## FX Basics

# Filtering

STOMPBOX DESIGN WORKSHOP

Esteban Maestre

*CCRMA - Stanford University*

*August 2015*

# FX Basics: **Filtering**

Filtering effects modify the frequency content of the audio signal, achieving **boosting or weakening** specific **frequency bands** or regions.

Although their broad application to processing sound signals dates back from the early days of recording, their use application  to processing guitar electrical signal may have started in the 1950s.

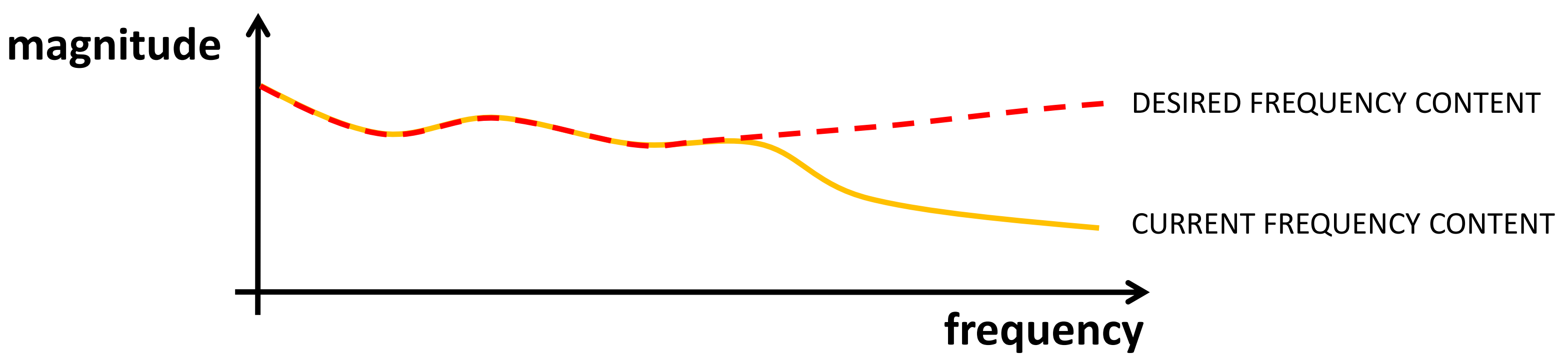
Filtering effects make use of **filters**, which are signal processors which **alter magnitude and phase** of signals by different amount s to different frequency components.

Ex:	equalization, wah-wah

# Equalization

Original term coined from the task of ‘adjusting the balance between of (or *equalize*)’ different frequency components of a signal.

Equalization is commonly achieved by means of a device specifically designed for a **user-friendly control** of the parameters governing the **behavior of filters** used for its construction.

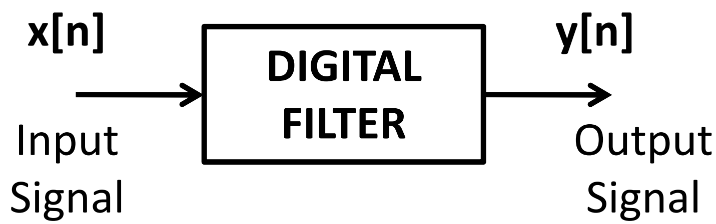



User-friendly interface to controlling filters so that a **desired alteration** is achieved…

**DIGITAL FILTERS!**

# Digital Filters

**Systems** that perform mathematical operations (multiplications and additions) to a **discrete input signal** x[n] to modify some of its characteristics and **obtain a discrete output signal** y[n].

