining filter type
%            Choices are: 'Base_Shelf' or 'Treble_Shelf'


%Error Check
if((strcmp(type,'Base_Shelf') ~= 1) && ...
        (strcmp(type,'Treble_Shelf') ~= 1))
    error(['Unsupported Filter Type: ' type]);
end
```

```
K = tan((pi * fc)/fs);
V0 = 10^(G/20);
root2 = 1/Q; %sqrt(2)

%Invert gain if a cut
if(V0 < 1)
    V0 = 1/V0;
end

%%%%%%%%%%%%%%%%%%%%
%    BASE BOOST
%%%%%%%%%%%%%%%%%%%%
if(( G > 0 ) & (strcmp(type,'Base_Shelf')))

    b0 = (1 + sqrt(V0)*root2*K + V0*K^2) / (1 + root2*K + K^2);
    b1 = (2 * (V0*K^2 - 1) ) / (1 + root2*K + K^2);
    b2 = (1 - sqrt(V0)*root2*K + V0*K^2) / (1 + root2*K + K^2);
    a1 =  (2 * (K^2 - 1) ) / (1 + root2*K + K^2);
    a2 =  (1 - root2*K + K^2) / (1 + root2*K + K^2);

%%%%%%%%%%%%%%%%%%%%
%    BASE CUT
%%%%%%%%%%%%%%%%%%%%
elseif (( G < 0 ) & (strcmp(type,'Base_Shelf')))

    b0 = (1 + root2*K + K^2) / (1 + root2*sqrt(V0)*K + V0*K^2);
```

```
    b1 =  (2 * (K^2 - 1) ) / (1 + root2*sqrt(V0)*K + V0*K^2);
    b2 =  (1 - root2*K + K^2) / (1 + root2*sqrt(V0)*K + V0*K^2);
    a1 =  (2 * (V0*K^2 - 1) ) / (1 + root2*sqrt(V0)*K + V0*K^2);
    a2 = (1 - root2*sqrt(V0)*K + V0*K^2) / ...
              (1 + root2*sqrt(V0)*K + V0*K^2);

%%%%%%%%%%%%%%%%%%%%
%   TREBLE BOOST
%%%%%%%%%%%%%%%%%%%%
elseif (( G > 0 ) & (strcmp(type,'Treble_Shelf')))

    b0 = (V0 + root2*sqrt(V0)*K + K^2) / (1 + root2*K + K^2);
    b1 =  (2 * (K^2 - V0) ) / (1 + root2*K + K^2);
    b2 = (V0 - root2*sqrt(V0)*K + K^2) / (1 + root2*K + K^2);
    a1 =  (2 * (K^2 - 1) ) / (1 + root2*K + K^2);
    a2 =  (1 - root2*K + K^2) / (1 + root2*K + K^2);

%%%%%%%%%%%%%%%%%%%%
%   TREBLE CUT
%%%%%%%%%%%%%%%%%%%%

elseif (( G < 0 ) & (strcmp(type,'Treble_Shelf')))

    b0 =   (1 + root2*K + K^2) / (V0 + root2*sqrt(V0)*K + K^2);
    b1 =    (2 * (K^2 - 1) ) / (V0 + root2*sqrt(V0)*K + K^2);
    b2 =    (1 - root2*K + K^2) / (V0 + root2*sqrt(V0)*K + K^2);
    a1 =    (2 * ((K^2)/V0 - 1) ) / (1 + root2/sqrt(V0)*K ...
```


Back
Close

```
                 + (K^2)/V0);
    a2 = (1 - root2/sqrt(V0)*K + (K^2)/V0) / ....
             (1 + root2/sqrt(V0)*K + (K^2)/V0);

%%%%%%%%%%%%%%%%%%%%
%   All-Pass
%%%%%%%%%%%%%%%%%%%%
else
    b0 = V0;
    b1 = 0;
    b2 = 0;
    a1 = 0;
    a2 = 0;
end

%return values
a = [  1, a1, a2];
b = [ b0, b1, b2];
```

Shelving Filter EQ MATLAB Example (Cont.)

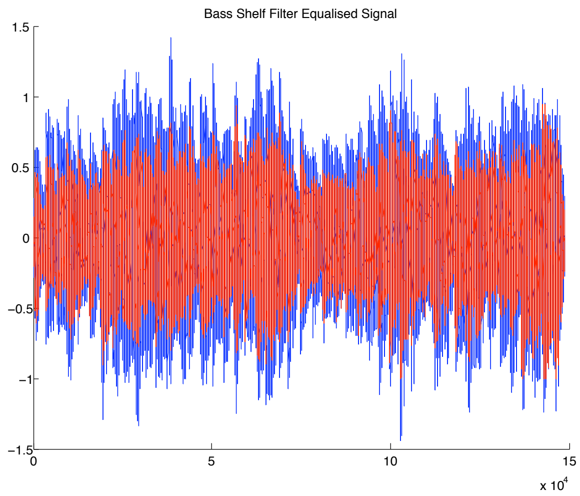
The following script shelving_eg.m illustrates how we use the shelving filter function to filter:

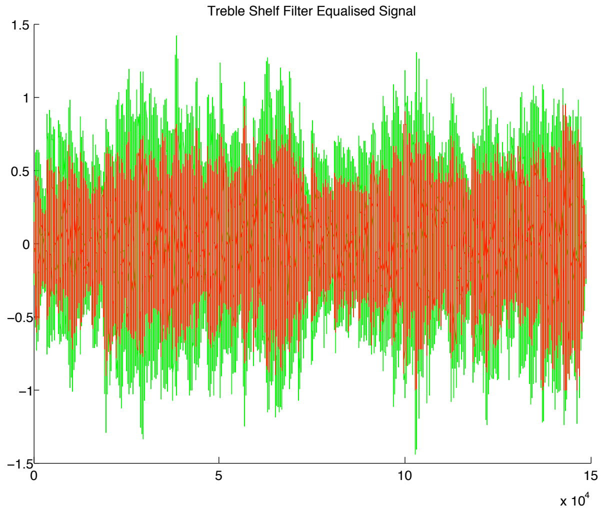
```
infile = 'acoustic.wav';

% read in wav sample
[ x, Fs, N ] = wavread(infile);

%set Parameters for Shelving Filter
% Change these to experiment with filter

G = 4; fcb = 300; Q = 3; type = 'Base_Shelf';

[b a] = shelving(G, fcb, Fs, Q, type);
yb = filter(b,a, x);

% write output wav files
wavwrite(yb, Fs, N, 'out_bassshelf.wav');

% plot the original and equalised waveforms
figure(1), hold on;
plot(yb,'b');
plot(x,'r');
title('Bass Shelf Filter Equalised Signal');
```

```
%Do treble shelf filter
fct = 600; type = 'Treble_Shelf';

[b a] = shelving(G, fct, Fs, Q, type);
yt = filter(b,a, x);

% write output wav files
wavwrite(yt, Fs, N, 'out_treblehelf.wav');

figure(1), hold on;
plot(yb,'g');
plot(x,'r');
title('Treble Shelf Filter Equalised Signal');
```

Shelving Filter EQ MATLAB Example (Cont.)

The output from the above code is (red plot is original audio):





Click here to hear: original audio, bass shelf filtered audio, treble shelf filtered audio.

Time-varying Filters

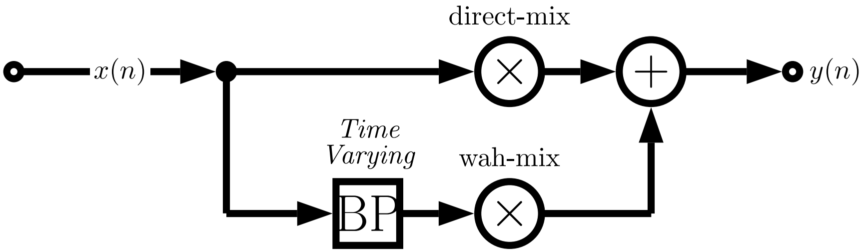
Some common effects are realised by simply time varying a filter in a couple of different ways:

Wah-wah — A bandpass filter with a time varying centre (resonant) frequency and a small bandwidth. Filtered signal mixed with direct signal.

Phasing — A notch filter, that can be realised as set of cascading IIR filters, again mixed with direct signal.

Wah-wah Example

The signal flow for a wah-wah is as follows:



where **BP** is a time varying frequency bandpass filter.

- A *phaser* is similarly implemented with a notch filter replacing the bandpass filter.
- A variation is the ***M*-fold wah-wah** filter where M tap delay bandpass filters spread over the entire spectrum change their centre frequencies simultaneously.
- A **bell effect** can be achieved with around a hundred M tap delays and narrow bandwidth filters

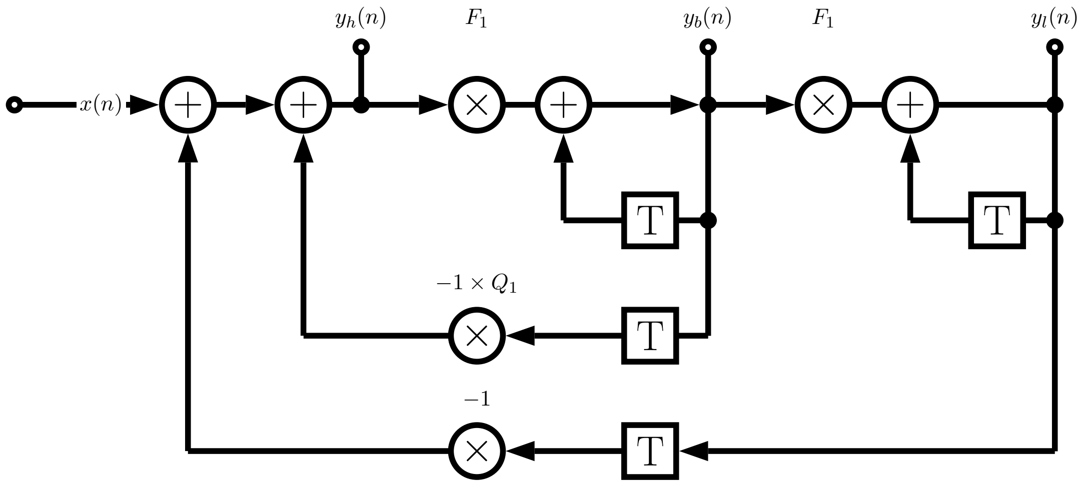
Time Varying Filter Implementation: State Variable Filter

In our audio application of time varying filters we now want independent control over the cut-off frequency and damping factor of a filter.

(Borrowed from analog electronics) we can implement a **State Variable Filter** to solve this problem.

- One further advantage is that we can **simultaneously** get lowpass, bandpass and highpass filter output.

The State Variable Filter



where:

$$
\begin{aligned}
x(n) &= \text{input signal} \\
y_l(n) &= \text{lowpass signal} \\
y_b(n) &= \text{bandpass signal} \\
y_h(n) &= \text{highpass signal}
\end{aligned}
$$

The State Variable Filter Algorithm

The algorithm difference equations are given by:

$$\begin{aligned}
y_l(n) &= F_1 y_b(n) + y_l(n-1) \\
y_b(n) &= F_1 y_h(n) + y_b(n-1) \\
y_h(n) &= x(n) - y_l(n-1) - Q_1 y_b(n-1)
\end{aligned}$$

with tuning coefficients F_1 and Q_1 related to the cut-off frequency, f_c, and damping, d:

$$F_1 = 2\sin(\pi f_c / f_s), \qquad \text{and} \quad Q_1 = 2d$$

MATLAB Wah-wah Implementation

We simply implement the State Variable Filter with a variable frequency, f_c. The code listing is wah_wah.m:

