

# Time-varying Filters

## Time-varying Filter Effects

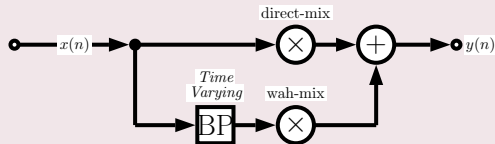
Some common effects are realised by simply time varying a filter in a couple of different ways:

**Wah-wah:** A bandpass filter with a (**modulated**) 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

Wah-wah, Signal flow diagram:



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

## Wah-wah Variations

- 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

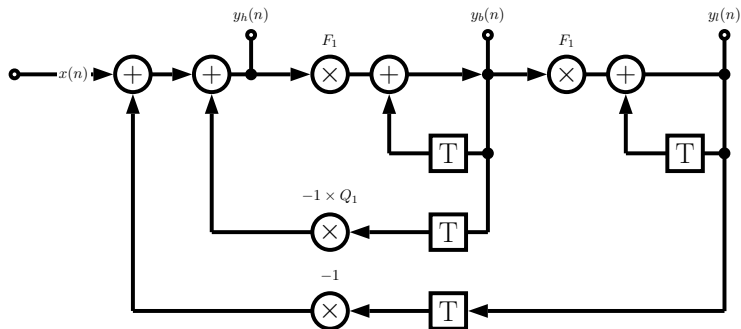
## The Practical State Variable Filter

In 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:

- $x(n)$  = input signal
- $y_l(n)$  = lowpass signal
- $y_b(n)$  = bandpass signal
- $y_h(n)$  = highpass signal

# The State Variable Filter Algorithm

State Variable Filter difference equations are given by:

$$y_l(n) = F_1 y_b(n) + y_l(n-1)$$

$$y_b(n) = F_1 y_h(n) + y_b(n-1)$$

$$y_h(n) = x(n) - y_l(n-1) - Q_1 y_b(n-1)$$

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

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

# MATLAB Wah-wah Implementation

## Making a Wah-wah

We simply implement the State Variable Filter with a Sinusoid **Modulated (variable) frequency**,  $f_c$ .

### 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  
%  
%  $y_l(n) = F1*y_b(n) + y_l(n-1)$   
%  $y_b(n) = F1*y_h(n) + y_b(n-1)$   
%  $y_h(n) = x(n) - y_l(n-1) - Q1*y_b(n-1)$   
%  
% vary Fc from 500 to 5000 Hz
```

# Wah-wah Implementation

## wah\_wah.m (Cont.):

```
infile = 'acoustic.wav';

% read in wav sample
[ x, Fs] = audioread(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;
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
```

# Wah-wah Implementation

## wah\_wah.m (Cont.):

```
% 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;
```



# Wah-wah Implementation

## wah\_wah.m (Cont.):

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

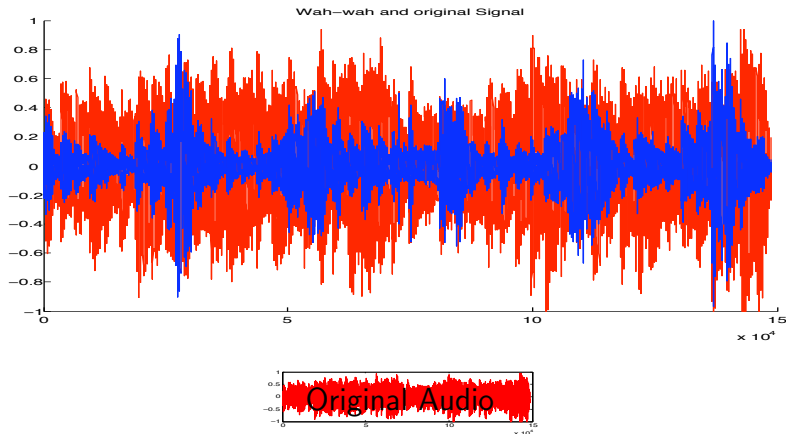
% first sample, to avoid referencing of negative signals
yh(1) = x(1);
yb(1) = F1*yh(1);
yl(1) = F1*yb(1);

% 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 and Output .....
```

# Wah-wah MATLAB Example (Cont.)

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



Click on images or here to hear: [original audio](#), [wah-wah audio](#).