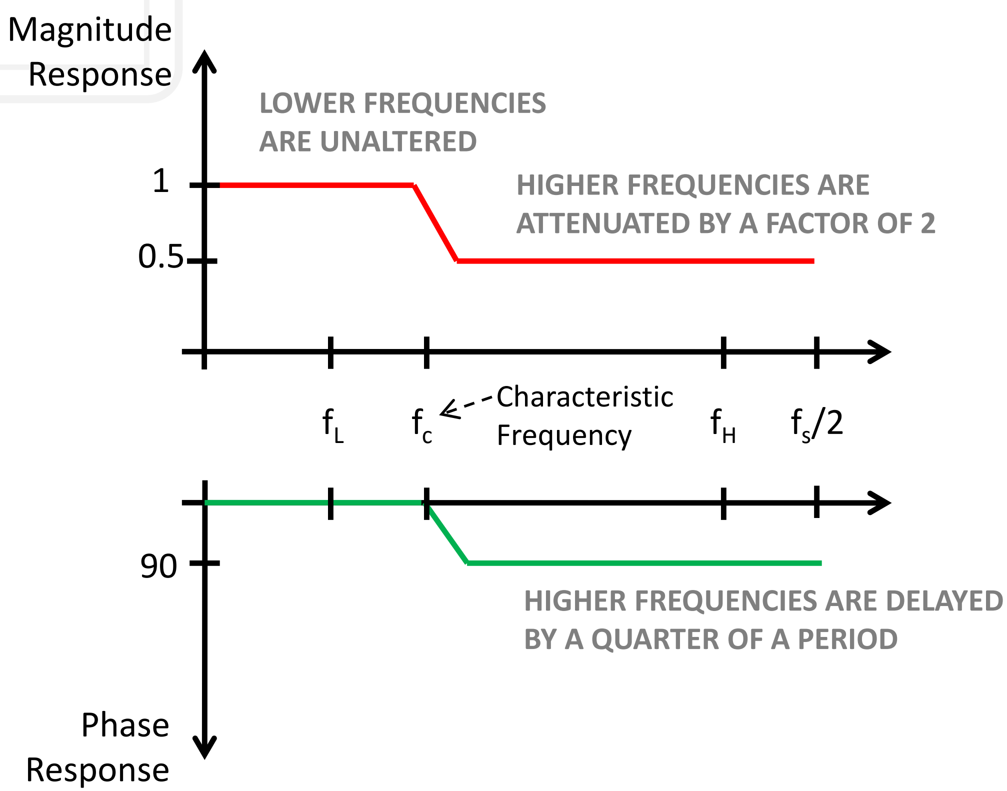


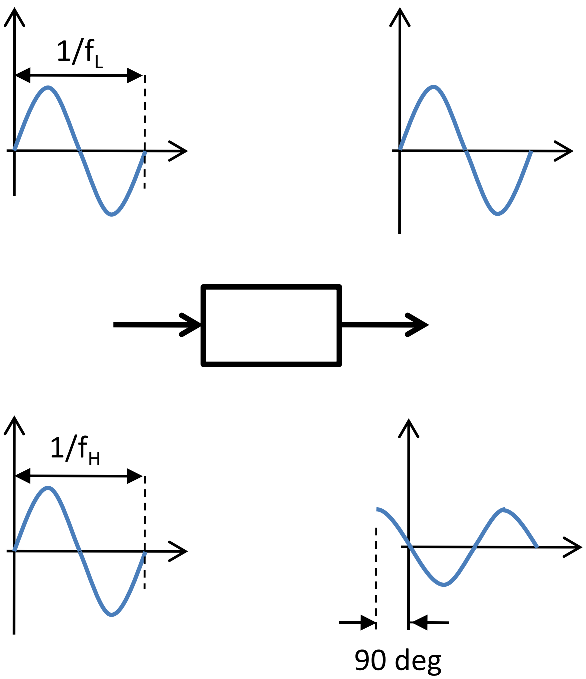

It is common to **describe a digital filter** in terms of **how it affects amplitude and phase** of **different frequency components** of a signal.

Ultimately, the design of digital filters is driven by such desired features. In general, **digital filter design** is not an easy task.