```
% wah_wah.m   state variable band pass
%
% BP filter with narrow pass band, Fc oscillates up and
% down the spectrum
% Difference equation taken from DAFX chapter 2
%
% Changing this from a BP to a BR/BS (notch instead of a bandpass) converts
%  this effect to a phaser
%
% yl(n) = F1*yb(n) + yl(n-1)
% yb(n) = F1*yh(n) + yb(n-1)
% yh(n) = x(n) - yl(n-1) - Q1*yb(n-1)
%
% vary Fc from 500 to 5000 Hz

infile = 'acoustic.wav';

% read in wav sample
[ x, Fs, N ] = wavread(infile);
```

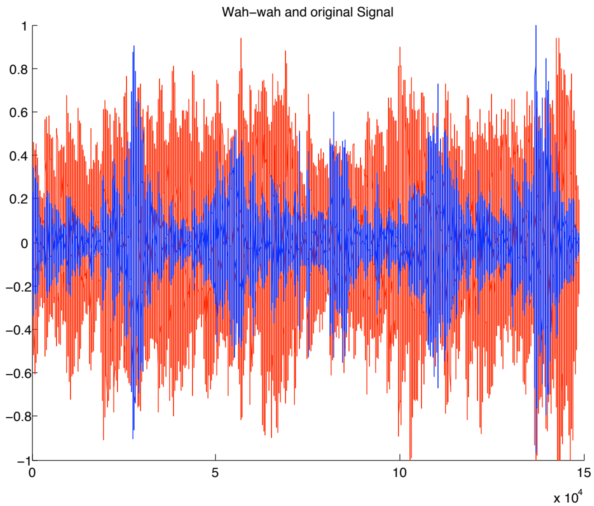
```
%%%%%%% EFFECT COEFFICIENTS %%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% damping factor
% lower the damping factor the smaller the pass band
damp = 0.05;

% min and max centre cutoff frequency of variable bandpass filter
minf=500;
maxf=3000;

% wah frequency, how many Hz per second are cycled through
Fw = 2000;
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


% change in centre frequency per sample (Hz)
delta = Fw/Fs;

% create triangle wave of centre frequency values
Fc=minf:delta:maxf;
while(length(Fc) < length(x) )
    Fc= [ Fc (maxf:-delta:minf) ];
    Fc= [ Fc (minf:delta:maxf) ];
end

% trim tri wave to size of input
Fc = Fc(1:length(x));
```

```
% difference equation coefficients
% must be recalculated each time Fc changes
F1 = 2*sin((pi*Fc(1))/Fs);
% this dictates size of the pass bands
Q1 = 2*damp;

yh=zeros(size(x));            % create empty out vectors
yb=zeros(size(x));
yl=zeros(size(x));

% first sample, to avoid referencing of negative signals
yh(1) = x(1);
yb(1) = F1*yh(1);
yl(1) = F1*yb(1);

% apply difference equation to the sample
for n=2:length(x),
    yh(n) = x(n) - yl(n-1) - Q1*yb(n-1);
    yb(n) = F1*yh(n) + yb(n-1);
    yl(n) = F1*yb(n) + yl(n-1);
    F1 = 2*sin((pi*Fc(n))/Fs);
end

%normalise
maxyb = max(abs(yb));
yb = yb/maxyb;
```

```
% write output wav files
wavwrite(yb, Fs, N, 'out_wah.wav');

figure(1)
hold on
plot(x,'r');
plot(yb,'b');
title('Wah-wah and original Signal');
```

Wah-wah MATLAB Example (Cont.)

The output from the above code is (red plot is original audio):



Click here to hear: original audio, wah-wah filtered audio.

Wah-wah Code Explained

Three main parts:

- Create a triangle wave to modulate the centre frequency of the bandpass filter.
- Implementation of state variable filter
- Repeated recalculation if centre frequency within the state variable filter loop.

Wah-wah Code Explained (Cont.)

Creation of triangle waveform we have seen previously— see waveforms.m.

- Slight modification of this code here to allow *'frequency values'* (Y-axis amplitude) to vary rather than frequency of the triangle waveform — here the frequency of the modulator wave is determined by wah-wah rate, `F_w`, usually a low frequency:

```
% min and max centre cutoff frequency of variable bandpass filter
minf=500; maxf=3000;
% wah frequency, how many Hz per second are cycled through
Fw = 2000;

% change in centre frequency per sample (Hz)
delta = Fw/Fs;
% create triangle wave of centre frequency values
Fc=minf:delta:maxf;
while(length(Fc) < length(x) )
    Fc= [ Fc (maxf:-delta:minf) ];
    Fc= [ Fc (minf:delta:maxf) ];
end
% trim tri wave to size of input
Fc = Fc(1:length(x));
```

Wah-wah Code Explained (Cont.)

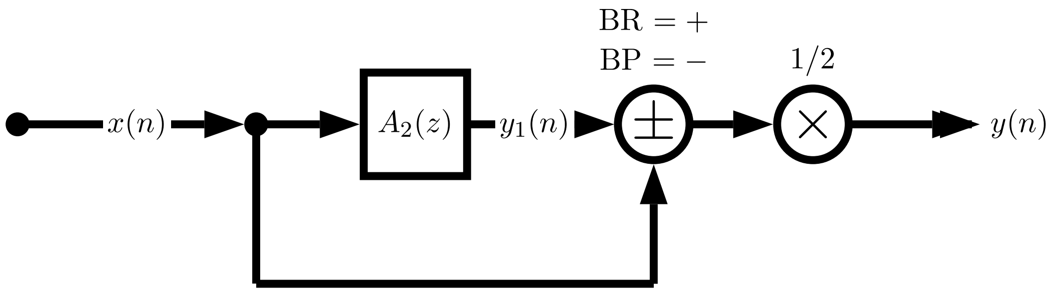
Note: As the Wah-wah rate is not likely to be in perfect sync with input waveform, `x`, we must trim it to the same `length` as `x`.

Modifications to Wah-wah

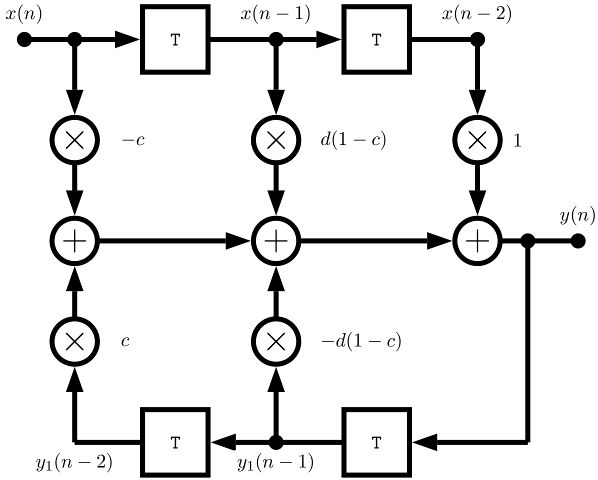
- Adding Multiple Delays with differing centre frequency filters but all modulated by same `Fc` gives an **M-fold wah-wah**
- Changing filter to a *notch* filter gives a **phaser**
  - **Notch Filter** (or bandreject/bandstop filter (BR/BS)) — attenuate frequencies in a narrow bandwidth (High Q factor) around cut-off frequency, u_0
- See Lab worksheet and useful for coursework

Bandreject (BR)/Bandpass(BP) Filters

(Sort of) Seen before (Peak Filter). Here we have, BR/BP:



where $A_2(z)$ (a second order allpass filter) is given by:



Bandreject (BR)/Bandpass(BP) Filters (Cont.)

The difference equation is given by:

$$
\begin{aligned}
y_1(n) &= -cx(n) + d(1-c)x(n-1) + x(n-2) \\
&\quad -d(1-c)y_1(n-1) + cy_1(n-2) \\
y(n) &= \frac{1}{2}(x(n) \pm y_1(n))
\end{aligned}
$$

where

$$d = -cos(2\pi f_c/f_s)$$

$$c = \frac{tan(2\pi f_c/f_s) - 1}{tan(2\pi f_c/f_s) + 1}$$

Bandreject $= +$
Bandpass $= -$

Delay Based Effects

Many useful audio effects can be implemented using a delay structure:

- Sounds reflected of walls
  - In a cave or large room we here an echo and also reverberation takes place – this is a different effect — **see later**
  - If walls are closer together repeated reflections can appear as parallel boundaries and we hear a modification of sound colour instead.
- Vibrato, Flanging, Chorus and Echo are examples of delay effects

Basic Delay Structure

We build basic delay structures out of some very basic FIR and IIR filters:

- We use *FIR* and *IIR comb filters*
- Combination of FIR and IIR gives the **Universal Comb Filter**

FIR Comb Filter

This simulates a single delay:

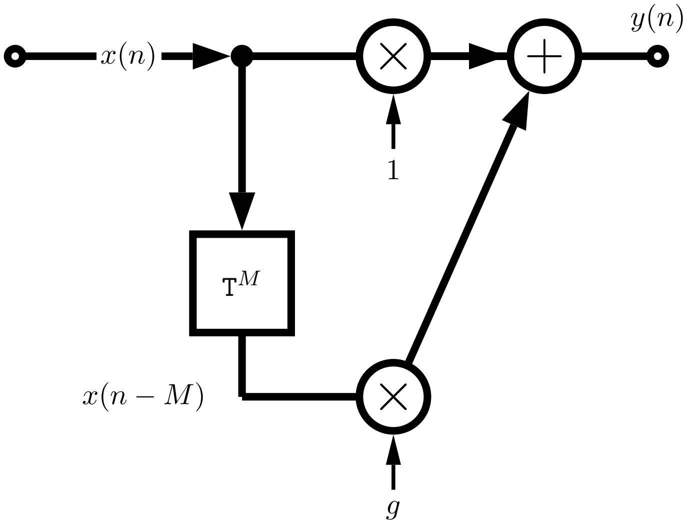
- The input signal is delayed by a given time duration, τ.
- The delayed (processed) signal is added to the input signal some amplitude gain, g
- The difference equation is simply:

$$y(n) = x(n) + gx(n - M) \quad \text{with} \quad M = \tau/f_s$$

- The transfer function is:

$$H(z) = 1 + gz^{-M}$$

FIR Comb Filter Signal Flow Diagram



FIR Comb Filter MATLAB Code

fircomb.m:

```
x=zeros(100,1);x(1)=1; % unit impulse signal of length 100

g=0.5; %Example gain

Delayline=zeros(10,1); % memory allocation for length 10

for n=1:length(x);
  y(n)=x(n)+g*Delayline(10);
  Delayline=[x(n);Delayline(1:10-1)];
end;
```

IIR Comb Filter

This simulates a single delay:

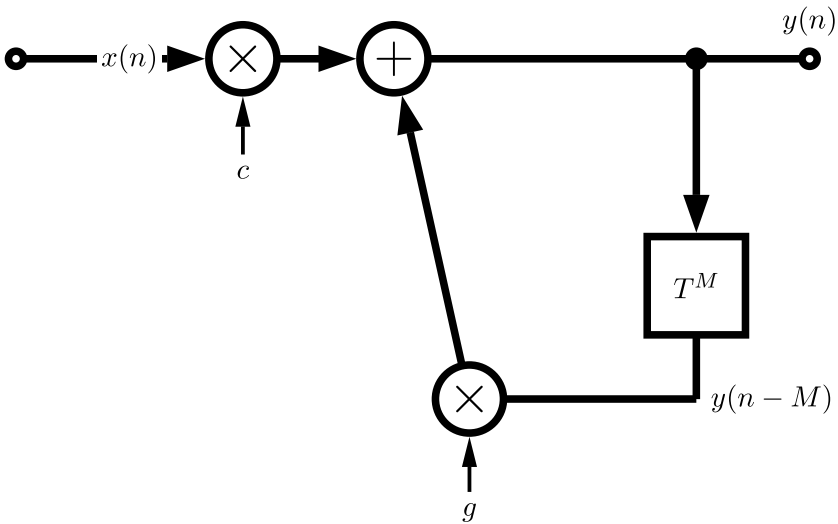
- Simulates *endless reflections* at both ends of cylinder.
- We get an endless series of responses, $y(n)$ to input, $x(n)$.
- The input signal circulates in delay line (delay time τ) that is fed back to the input..
- Each time it is fed back it is attenuated by g.
- Input sometime scaled by c to **compensate** for high amplification of the structure.
- The difference equation is simply:

$$y(n) = Cx(n) + gy(n - M) \quad \text{with} \quad M = \tau/f_s$$

- The transfer function is:

$$H(z) = \frac{c}{1 - gz^{-M}}$$

IIR Comb Filter Signal Flow Diagram



IIR Comb Filter MATLAB Code

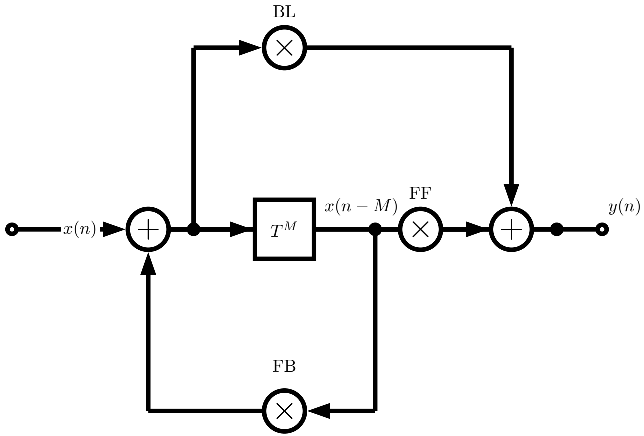
iircomb.m:

```
x=zeros(100,1);x(1)=1; % unit impulse signal of length 100

g=0.5;

Delayline=zeros(10,1); % memory allocation for length 10

for n=1:length(x);
   y(n)=x(n)+g*Delayline(10);
   Delayline=[y(n);Delayline(1:10-1)];
end;
```

Universal Comb Filter

The combination of the FIR and IIR comb filters yields the **Universal Comb Filter**:

- Basically this is an **allpass filter** with an M sample delay operator and an additional multiplier, FF.



- Parameters: FF = feedforward, FB = feedbackward, BL = blend

Universal Comb Filter Parameters

Universal in that we can form any comb filter, an allpass or a delay:

	BL	FB	FF
FIR Comb	1	0	g
IIR Comb	1	g	0
Allpass	a	$-a$	1
delay	0	0	1

Universal Comb Filter MATLAB Code

unicomb.m:

```
x=zeros(100,1);x(1)=1; % unit impulse signal of length 100

BL=0.5;
FB=-0.5;
FF=1;
M=10;

Delayline=zeros(M,1); % memory allocation for length 10

for n=1:length(x);
  xh=x(n)+FB*Delayline(M);
  y(n)=FF*Delayline(M)+BL*xh;
  Delayline=[xh;Delayline(1:M-1)];
end;
```

Vibrato - A Simple Delay Based Effect

- **Vibrato** — Varying the time delay periodically
- If we vary the distance between and observer and a sound source (*cf. Doppler effect*) we here a change in pitch.
- Implementation: A Delay line and a low frequency oscillator (LFO) to vary the delay.
- Only listen to the delay — no forward or backward feed.
- Typical delay time = 5–10 Ms and LFO rate 5–14Hz.

Vibrato MATLAB Code

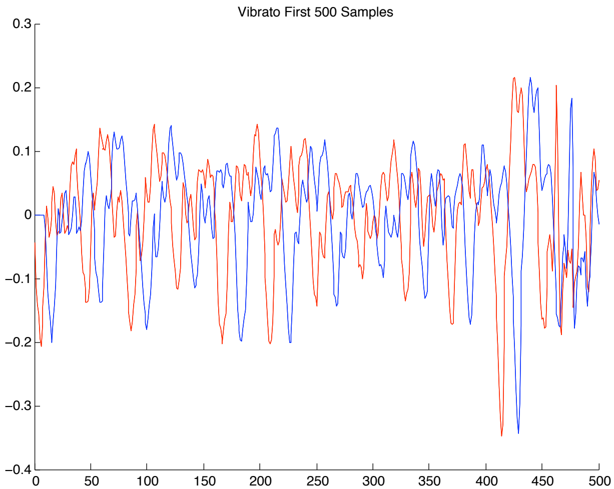
vibrato.m function, Use vibrato_eg.m to call function:

```
function y=vibrato(x,SAMPLERATE,Modfreq,Width)

ya_alt=0;
Delay=Width; % basic delay of input sample in sec
DELAY=round(Delay*SAMPLERATE); % basic delay in # samples
WIDTH=round(Width*SAMPLERATE); % modulation width in # samples
if WIDTH>DELAY
  error('delay greater than basic delay !!!');
  return;
end;

MODFREQ=Modfreq/SAMPLERATE; % modulation frequency in # samples
LEN=length(x);          % # of samples in WAV-file
L=2+DELAY+WIDTH*2;      % length of the entire delay
Delayline=zeros(L,1); % memory allocation for delay
y=zeros(size(x));       % memory allocation for output vector
```

```
for n=1:(LEN-1)
  M=MODFREQ;
  MOD=sin(M*2*pi*n);
  ZEIGER=1+DELAY+WIDTH*MOD;
  i=floor(ZEIGER);
  frac=ZEIGER-i;
  Delayline=[x(n);Delayline(1:L-1)];
  %---Linear Interpolation-----------------------------
  y(n,1)=Delayline(i+1)*frac+Delayline(i)*(1-frac);
  %---Allpass Interpolation------------------------------
  %y(n,1)=(Delayline(i+1)+(1-frac)*Delayline(i)-(1-frac)*ya_alt);
  %ya_alt=ya(n,1);
end
```

Vibrato MATLAB Example (Cont.)

The output from the above code is (red plot is original audio):



Click here to hear: original audio, vibrato audio.

Vibrato MATLAB Code Explained

Click here to hear: original audio, vibrato audio.

The code should be relatively self explanatory, except for one part:

- We work out the delay (modulated by a sinusoid) at each step, `n`:

```
M=MODFREQ;
MOD=sin(M*2*pi*n);
ZEIGER=1+DELAY+WIDTH*MOD;
```

- We then work out the nearest sample step: `i=floor(ZEIGER);`
- The **problem** is that we have a **fractional delay line** value: `ZEIGER`

```
ZEIGER = 11.2779
i = 11

ZEIGER = 11.9339
i = 11

ZEIGER = 12.2829
i = 12
```

Fractional Delay Line - Interpolation

- To improve effect we can use some form of interpolation to compute the output, $y(n)$.
  - Above uses Linear Interpolation

    ```
    y(n,1)=Delayline(i+1)*frac+Delayline(i)*(1-frac);
    ```

    or:

    $$y(n) = x(n-(M+1)).frac + x(n-M).(1-frac)$$

  - Alternatives (commented in code)

    ```
    %---Allpass Interpolation------------------------------
    %y(n,1)=(Delayline(i+1)+(1-frac)*Delayline(i)-...
                      (1-frac)*ya_alt);
    %ya_alt=y(n,1);
    ```

    or:

    $$\begin{aligned} y(n) &= x(n-(M+1)).frac + x(n-M).(1-frac) - \\ & \quad y(n-1).(1-frac) \end{aligned}$$

  - or spline based interpolation — see DAFX book p68-69.

Comb Filter Delay Effects: Flanger, Chorus, Slapback, Echo

- A few popular effects can be made with a comb filter (FIR or IIR) and some modulation
- Flanger, Chorus, Slapback, Echo same basic approach but *different sound* outputs:

Effect	Delay Range (ms)	Modulation
Resonator	$0 \ldots 20$	None
Flanger	$0 \ldots 15$	Sinusoidal ($\approx$ 1 Hz)
Chorus	$10 \ldots 25$	Random
Slapback	$25 \ldots 50$	None
Echo	> 50	None

- Slapback (or doubling) — quick repetition of the sound,
  Flanging — continuously varying LFO of delay,
  Chorus — **multiple copies** of sound delayed by small random delays

Flanger MATLAB Code

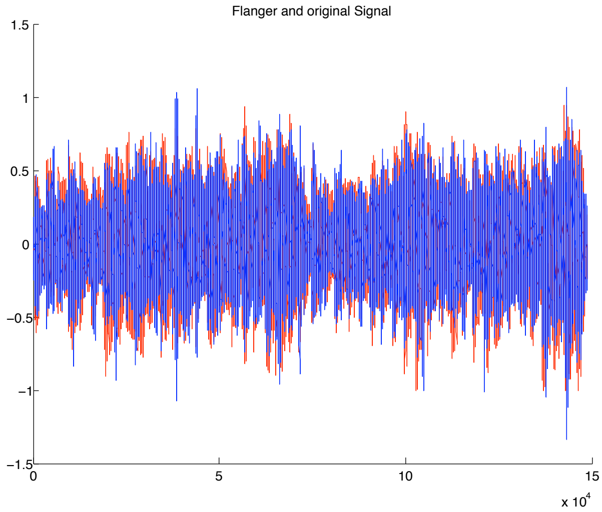
flanger.m:

```
% Creates a single FIR delay  with the delay time oscillating from
%  Either 0-3 ms or 0-15 ms at 0.1 - 5 Hz

 infile='acoustic.wav';
outfile='out_flanger.wav';

% read the sample waveform
[x,Fs,bits] = wavread(infile);

% parameters to vary the effect %
max_time_delay=0.003; % 3ms max delay in seconds
rate=1; %rate of flange in Hz

index=1:length(x);

% sin reference to create oscillating delay
sin_ref = (sin(2*pi*index*(rate/Fs)))';

%convert delay in ms to max delay in samples
max_samp_delay=round(max_time_delay*Fs);

% create empty out vector
y = zeros(length(x),1);
```

```
% to avoid referencing of negative samples
y(1:max_samp_delay)=x(1:max_samp_delay);

% set amp suggested coefficient from page 71 DAFX
amp=0.7;

% for each sample
for i = (max_samp_delay+1):length(x),
  cur_sin=abs(sin_ref(i));    %abs of current sin val 0-1
  % generate delay from 1-max_samp_delay and ensure whole number
  cur_delay=ceil(cur_sin*max_samp_delay);
  % add delayed sample
  y(i) = (amp*x(i)) + amp*(x(i-cur_delay));
end

