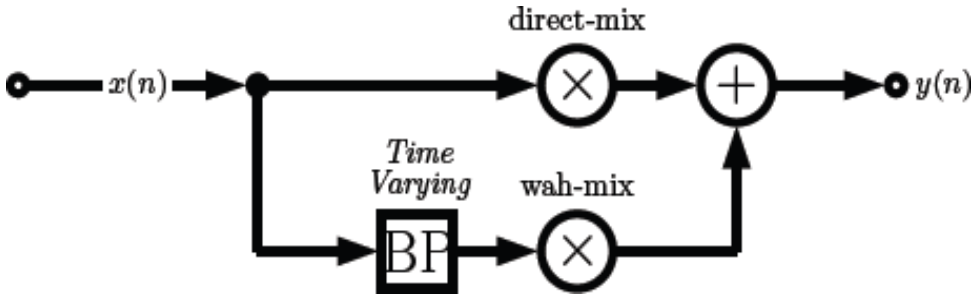
# 可參數化的Wah-Wah效果器

# WAH-WAH

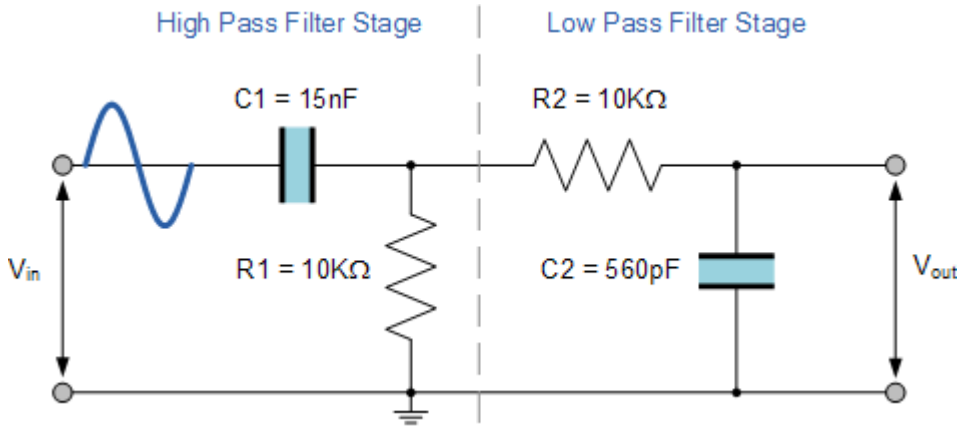
- A band-pass filter with time-varying resonant frequency and a small bandwidth. Filtered signal is mixed with direct signal.




- The Wah filter can be also called a *state variable filter*. A state variable filter is a type of active filter. It consists of one or more integrators, connected in some feedback configuration. One configuration is such that it can produce band pass, low pass and high pass outputs from a single input.

# LET'S UNDERSTAND BAND-PASS FILTERS

- Ahigh pass filter followed by a low-pass filter gives a band pass filter.