# Digital Filters (ii)

FX Basics:
**Filtering**

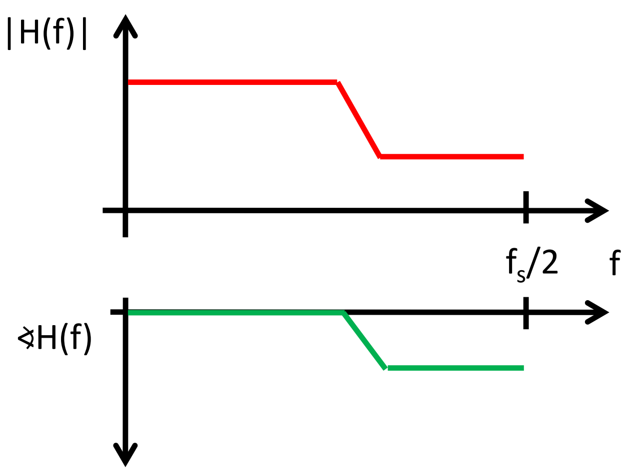


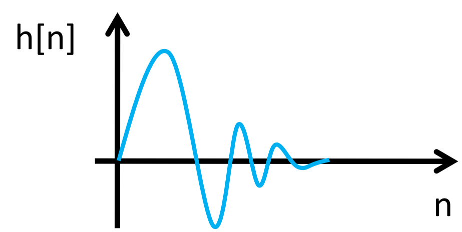
# Digital Filters (iii)

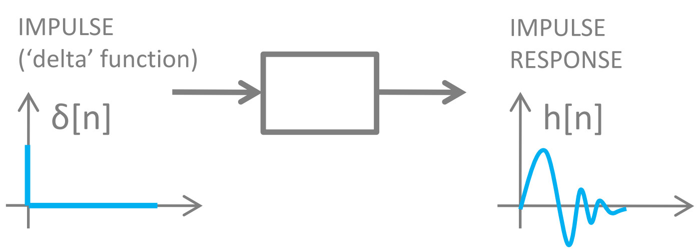
FREQUENY DOMAIN | DFT | TIME DOMAIN

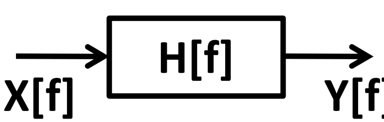
**FREQUENCY RESPONSE**




**IMPULSE RESPONSE**

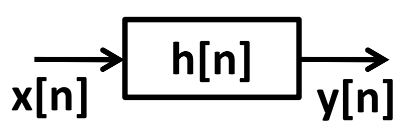



PRODUCT

**Y[f] = X[f]·H[f]**




CONVOLUTION

**y[n] = x[n]*h[n]**

# Digital Filters (iv)

How to explore the frequency domain response of a given filter?
Among other options…

**SINUSOIDAL ANALYSIS**

- Generate a sinusoidal $x_i[n]$ for each frequency $f_i$ to study
- Feed filter with each sinusoidal signal $x_i[n]$ and obtain a sinusoidal $y_i[n]$
- Obtain magnitude and phase responses for each frequency $f_i$ :

$$|H(f_i)| = A(y_i)/A(x_i) \qquad\qquad \measuredangle H[f_i] = \measuredangle y_i - \measuredangle x_i$$

**IMPULSE RESPONSE**

- Generate an impulse 'delta' signal $\delta[n]$
- Feed filter with signal $\delta[n]$ and obtain output signal $h[n]$
- Obtain $H(f)$ via DFT( $h[n]$ )
- Obtain magnitude and phase responses as:

$$|H[f]| = \text{abs}(\ H[\Omega]\ ) \qquad\qquad \measuredangle H[\Omega] = \text{angle}(\ H[\Omega]\ )$$

# Digital Filters (v)

FX Basics:
**Filtering**

Digital filters are commonly expressed by their **difference equation**:

$$y[n] = b_0 \cdot x[n] + b_1 \cdot x[n-1] + \ldots + b_M \cdot x[n-M] \longleftarrow \text{CURRENT AND PREVIOUS INPUT SAMPLES}$$

$$- a_1 \cdot y[n-1] - \ldots - a_N \cdot y[n-N] \longleftarrow \text{PREVIOUS OUTPUT SAMPLES}$$

$$= \underbrace{\sum_{i=0}^{M} b_i \cdot x[n-i]}_{\text{NON-RECURSIVE PART}} - \underbrace{\sum_{j=1}^{N} a_j \cdot y[n-j]}_{\text{RECURSIVE PART}}$$