% write output
wavwrite(y,Fs,outfile);
```

Flanger MATLAB Example (Cont.)

The output from the above code is (red plot is original audio):



Click here to hear: original audio, flanged audio.

Modulation

Modulation is the process where parameters of a sinusoidal signal (amplitude, frequency and phase) are modified or varied by an audio signal.

We have met some example effects that could be considered as a class of modulation already:

Amplitude Modulation — Wah-wah, Phaser

Phase Modulation — Vibrato, Chorus, Flanger

We will now introduce some Modulation effects.

Ring Modulation

Ring modulation (RM) is where the audio *modulator* signal, $x(n)$ is multiplied by a sine wave, $m(n)$, with a *carrier* frequency, f_c.

- This is very simple to implement digitally:

$$y(n) = x(n).m(n)$$

- Although audible result is easy to comprehend for simple signals things get more complicated for signals having numerous partials
- If the modulator is also a sine wave with frequency, f_x then one hears the sum and difference frequencies: $f_c + f_x$ and $f_c - f_x$, for example.
- When the input is *periodic* with at a fundamental frequency, f_0, then a spectrum with amplitude lines at frequencies $|kf_0 \pm f_c|$
- Used to create robotic speech effects on old sci-fi movies and can create some odd almost non-musical effects if not used with care. (Original speech)

MATLAB Ring Modulation

Two examples, a sine wave and an audio sample being modulated by a sine wave, ring_mod.m

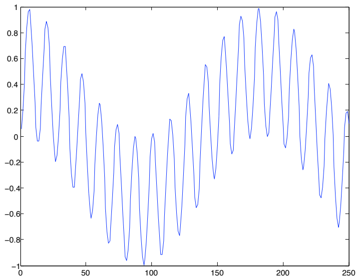
```
filename='acoustic.wav';

% read the sample waveform
[x,Fs,bits] = wavread(filename);

index = 1:length(x);

% Ring Modulate with a sine wave frequency Fc
Fc = 440;
carrier= sin(2*pi*index*(Fc/Fs))';

% Do Ring Modulation
y = x.*carrier;

% write output
wavwrite(y,Fs,bits,'out_ringmod.wav');
```

Click here to hear: original audio, ring modulated audio.

MATLAB Ring Modulation: Two sine waves

```
% Ring Modulate with a sine wave frequency Fc
Fc = 440;
carrier= sin(2*pi*index*(Fc/Fs))';

%create a modulator sine wave frequency Fx
Fx = 200;
modulator = sin(2*pi*index*(Fx/Fs))';

% Ring Modulate with sine wave, freq. Fc
y = modulator.*carrier;

% write output
wavwrite(y,Fs,bits,'twosine_ringmod.wav');
```

Output of Two sine wave ring modulation ($f_c = 440$, $f_x = 380$)



Click here to hear: Two RM sine waves ($f_c = 440$, $f_x = 200$)

Amplitude Modulation

Amplitude Modulation (AM) is defined by:

$$y(n) = (1 + \alpha m(n)).x(n)$$

- Normalise the peak amplitude of M(n) to 1.
- α is *depth of modulation*

  $\alpha = 1$ gives maximum modulation

  $\alpha = 0$ tuns off modulation
- $x(n)$ is the audio **carrier** signal
- $m(n)$ is a low-frequency oscillator **modulator**.
- When $x(n)$ and $m(n)$ both sine waves with frequencies f_c and f_x respectively we here **three** frequencies: carrier, difference and sum: f_c, $f_c - f_x$, $f_c + f_x$.

Amplitude Modulation: Tremolo

A common audio application of AM is to produce a **tremolo** effect:

- Set modulation frequency of the sine wave to below 20Hz

The MATLAB code to achieve this is tremolo1.m

```
% read the sample waveform
filename='acoustic.wav';
[x,Fs,bits] = wavread(filename);

index = 1:length(x);

Fc = 5;
alpha = 0.5;

trem=(1+ alpha*sin(2*pi*index*(Fc/Fs)))';
y = trem.*x;

% write output
wavwrite(y,Fs,bits,'out_tremolo1.wav');
```

Click here to hear: original audio, AM tremolo audio.

Tremolo via Ring Modulation

If you ring modulate with a triangular wave (or try another waveform) you can get tremolo via RM, tremolo2.m

```
% read the sample waveform
filename='acoustic.wav';
[x,Fs,bits] = wavread(filename);

% create triangular wave LFO
delta=5e-4;
minf=-0.5;
maxf=0.5;

trem=minf:delta:maxf;
while(length(trem) < length(x) )
   trem=[trem (maxf:-delta:minf)];
   trem=[trem (minf:delta:maxf)];
end

%trim trem
trem = trem(1:length(x))';

%Ring mod with triangular, trem
y= x.*trem;

% write output
wavwrite(y,Fs,bits,'out_tremolo2.wav');
```

Click here to hear: original audio, RM tremolo audio.

Non-linear Processing

Non-linear Processors are characterised by the fact that they create (intentional or unintentional) harmonic and inharmonic frequency components not present in the original signal.

Three major categories of non-linear processing:

Dynamic Processing: control of signal envelop — aim to minimise harmonic distortion Examples: Compressors, Limiters

Intentional non-linear harmonic processing: Aim to introduce strong harmonic distortion. Examples: Many electric guitar effects such as distortion

Exciters/Enhancers: add additional harmonics for subtle sound improvement.

Limiter

A **Limiter** is a device that controls high peaks in a signal but aims to change the dynamics of the main signal as little as possible:

- A limiter makes use of a peak level measurement and aims to react very quickly to **scale** the level if it is above some threshold.
- By lowering peaks the overall signal can be boosted.
- Limiting used not only on single instrument but on final (multichannel) audio for CD mastering, radio broadcast *etc.*

MATLAB Limiter Example

The following code creates a modulated sine wave and then limits the amplitude when it exceeds some threshold,The MATLAB code to achieve this is limiter.m:

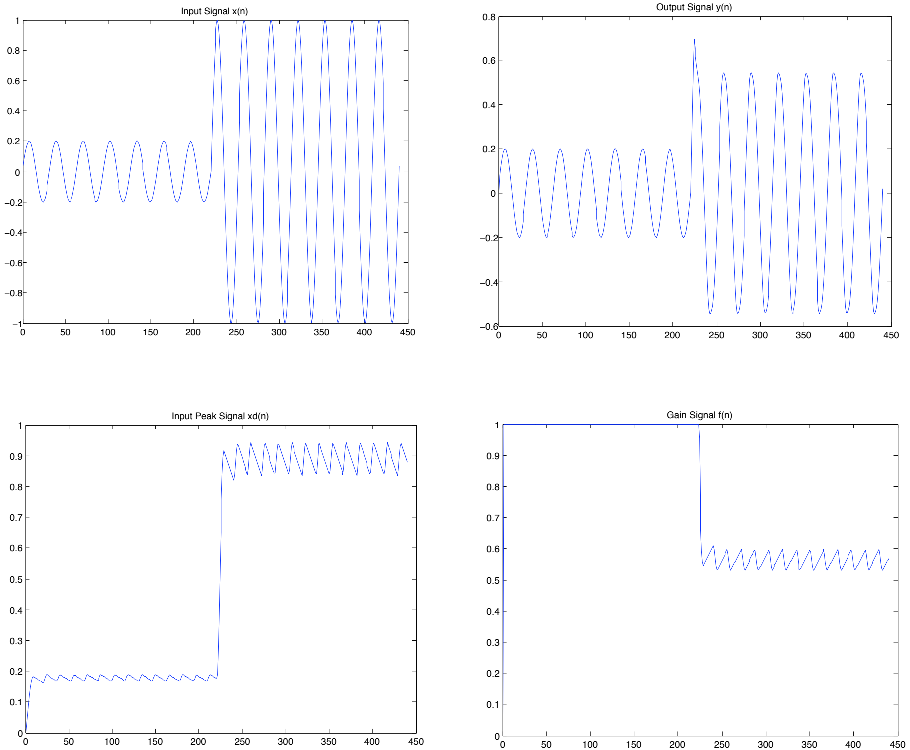
```
%Create a sine wave with amplitude
% reduced for half its duration

anzahl=220;
for n=1:anzahl,
   x(n)=0.2*sin(n/5);
end;
for n=anzahl+1:2*anzahl;
   x(n)=sin(n/5);
end;
```

MATLAB Limiter Example (Cont.)

```
% do Limiter

slope=1;
tresh=0.5;
rt=0.01;
at=0.4;

xd(1)=0; % Records Peaks in x
for n=2:2*anzahl;
  a=abs(x(n))-xd(n-1);
  if a<0, a=0; end;
    xd(n)=xd(n-1)*(1-rt)+at*a;
    if xd(n)>tresh,
      f(n)=10^(-slope*(log10(xd(n))-log10(tresh)));
      % linear calculation of f=10^(-LS*(X-LT))
    else f(n)=1;
  end;
  y(n)=x(n)*f(n);
end;
```

MATLAB Limiter Example (Cont.)

Display of the signals from the above limiter example:



Compressors/Expanders

Compressors are used to reduce the dynamics of the input signal:

- Quiet parts are **modified**
- Loud parts with are reduced according to some static curve.
- A bit like a limiter and uses again to boost overall signals in mastering or other applications.
- Used on vocals and guitar effects.

Expanders operate on low signal levels and boost the dynamics is these signals.

- Used to create a more lively sound characteristic

MATLAB Compressor/Expander

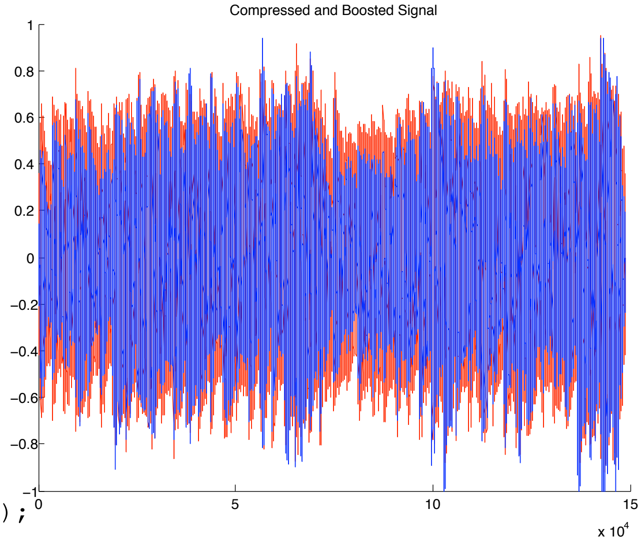
A MATLAB function for Compression/Expansion, compexp.m:

```
function y=compexp(x,comp,release,attack,a,Fs)
% Compressor/expander
% comp - compression: 0>comp>-1, expansion: 0<comp<1
% a     - filter parameter <1
h=filter([(1-a)^2],[1.0000 -2*a a^2],abs(x));
h=h/max(h);
h=h.^comp;
y=x.*h;
y=y*max(abs(x))/max(abs(y));
```

MATLAB Compressor/Expander (Cont.)

A **compressed signal** looks like this , compression_eg.m:

```
% read the sample waveform
filename='acoustic.wav';
[x,Fs,bits] = wavread(filename);

comp = -0.5; %set compressor
a = 0.5;
y = compexp(x,comp,a,Fs);

% write output
wavwrite(y,Fs,bits,...
    'out_compression.wav');

figure(1);
hold on
plot(y,'r');
plot(x,'b');
title('Compressed and Boosted Signal');
```

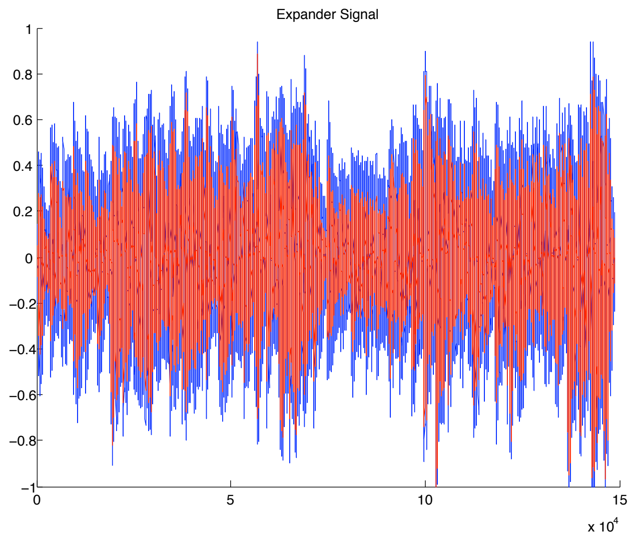


Click here to hear: original audio, compressed audio.

MATLAB Compressor/Expander (Cont.)

An **expanded signal** looks like this , expander_eg.m:

```
% read the sample waveform
filename='acoustic.wav';
[x,Fs,bits] = wavread(filename);

comp = 0.5; %set expander
a = 0.5;
y = compexp(x,comp,a,Fs);

% write output
wavwrite(y,Fs,bits,...
    'out_compression.wav');

figure(1);
hold on
plot(y,'r');
plot(x,'b');
title('Expander Signal');
```



Click here to hear: original audio, expander audio.

Overdrive, Distortion and Fuzz

Distortion plays an important part in electric guitar music, especially rock music and its variants.
Distortion can be applied as an effect to other instruments including vocals.

Overdrive — Audio at a low input level is driven by higher input levels in a non-linear curve characteristic

Distortion — a wider tonal area than overdrive operating at a higher non-linear region of a curve

Fuzz — complete non-linear behaviour, harder/harsher than distortion

Overdrive

For overdrive, **Symmetrical soft clipping** of input values has to be performed. A simple three layer *non-linear soft saturation* scheme may be:

$$f(x) = \begin{cases} 2x & \text{for } 0 \leq x < 1/3 \\ \frac{3-(2-3x)^2}{3} & \text{for } 1/3 \leq x < 2/3 \\ 1 & \text{for } 2/3 \leq x \leq 1 \end{cases}$$

- In the lower third the output is liner — multiplied by 2.
- In the middle third there is a non-linear (quadratic) output response
- Above 2/3 the output is set to 1.

MATLAB Overdrive Example

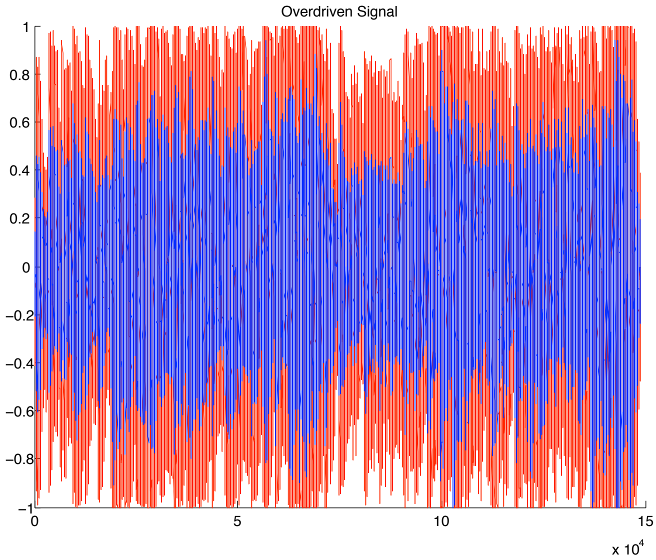
The MATLAB code to perform symmetrical soft clipping is, symclip.m:

```
function y=symclip(x)
% y=symclip(x)
% "Overdrive" simulation with symmetrical clipping
% x    - input
N=length(x);
y=zeros(1,N); % Preallocate y
th=1/3; % threshold for symmetrical soft clipping
        % by Schetzen Formula
for i=1:1:N,
   if abs(x(i))< th, y(i)=2*x(i);end;
   if abs(x(i))>=th,
     if x(i)> 0, y(i)=(3-(2-x(i)*3).^2)/3; end;
     if x(i)< 0, y(i)=-(3-(2-abs(x(i))*3).^2)/3; end;
   end;
   if abs(x(i))>2*th,
     if x(i)> 0, y(i)=1;end;
     if x(i)< 0, y(i)=-1;end;
   end;
end;
```

MATLAB Overdrive Example (Cont.)

An **overdriven signal** looks like this , overdrive_eg.m:

```
% read the sample waveform
filename='acoustic.wav';
[x,Fs,bits] = wavread(filename);

% call symmetrical soft clipping
% function
y = symclip(x);

% write output
wavwrite(y,Fs,bits,...
        'out_overdrive.wav');

figure(1);
hold on
plot(y,'r');
plot(x,'b');
title('Overdriven Signal');
```



Click here to hear: original audio, overdriven audio.

Distortion/Fuzz

A non-linear function commonly used to simulate distortion/fuzz is given by:

$$f(x) = \frac{x}{|x|}(1 - e^{\alpha x^2/|x|})$$

- This a non-linear exponential function:
- The gain, α, controls level of distortion/fuzz.
- Common to mix part of the distorted signal with original signal for output.

MATLAB Fuzz Example

The MATLAB code to perform non-linear gain is, fuzzexp.m:

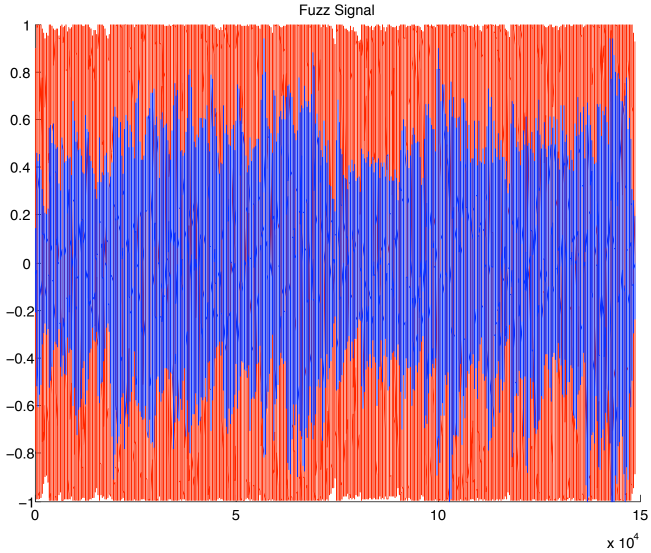
```
function y=fuzzexp(x, gain, mix)
% y=fuzzexp(x, gain, mix)
% Distortion based on an exponential function
% x     - input
% gain - amount of distortion, >0->
% mix  - mix of original and distorted sound, 1=only distorted
q=x*gain/max(abs(x));
z=sign(-q).*(1-exp(sign(-q).*q));
y=mix*z*max(abs(x))/max(abs(z))+(1-mix)*x;
y=y*max(abs(x))/max(abs(y));
```

Note: function allows to `mix` input and fuzz signals at output

MATLAB Fuzz Example (Cont.)

An **fuzzed up signal** looks like this , fuzz_eg.m:

```
filename='acoustic.wav';

% read the sample waveform
[x,Fs,bits] = wavread(filename);

% Call fuzzexp
gain = 11; % Spinal Tap it
mix = 1; % Hear only fuzz
y = fuzzexp(x,gain,mix);

% write output
wavwrite(y,Fs,bits,'out_fuzz.wav');
```



Click here to hear: original audio, Fuzz audio.

Reverb/Spatial Effects

The final set of effects we look at are effects that change to spatial localisation of sound. There a many examples of this type of processing we will study two briefly:

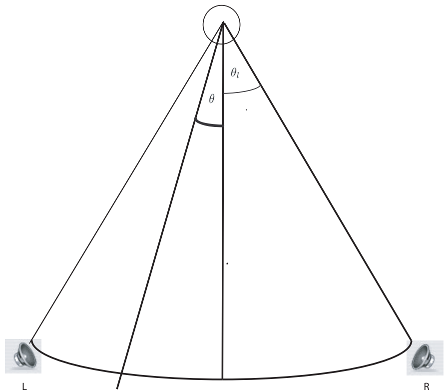
Panning in stereo audio

Reverb — a small selection of reverb algorithms

Panning

The simple problem we address here is mapping a monophonic sound source across a stereo audio image such that the sound starts in one speaker (R) and is moved to the other speaker (L) in n time steps.

- We assume that we listening in a central position so that the angle between two speakers is the same, i.e. we subtend an angle $2\theta_l$ between 2 speakers. We assume for simplicity, in this case that $\theta_l = 45°$



Panning Geometry

- We seek to obtain to signals one for each Left (L) and Right (R) channel, the gains of which, g_L and g_R, are applied to steer the sound across the stereo audio image.
- This can be achieved by simple 2D rotation, where the angle we sweep is θ:

$$\mathbf{A}_\theta = \begin{bmatrix} \cos\theta & \sin\theta \\ -\sin\theta & \cos\theta \end{bmatrix}$$

and

$$\begin{bmatrix} g_L \\ g_R \end{bmatrix} = \mathbf{A}_\theta.\mathbf{x}$$

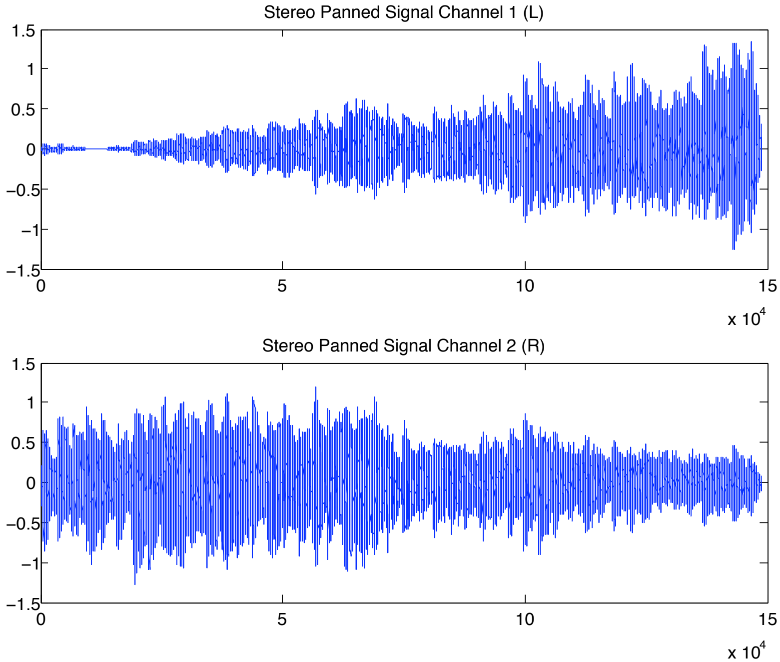
where $\mathbf{x}$ is a segment of mono audio

MATLAB Panning Example

The MATLAB code to do panning, matpan.m:

```
% read the sample waveform
filename='acoustic.wav';
[monox,Fs,bits] = wavread(filename);

initial_angle = -40; %in degrees
final_angle = 40;    %in degrees
segments = 32;
angle_increment = (initial_angle - final_angle)/segments * pi / 180;
                % in radians
lenseg = floor(length(monox)/segments) - 1;
pointer = 1;
angle = initial_angle * pi / 180; %in radians

y=[[];[]];

for i=1:segments
 A =[cos(angle), sin(angle); -sin(angle), cos(angle)];
 stereox = [monox(pointer:pointer+lenseg)'; monox(pointer:pointer+lenseg)'];
 y = [y, A * stereox];
 angle = angle + angle_increment; pointer = pointer + lenseg;
end;