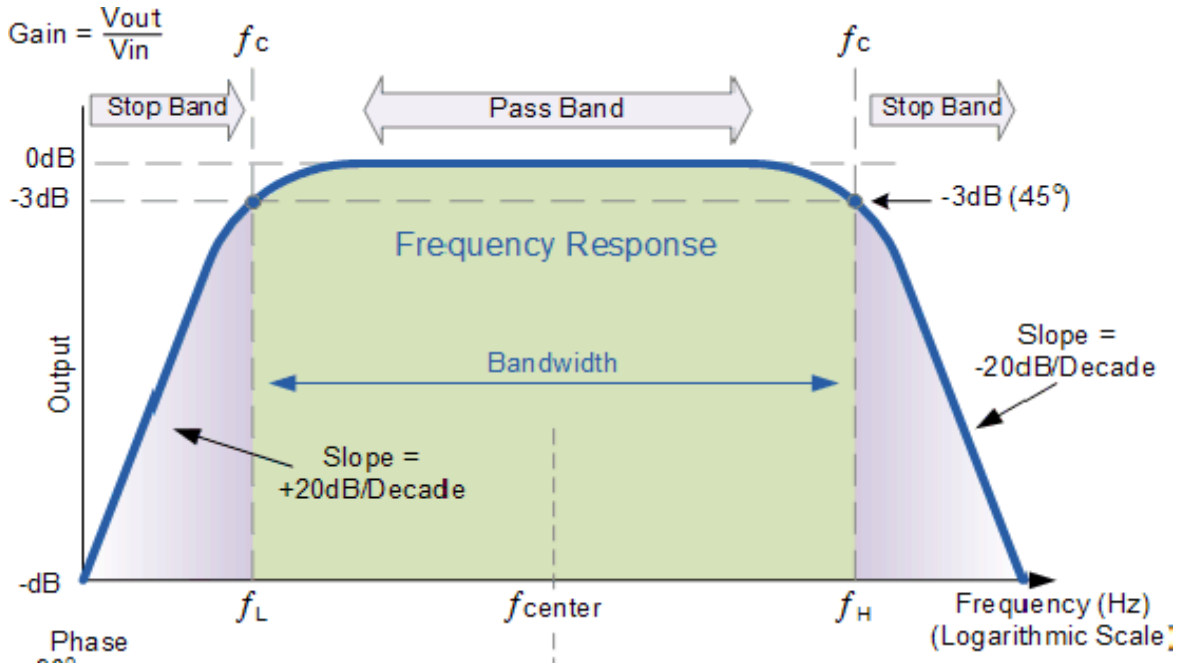



- <u>Centre Frequency Equation</u>

$$f_r = \sqrt{(f_L x f_H)}$$

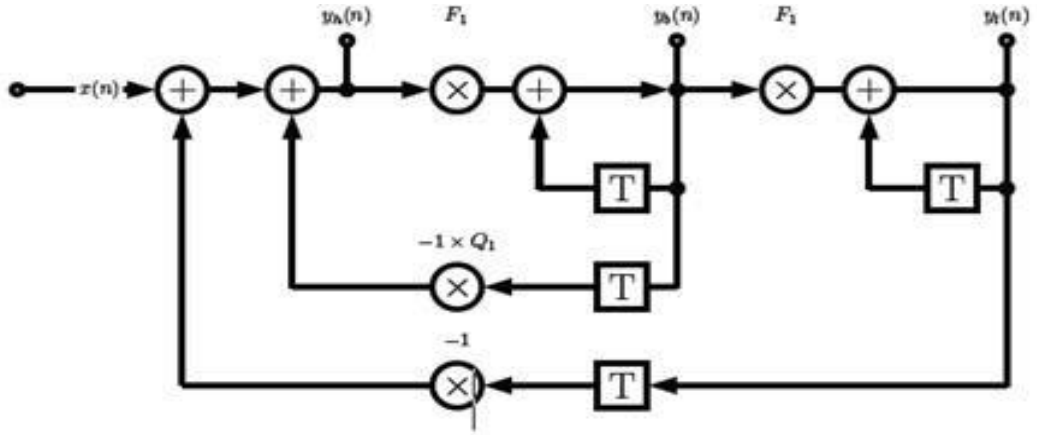
- Where, $f_r$ is the resonant or centre frequency
- $f_L$ is the lower -3dB cut-off frequency point
- $f_H$ is the upper -3db cut-off frequency point

## FREQUENCY RESPONSE OF A SECOND ORDER BAND PASS FILTER




In case of Wah filters, the resonant frequency (f center) varies with time and the bandwitdth is small.

# BLOCK DIAGRAM OF WAH FILTER




where:

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

DIFFERENCE EQUATIONS:

$$y_l(n) = F_1 y_b(n) + y_l(n-1)$$
$$y_b(n) = F_1 y_h(n) + y_b(n-1)$$
$$y_h(n) = x(n) - y_l(n-1) - Q_1 y_b(n-1)$$

Where

$$F_1 = 2\sin\left(pi * \frac{f_c}{f_s}\right)$$
$$Q_1 = 2d$$

fc=cut off frequency, d= damping.

# Pseudo code

```
%read wavefile
Fs,x = wavread( input.wav)
%EFFECT COEFFICIENTS
% 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
while(length(Fc) < length(x) )
        Fc= append(maxf:minf:delta)   %up
        Fc= append(maxf:minf:-delta)  %down
% trim tri wave to size of input
Fc=Fc[1:length(x)]
```

# Pseudo code

```
% difference equation coefficients
% must be recalculated each time Fc changes
F1 = 2*sin((pi*Fc[0])/Fs)
% this dictates size of the pass bands
Q1 = 2*damp
% create empty out vectors
yh=zeros(size(x))
yb=zeros(size(x))
yl=zeros(size(x))
% first sample, to avoid referencing of negative signals
yh[0] = x[0]
yb[0] = F1*yh[0]
yl[0] = F1*yb[0]
```

```
% apply difference equation to the sample
for n in range(1,len(x)-1)
    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)

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

%write wavefile
wavwrite( wahwah.wav,yb,Fs)
```

# 注意事項

- 繳交期限2018/4/25 17:00( 五點以前驗收繳交為 A 。Office Hour結束之前繳交為B。當周日午夜前繳交為C。之後以缺交論F。 )
- 作業請繳交至FTP : 140.116.82.230
  - username：signalsystem107
  - password：screamlab
- 格式
  - 所有程式限定使用Python，Matlab或是C語言
  - 命名規格(壓縮檔標題) :lab6_學號_姓名_vX (X為版本號)
    - Ex:lab6_F71234567_王大明_v1
  - 內容 : lab6.m or lab6.py、wahwah.wav