**$b_i$ , $a_j$** $\longrightarrow$ FILTER COEFFICIENTS

**max(M,N)** $\longrightarrow$ FILTER ORDER

…or by their **transfer function** (in the frequency domain, through the 'Z' transform):

$$H(z) = \frac{Y(z)}{X(z)} = \frac{b_0 + b_1 \cdot z^{-1} + \ldots + b_N \cdot z^{-M}}{1 + a_1 \cdot z^{-1} + \ldots + a_N \cdot z^{-N}}$$

**$z^{-M}$** denotes *M* samples of delay

# Digital Filters (vi)

Two main types of digital filters:

$$H(z) = \frac{b_0 + b_1 \cdot z^{-1} + \ldots + b_M \cdot z^{-M}}{1 + a_1 \cdot z^{-1} + \ldots + a_N \cdot z^{-N}}$$

**F**inite **I**mpulse **R**esponse ( **FIR** )

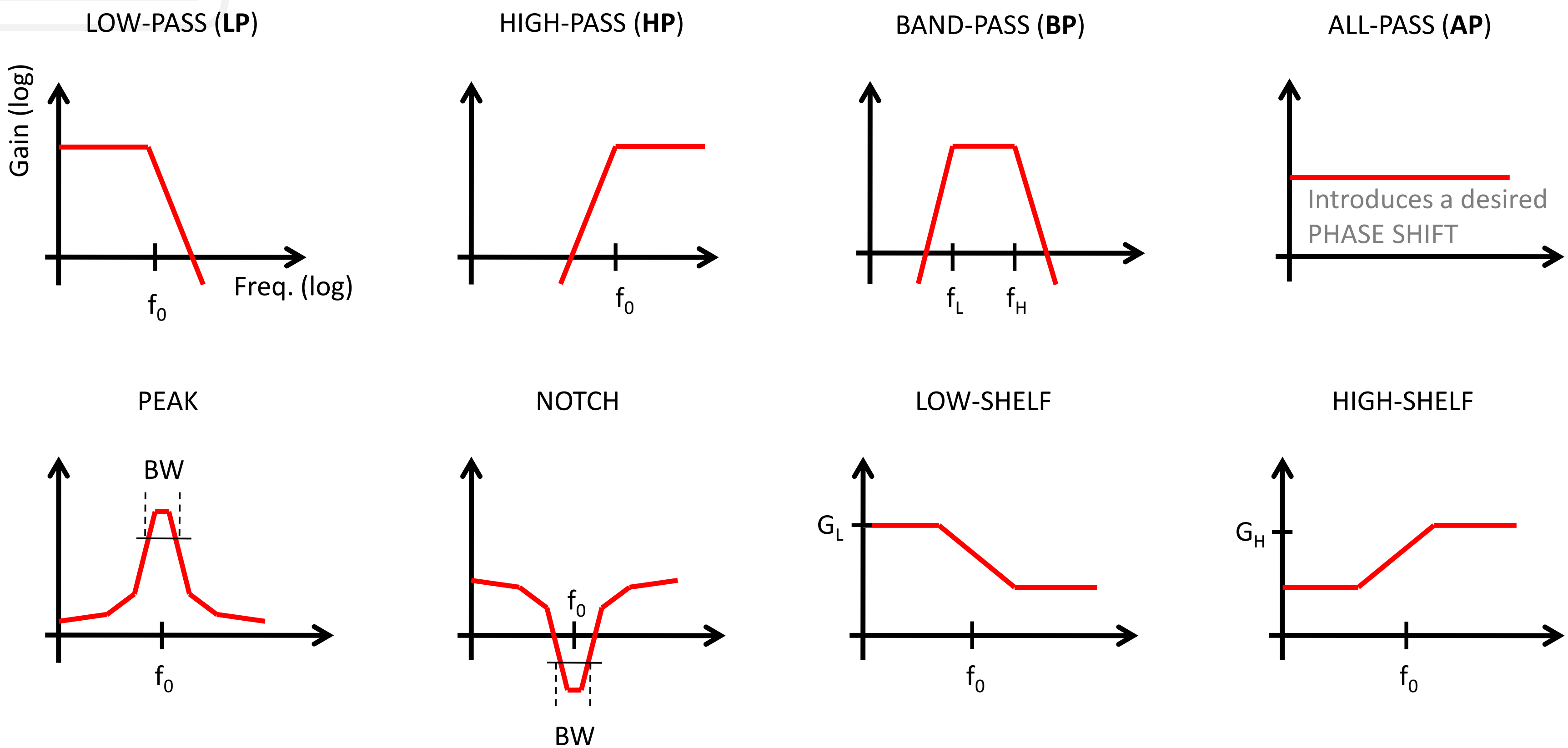
- Presents only $b_i$ coefficients being non-zero : **NON-RECURSIVE**
- Finite h[n]
- Phase response is linear