% write output
wavwrite(y',Fs,bits,'out_stereopan.wav');
```

MATLAB Panning Example (Cont.)



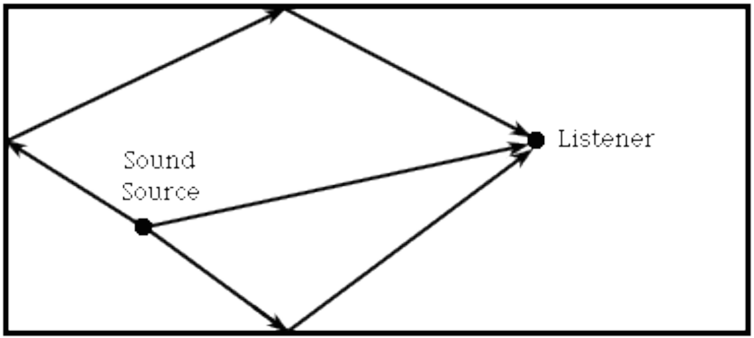
Click here to hear: original audio, stereo panned audio.

Reverb

Reverberation (**reverb** for short) is probably one of the most heavily used effects in music.

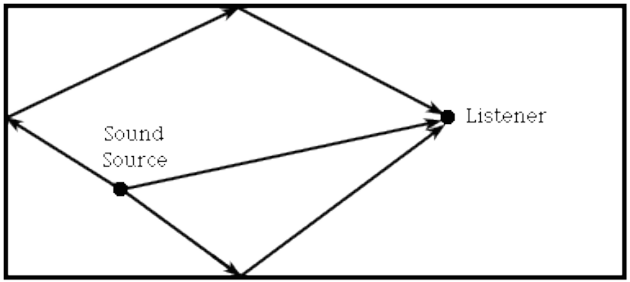
Reverberation is the result of the many reflections of a sound that occur in a room.

- From any sound source, say a speaker of your stereo, there is a direct path that the sounds covers to reach our ears.
- Sound waves can also take a slightly longer path by reflecting off a wall or the ceiling, before arriving at your ears.



The Spaciousness of a Room

- A reflected sound wave like this will arrive **a little later** than the direct sound, since it travels a longer distance, and is generally a little weaker, as the walls and other surfaces in the room will absorb some of the sound energy.
- Reflected waves can again bounce off another wall before arriving at your ears, and so on.
- This series of delayed and attenuated sound waves is what we call **reverb**, and this is what creates the *spaciousness* sound of a room.
- Clearly large rooms such as concert halls/cathedrals will have a much more spaciousness reverb than a living room or bathroom.



Reverb v. Echo

Is reverb just a series of echoes?

Echo — implies a distinct, delayed version of a sound,

- *E.g.* as you would hear with a delay more than one or two-tenths of a second.

Reverb — each delayed sound wave arrives in such a short period of time that we do not perceive each reflection as a copy of the original sound.

- Even though we can't discern every reflection, we still hear the effect that the entire series of reflections has.

Reverb v. Delay

Can a simple delay device with feedback produce reverberation?

Delay can produce a similar effect **but** there is one very important feature that a simple delay unit will not produce:

- The rate of arriving reflections changes over time
- Delay can only simulate reflections with a fixed time interval.

Reverb — for a short period after the direct sound, there is generally a set of well defined directional reflections that are directly related to the shape and size of the room, and the position of the source and listener in the room.

- These are the *early reflections*
- After the early reflections, the rate of the arriving reflections increases greatly are more random and difficult to relate to the physical characteristics of the room.
  This is called the *diffuse reverberation*, or the *late reflections*.
- Diffuse reverberation is the **primary factor** establishing a room's 'spaciousness' — it decays exponentially in good concert halls.

Reverb Simulations

There are many ways to simulate reverb.

We will only study two classes of approach here (there are others):

- Filter Bank/Delay Line methods
- Convolution/Impulse Response methods

Schroeder's Reverberator

- Early digital reverberation algorithms tried to mimic the a rooms reverberation by primarily using **two types** of infinite impulse response (IIR) filters.

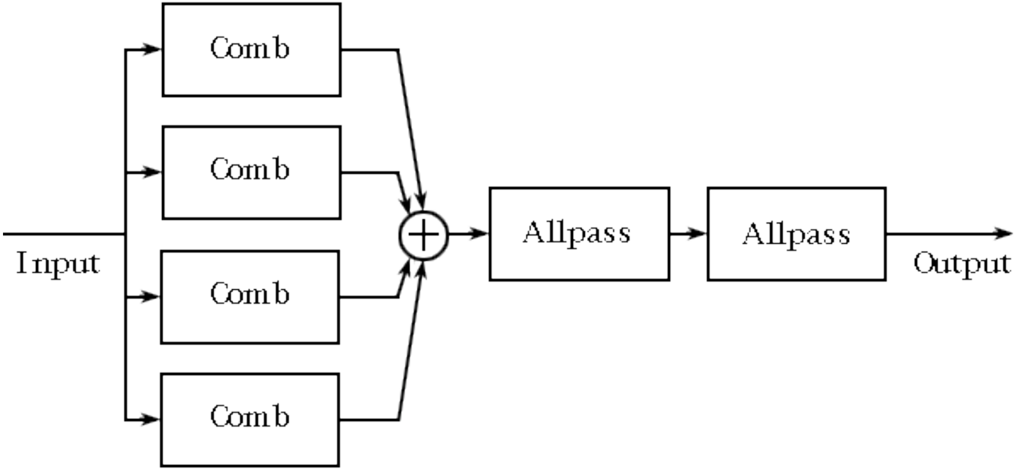
  Comb filter — usually in parallel banks

  Allpass filter — usually sequentially after comb filter banks

- A delay is (set via the feedback loops allpass filter) aims to make the output would gradually decay.

Schroeder's Reverberator (Cont.)

An example of one of Schroeder's well-known reverberator designs uses four comb filters and two allpass filters:



Note:This design does not create the increasing arrival rate of reflections, and is rather primitive when compared to current algorithms.

MATLAB Schroeder Reverb

The MATLAB function to do Schroeder Reverb, schroeder1.m:

```
function [y,b,a]=schroeder1(x,n,g,d,k)
%This is a reverberator based on Schroeder's design which consists of n all
%pass filters in series.
%
%The structure is:  [y,b,a] = schroeder1(x,n,g,d,k)
%
%where x = the input signal
%      n = the number of allpass filters
%      g = the gain of the allpass filters (should be less than 1 for stability)
%      d = a vector which contains the delay length of each allpass filter
%      k = the gain factor of the direct signal
%      y = the output signal
%      b = the numerator coefficients of the transfer function
%      a = the denominator coefficients of the transfer function
%
% note: Make sure that d is the same length as n.
%

% send the input signal through the first allpass filter
[y,b,a] = allpass(x,g,d(1));

% send the output of each allpass filter to the input of the next allpass filter
for i = 2:n,
   [y,b1,a1] = allpass(y,g,d(i));
   [b,a] = seriescoefficients(b1,a1,b,a);
end
```

```
% add the scaled direct signal
y = y + k*x;

% normalize the output signal
y = y/max(y);
```

The support files to do the filtering (for following reverb methods also) are here:

- delay.m,
- seriescoefficients.m,
- parallelcoefficients.m,
- fbcomb.m,
- ffcomb.m,
- allpass.m

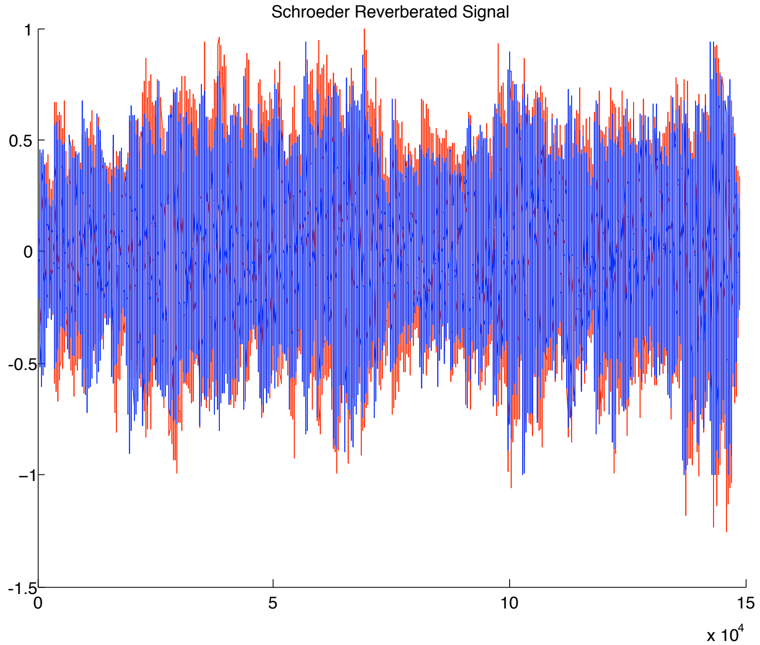
MATLAB Schroeder Reverb (Cont.)

An example script to call the function is as follows, reverb_schroeder_eg.m:

```
% reverb_Schroeder1_eg.m
% Script to call the Schroeder1 Reverb Algoritm

% read the sample waveform
filename='../acoustic.wav';
[x,Fs,bits] = wavread(filename);

% Call Schroeder1 reverb
%set the number of allpass filters
n = 6;
%set the gain of the allpass filters
g = 0.9;
%set delay of each allpass filter in number of samples
%Compute a random set of milliseconds and use sample rate
rand('state',sum(100*clock))
d = floor(0.05*rand([1,n])*Fs);
%set gain of direct signal
k= 0.2;

[y b a] = schroeder1(x,n,g,d,k);

% write output
wavwrite(y,Fs,bits,'out_schroederreverb.wav');
```

MATLAB Schroeder Reverb (Cont.)

The input signal (blue) and reverberated signal (red) look like this:



Click here to hear: original audio, Schroeder reverberated audio.

MATLAB Schroeder Reverb (Cont.)

The MATLAB function to do the more classic 4 comb and 2 allpass filter Schroeder Reverb, schroeder2.m:

```
function [y,b,a]=schroeder2(x,cg,cd,ag,ad,k)
%This is a reverberator based on Schroeder's design which consists of 4
% parallel feedback comb filters in series with 2 cascaded all pass filters.
%
%The structure is:  [y,b,a] = schroeder2(x,cg,cd,ag,ad,k)
%
%where x = the input signal
%       cg = a vector of length 4 which contains the gain of each of the
%               comb filters (should be less than 1 for stability)
%       cd = a vector of length 4 which contains the delay of each of the
%              comb filters
%       ag = the gain of the allpass filters (should be less than 1 for stability)
%       ad = a vector of length 2 which contains the delay of each of the
%               allpass filters
%       k = the gain factor of the direct signal
%       y = the output signal
%       b = the numerator coefficients of the transfer function
%       a = the denominator coefficients of the transfer function
%