**I**nfinite **I**mpulse **R**esponse ( **IIR** )

- Presents both $b_i$ and $a_j$ coefficients being non-zero: **RECURSIVE**
- Infinite h[n]
- Phase response is non-linear
- Need less computations for similar desired characteristics
- May suffer from numerical problems due to feedback

# **Digital Filters** (vii)

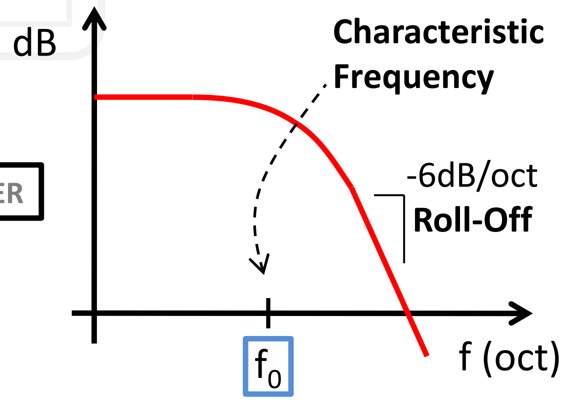
Some **prototypical** basic filters ( <u>magnitude</u> response ):



# Digital Filters (viii)

**LPF** (Butterworth) design parameters/constraints:

1st ORDER




Desired $\{\mathbf{f_0}\}$

↓

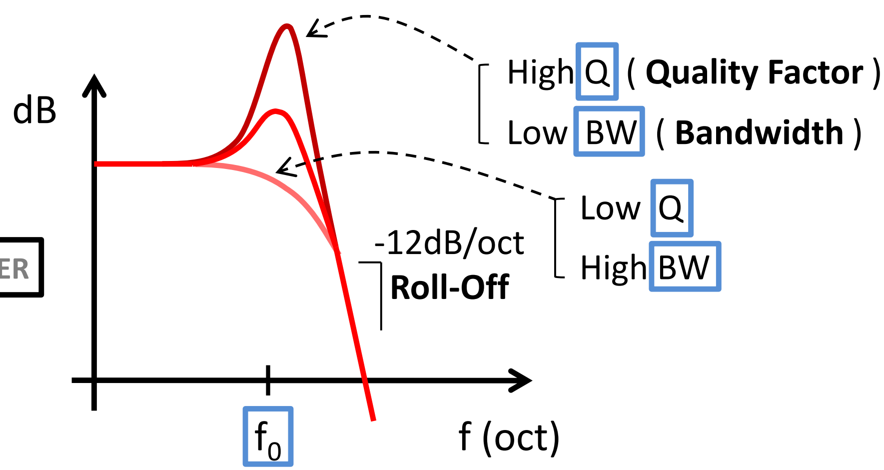
**DESIGN PROCEDURE**

↓

Filter Coefficients $\{\mathbf{b_i},\mathbf{a_j}\}$

---

2nd ORDER




Desired $\{\mathbf{f_0},\mathbf{Q}\}$ or $\{\mathbf{f_0},\mathbf{BW}\}$

↓

**DESIGN PROCEDURE**

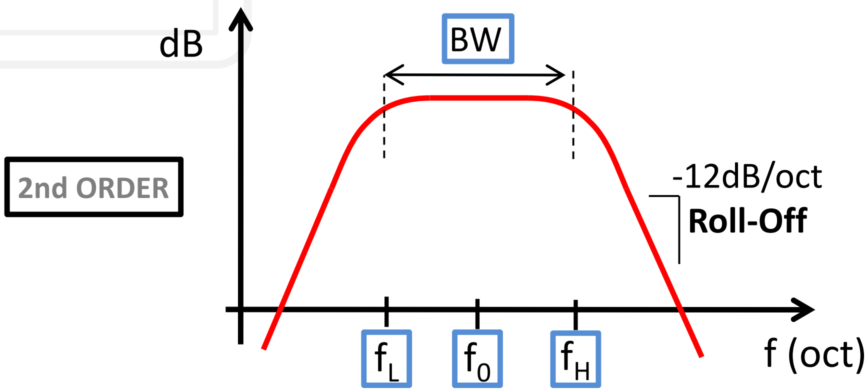
↓

Filter Coefficients $\{\mathbf{b_i},\mathbf{a_j}\}$

FX Basics: **Filtering**

# **Digital Filters** (ix)

**BPF** design parameters/constraints:




Desired $\{\mathbf{f_L,f_H}\}$ or $\{\mathbf{f_0,BW}\}$

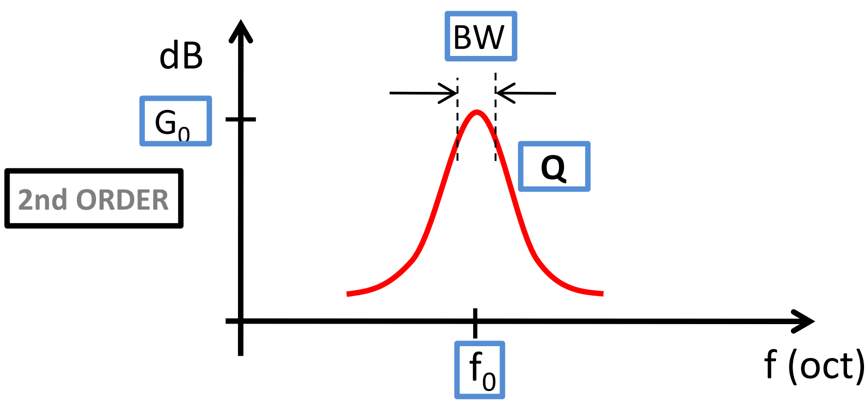
↓

**DESIGN PROCEDURE**

↓

Filter Coefficients $\{\mathbf{b_i,a_j}\}$

**PEAK** design parameters/constraints:




Desired $\{\mathbf{f_0,G_0,BW}\}$ or $\{\mathbf{f_0,G_0,Q}\}$

↓

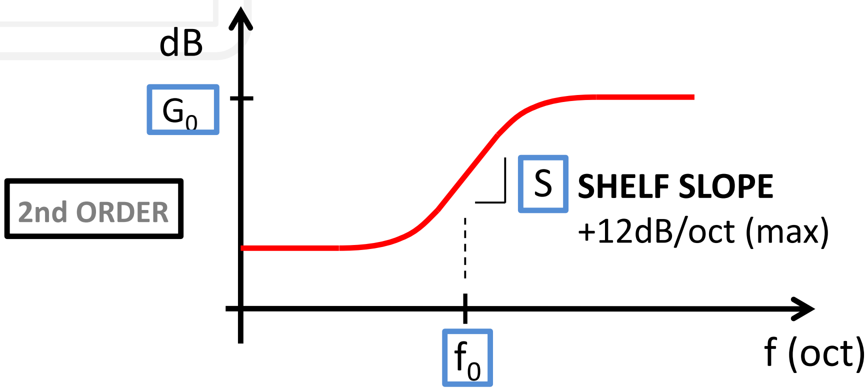
**DESIGN PROCEDURE**

↓

Filter Coefficients $\{\mathbf{b_i,a_j}\}$

# Digital Filters (x)

**HIGH-SHELF** design parameters/constraints:




Desired $\{\mathbf{f_0},\mathbf{G_0},\mathbf{S}\}$

↓

**DESIGN PROCEDURE**

↓

Filter Coefficients $\{\mathbf{b_i},\mathbf{a_j}\}$

All these filters functions can be implemented by means of the 2nd order '**BIQUAD**' section:

$$H(z) = \frac{b_0 + b_1 \cdot z^{-1} + b_2 \cdot z^{-2}}{a_0 + a_1 \cdot z^{-1} + a_N \cdot z^{-2}}$$

How to design them? Extensive theory & literature!!