% send the input to each of the 4 comb filters separately
[outcomb1,b1,a1] = fbcomb(x,cg(1),cd(1));
[outcomb2,b2,a2] = fbcomb(x,cg(2),cd(2));
[outcomb3,b3,a3] = fbcomb(x,cg(3),cd(3));
[outcomb4,b4,a4] = fbcomb(x,cg(4),cd(4));
```

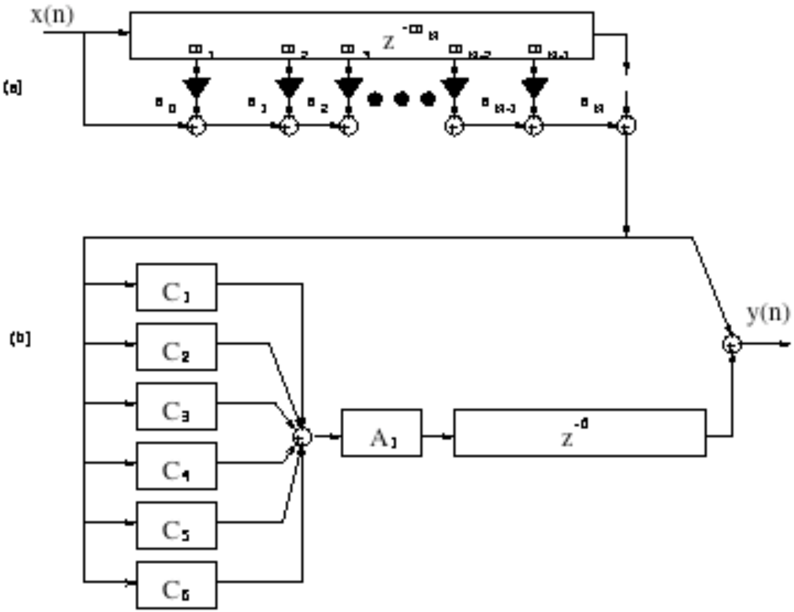
```
% sum the ouptut of the 4 comb filters
apinput = outcomb1 + outcomb2 + outcomb3 + outcomb4;

%find the combined filter coefficients of the the comb filters
[b,a]=parallelcoefficients(b1,a1,b2,a2);
[b,a]=parallelcoefficients(b,a,b3,a3);
[b,a]=parallelcoefficients(b,a,b4,a4);

% send the output of the comb filters to the allpass filters
[y,b5,a5] = allpass(apinput,ag,ad(1));
[y,b6,a6] = allpass(y,ag,ad(2));

%find the combined filter coefficients of the the comb filters in
% series with the allpass filters
[b,a]=seriescoefficients(b,a,b5,a5);
[b,a]=seriescoefficients(b,a,b6,a6);

% add the scaled direct signal
y = y + k*x;

% normalize the output signal
y = y/max(y);
```

Moorer's Reverberator

Moorer's reverberator build's on Schroeder:

- Parallel comb filters with different delay lengths are used to simulate modes of a room, and sound reflecting between parallel walls
- Allpass filters to increase the reflection density (diffusion).
- Lowpass filters inserted in the feedback loops to alter the reverberation time as a function of frequency
  - Shorter reverberation time at higher frequencies is caused by air absorption and reflectivity characteristics of wall).
  - Implement a dc-attenuation, and a frequency dependent attenuation.
  - Different in each comb filter because their coefficients depend on the delay line length

Moorer's Reverberator



(a) Tapped delay lines simulate *early reflections* —- forwarded to (b)

(b) Parallel comb filters which are then allpass filtered and delayed before being added back to early reflections — simulates *diffuse reverberation*

MATLAB Moorer Reverb

The MATLAB function to do Moorer' Reverb, moorer.m:

```
function [y,b,a]=moorer(x,cg,cg1,cd,ag,ad,k)
%This is a reverberator based on Moorer's design which consists of 6
% parallel feedback comb filters  (each with a low pass filter in the
% feedback loop) in series with an all pass filter.
%
%The structure is:  [y,b,a] = moorer(x,cg,cg1,cd,ag,ad,k)
%
%where x = the input signal
%      cg = a vector of length 6 which contains g2/(1-g1) (this should be less
%              than 1 for stability), where g2 is the feedback gain of each of the
%              comb filters and g1 is from the following parameter
%      cg1 = a vector of length 6 which contains the gain of the low pass
%              filters in the feedback loop of each of the comb filters (should be
%              less than 1 for stability)
%      cd = a vector of length 6 which contains the delay of each of comb filter
%      ag = the gain of the allpass filter (should be less than 1 for stability)
%      ad = the delay of the allpass filter
%      k = the gain factor of the direct signal
%      y = the output signal
%      b = the numerator coefficients of the transfer function
%      a = the denominator coefficients of the transfer function
%
```

MATLAB Moorer Reverb (Cont.)

```
% send the input to each of the 6 comb filters separately
[outcomb1,b1,a1] = lpcomb(x,cg(1),cg1(1),cd(1));
[outcomb2,b2,a2] = lpcomb(x,cg(2),cg1(2),cd(2));
[outcomb3,b3,a3] = lpcomb(x,cg(3),cg1(3),cd(3));
[outcomb4,b4,a4] = lpcomb(x,cg(4),cg1(4),cd(4));
[outcomb5,b5,a5] = lpcomb(x,cg(5),cg1(5),cd(5));
[outcomb6,b6,a6] = lpcomb(x,cg(6),cg1(6),cd(6));

% sum the ouptut of the 6 comb filters
apinput = outcomb1 + outcomb2 + outcomb3 + outcomb4 + outcomb5 + outcomb6;

%find the combined filter coefficients of the the comb filters
[b,a]=parallelcoefficients(b1,a1,b2,a2);
[b,a]=parallelcoefficients(b,a,b3,a3);
[b,a]=parallelcoefficients(b,a,b4,a4);
[b,a]=parallelcoefficients(b,a,b5,a5);
[b,a]=parallelcoefficients(b,a,b6,a6);

% send the output of the comb filters to the allpass filter
[y,b7,a7] = allpass(apinput,ag,ad);

%find the combined filter coefficients of the the comb filters in series
% with the allpass filters
[b,a]=seriescoefficients(b,a,b7,a7);

% add the scaled direct signal
y = y + k*x;

% normalize the output signal
y = y/max(y);
```

MATLAB Moorer Reverb (Cont.)

An example script to call the function is as follows, reverb_moorer_eg.m:

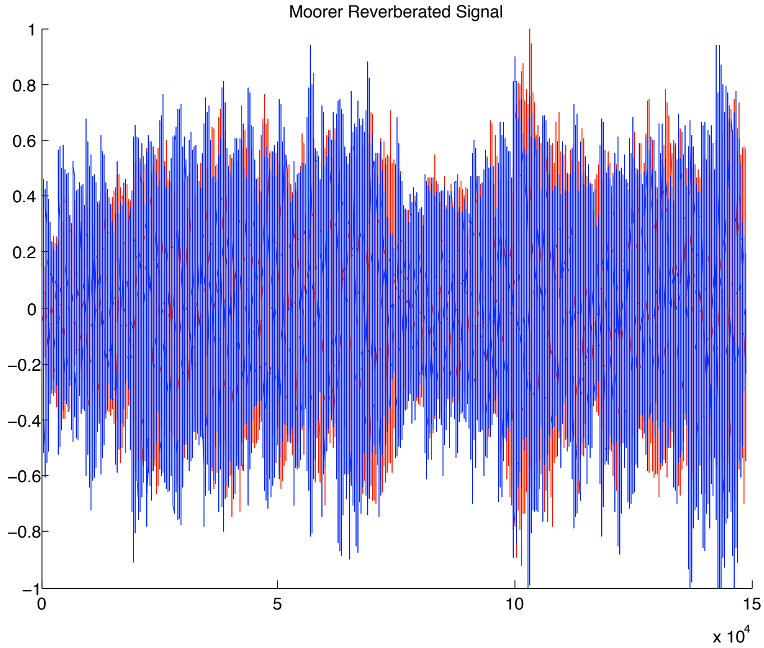
```
% reverb_moorer_eg.m
% Script to call the Moorer Reverb Algoritm

% read the sample waveform
filename='../acoustic.wav';
[x,Fs,bits] = wavread(filename);

% Call moorer reverb
%set delay of each comb filter
%set delay of each allpass filter in number of samples
%Compute a random set of milliseconds and use sample rate
rand('state',sum(100*clock))
cd = floor(0.05*rand([1,6])*Fs);

% set gains of 6 comb pass filters
g1 = 0.5*ones(1,6);
%set feedback of each comb filter
g2 = 0.5*ones(1,6);
% set input cg and cg1 for moorer function see help moorer
cg = g2./(1-g1);
cg1 = g1;
```

MATLAB Moorer Reverb (Cont.)

```
%set gain of allpass filter
ag = 0.7;
%set delay of allpass filter
ad = 0.08*Fs;
%set direct signal gain
k = 0.5;

[y b a] = moorer(x,cg,cg1,cd,ag,ad,k);

% write output
wavwrite(y,Fs,bits,'out_moorerreverb.wav');
```

MATLAB Moorer Reverb (Cont.)

The input signal (blue) and reverberated signal (red) look like this:



Click here to hear: original audio, Moorer reverberated audio.

Convolution Reverb

If the impulse response of the room is known then the most faithful reverberation method would be to **convolve** it with the input signal.

- Due usual length of the target response it is not feasible to implement this with filters — several hundreds of taps in the filters would be required.
- However, convolution readily implemented using FFT:
  - Recall: The **discrete convolution** formula:

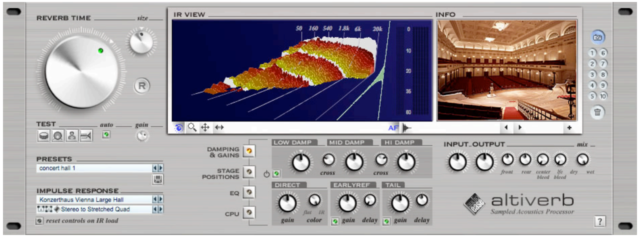
$$y(n) = \sum_{k=-\infty}^{\infty} x(k).h(n-k) = x(n) * h(n)$$

  - Recall: The **convolution theorem** which states that:
    If $f(x)$ and $g(x)$ are two functions with Fourier transforms $F(u)$ and $G(u)$, then the Fourier transform of the convolution $f(x)*g(x)$ is simply the product of the Fourier transforms of the two functions, $F(u)G(u)$.

Commercial Convolution Reverbs

Commercial examples:

- Altiverb — one of the first mainstream convolution reverb effects units
- Most sample based synthesisers (E.g. Kontakt, Intakt) provide some convolution reverb effect
- Dedicated sample-based software instruments such as Garritan Violin and PianoTeq Piano use convolution not only for reverb simulation but also to simulate key responses of the instruments body vibration.



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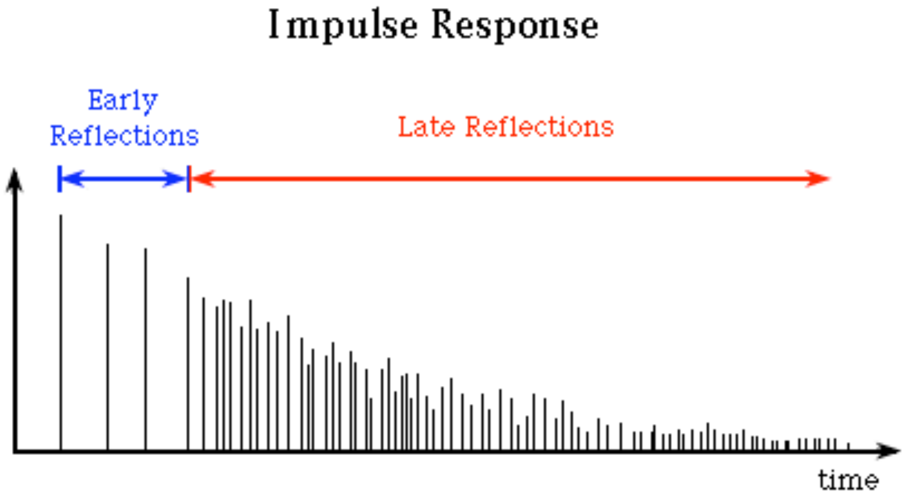
Room Impulse Responses

Apart from providing a high (professional) quality recording of a room's impulse response, the process of using an impulse response is quite straightforward:

- Record a short impulse (hand clap,drum hit) in the room.
- Room impulse responses can be simulated in software also.
- The impulse encodes the rooms reverb characteristics:



MATLAB Convolution Reverb

Let's develop a fast convolution routine, fconv.m:

```
function [y]=fconv(x, h)
%FCONV Fast Convolution
%   [y] = FCONV(x, h) convolves x and h, and normalizes the output
%         to +-1.
%       x = input vector
%       h = input vector
%

Ly=length(x)+length(h)-1;  %
Ly2=pow2(nextpow2(Ly));    % Find smallest power of 2 that is > Ly
X=fft(x, Ly2);     % Fast Fourier transform
H=fft(h, Ly2);             % Fast Fourier transform
Y=X.*H;                    % DO CONVOLUTION
y=real(ifft(Y, Ly2));      % Inverse fast Fourier transform
y=y(1:1:Ly);               % Take just the first N elements
y=y/max(abs(y));           % Normalize the output
```

See also: MATLAB built in function `conv()`

MATLAB Convolution Reverb (Cont.)

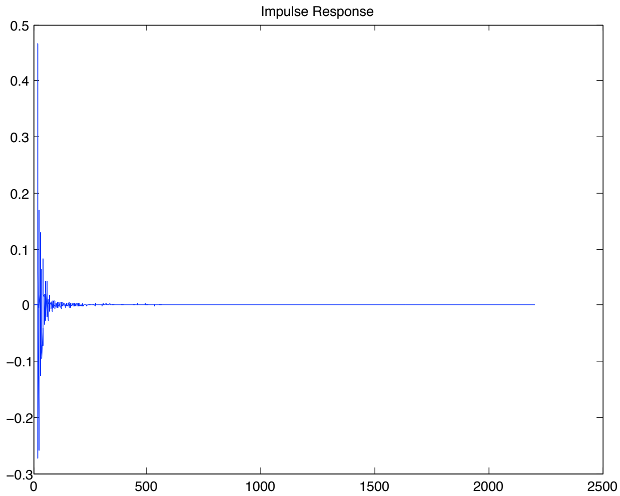
An example of how we call this function given an input signal and suitable impulse response, reverb_convolution_eg.m:

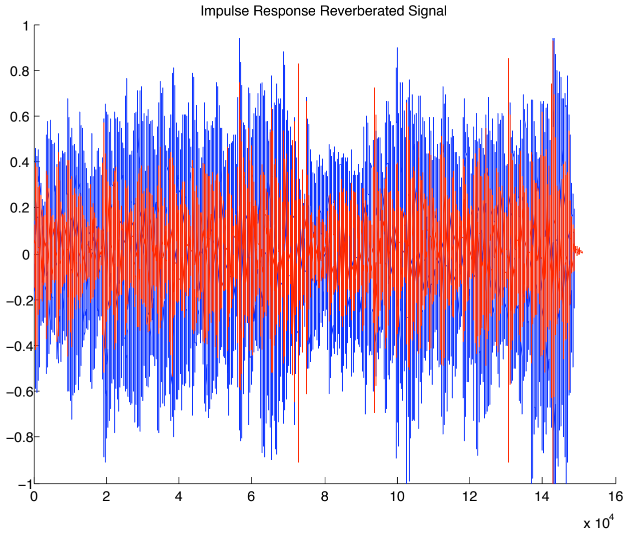
```
% reverb_convolution_eg.m
% Script to call implement Convolution Reverb

% read the sample waveform
filename='../acoustic.wav';
[x,Fs,bits] = wavread(filename);

% read the impulse response waveform
filename='impulse_room.wav';
[imp,Fsimp,bitsimp] = wavread(filename);

% Do convolution with FFT
y = fconv(x,imp);

% write output
wavwrite(y,Fs,bits,'out_IRreverb.wav');
```

MATLAB Convolution Reverb (Cont.)

Some example results:

Living Room Impulse Response Convolution Reverb:

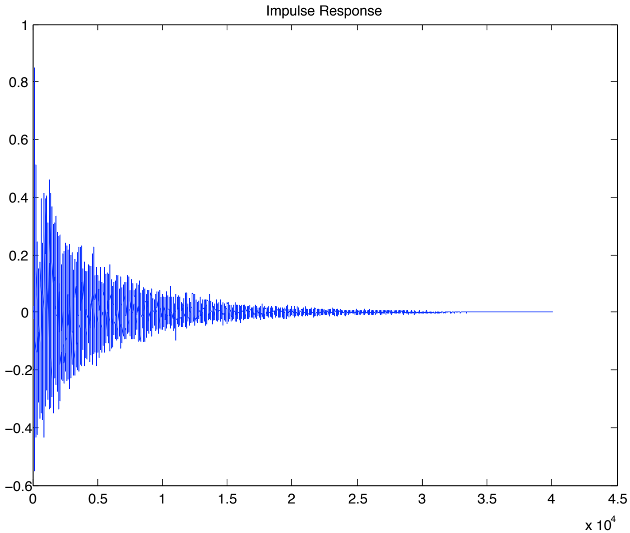


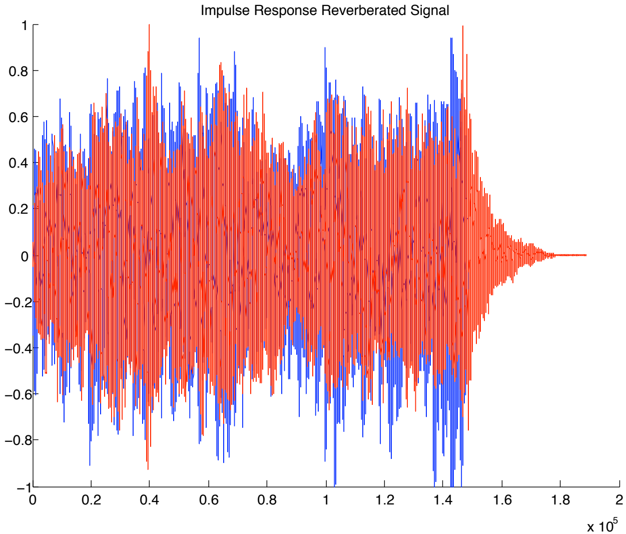


Click here to hear: original audio,
room impulse response audio, room impulse reverberated audio.

MATLAB Convolution Reverb (Cont.)

Cathedral Impulse Response Convolution Reverb:



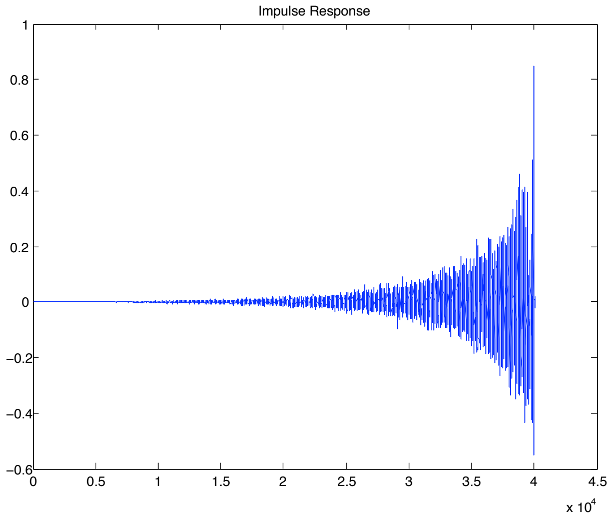


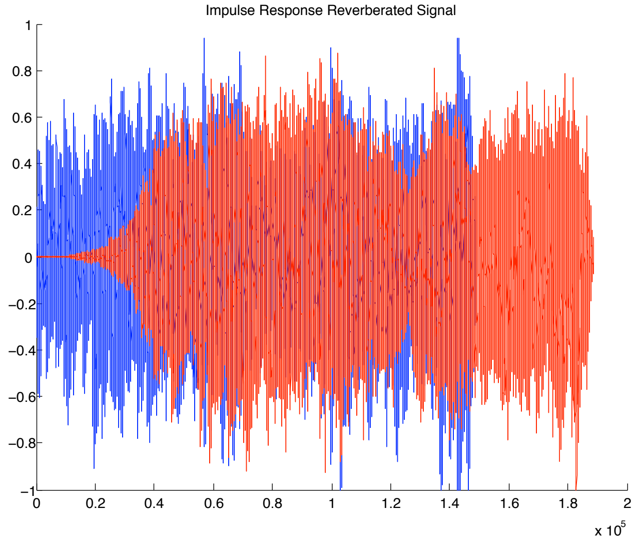
Click here to hear: original audio, cathedral impulse response audio, cathedral reverberated audio.

MATLAB Convolution Reverb (Cont.)

It is easy to implement some odd effects also

Reverse Cathedral Impulse Response Convolution Reverb:



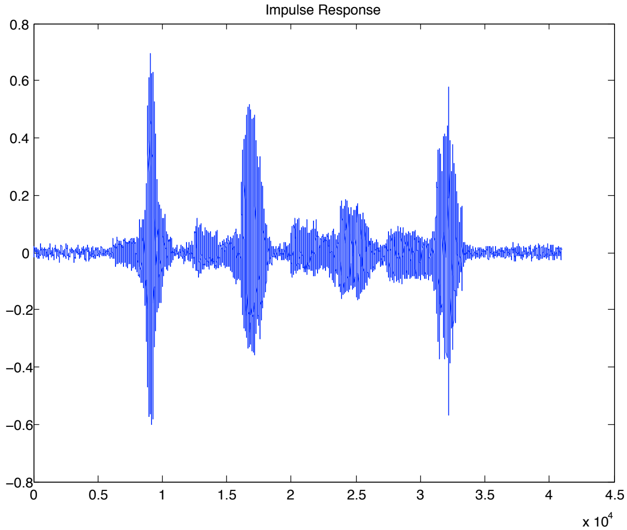


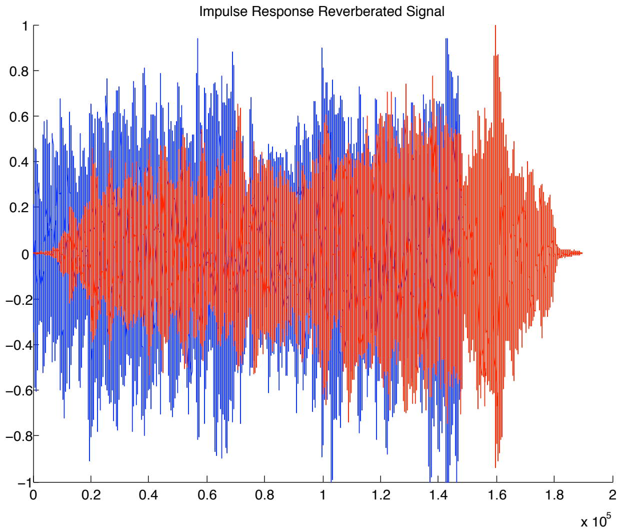
Click here to hear: original audio,
reverse cathedral impulse response audio,
reverse cathedral reverberated audio.

MATLAB Convolution Reverb (Cont.)

You can basically convolve with anything:

Speech Impulse Response Convolution Reverb:





Click here to hear: original audio,
speech *'impulse response'* audio, speech impulse reverberated audio.