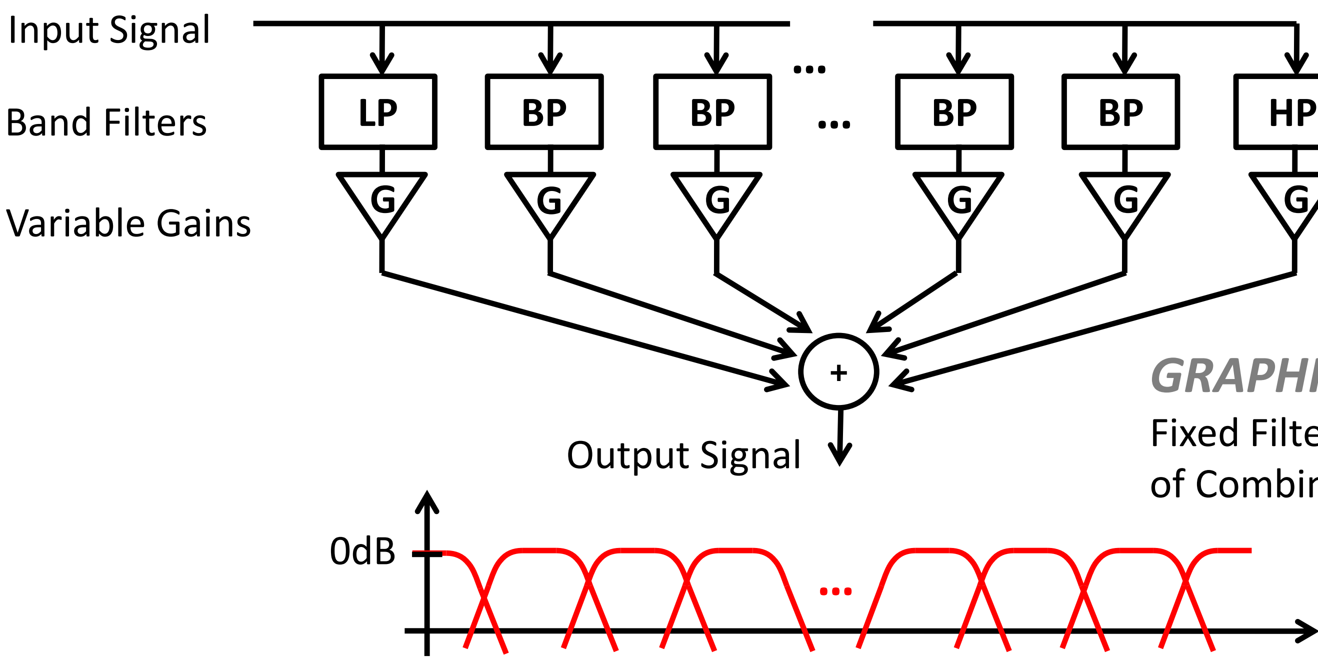
→ Quick method: R. Bristow-Johnson's cookbook:

http://www.musicdsp.org/files/Audio-EQ-Cookbook.txt

pd~ 04_stomp_filtering_1.pd

# **Equalization** (ii)

**N-BAND EQUALIZER** by **PARALLEL**
BAND-DEDICATED, FIXED FILTERS




***GRAPHIC EQUALIZER***
Fixed Filters: only Control of Combination (Gains)

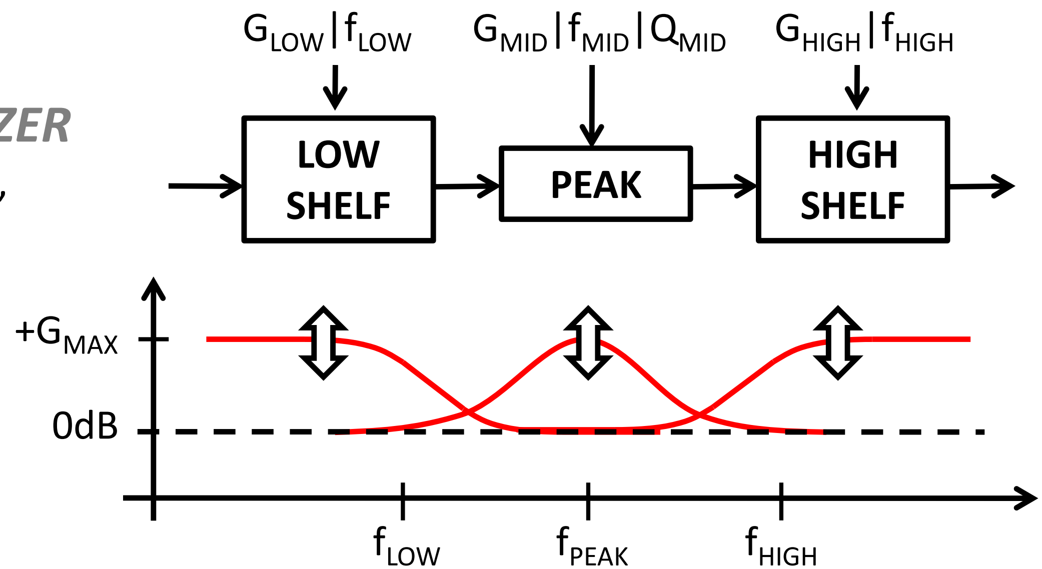
# Equalization (iii)



**N-BAND EQUALIZER** by **CASCADE** of
BAND-DEDICATED, CONTROLLABLE SECTIONS

Input Signal → **Section 1** → … → **Section N** → Output Signal

***PARAMETRIC EQUALIZER***
Variable Gains, Frequencies, and Bandwidths.




**ISSUE:** Changes in Gain/Frequency lead to Q/BW variation…

Constant-Q filters!

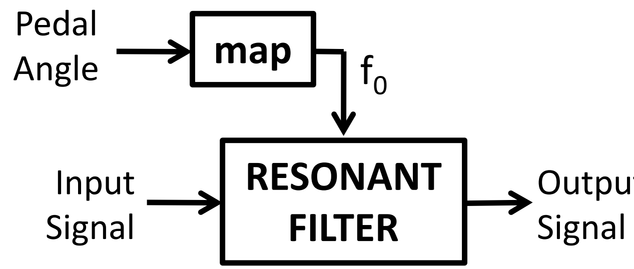
http://www.rane.com/note101.html

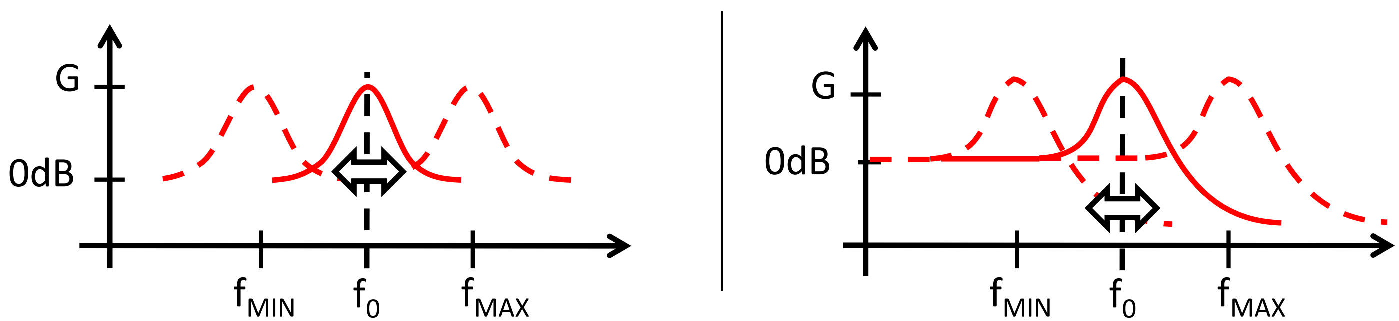
# Wah-wah

Dating back from the 60s, its name was given after voice tone modulation (*formant* shift) caused by transition between vowels.

http://www.geofex.com/article_folders/wahpedl/voicewah.htm

In its most basic form, it consists on shifting the center frequency of a resonant filter
(Peak BP or LP)




pd~ 05_stomp_filtering_2.pd