COM3502-4502-6502
SPEECH PROCESSING

Lecture 17
Linear Filters

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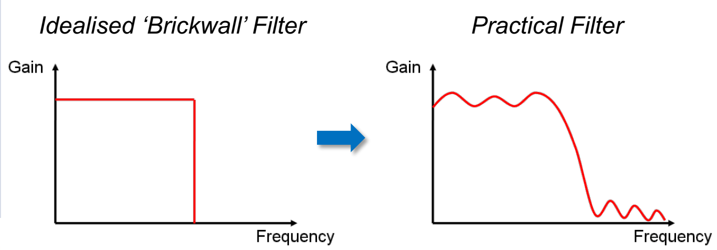
Idealised *vs.* Practical Filters

- Filters are often described in idealised terms, e.g. as 'brickwall' filters
- In reality, practical filters approximate the idealised forms



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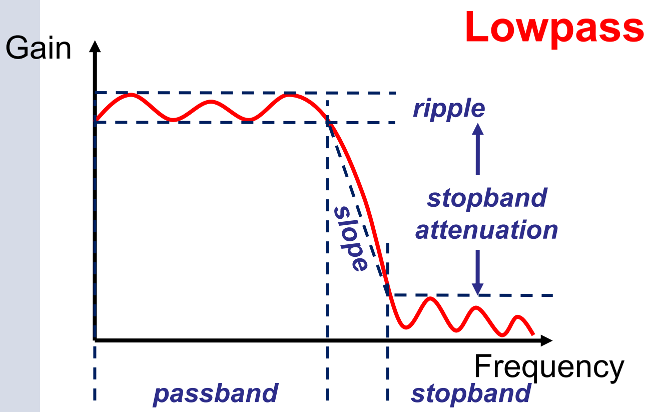
Practical Filters



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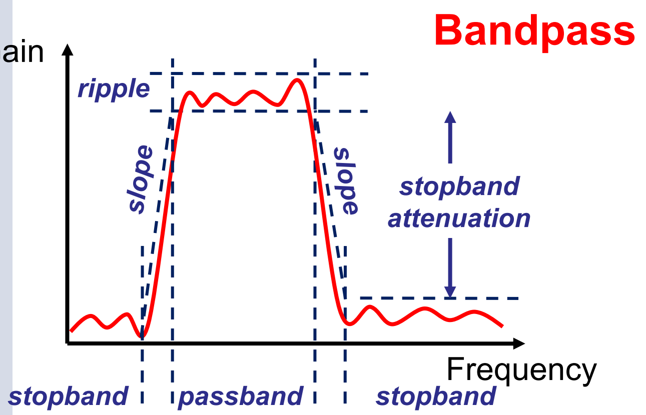
Practical Filters



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Filter Design

- The behaviour of a filter is defined by the number and the location of its ‘poles’ and ‘zeros’
(*as we saw in Lecture #16*)
- The complexity of a filter is often characterised by its ‘order’
- In a digital filter, order is defined as the number of *past* (*input or output*) values that are involved in the calculation
- For example, a filter that takes two past values is termed a ‘second-order filter’
- Filter order can thus be related to the number of poles and zeros
- Hence, the degree to which a practical filter approximates the idealised form is a function of its order
(*i.e. the number of poles and zeros*)

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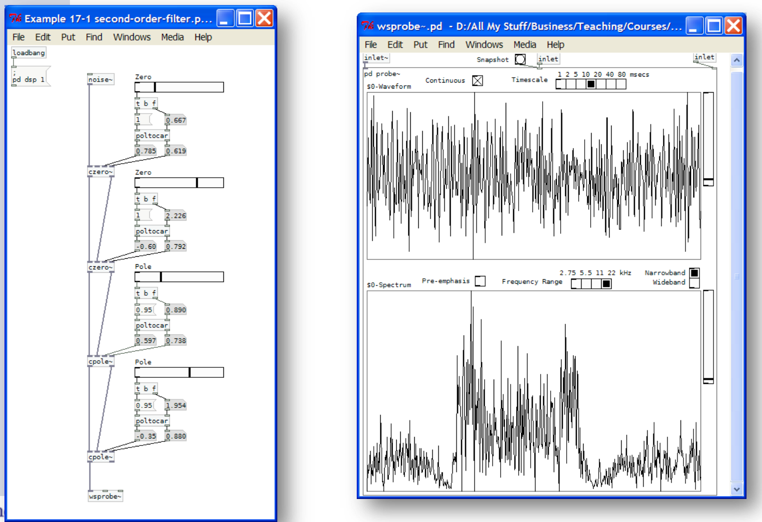
2-Pole 2-Zero 1st-Order Filtering



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General Difference Equation

- The general difference equation may include any number of past inputs and outputs …

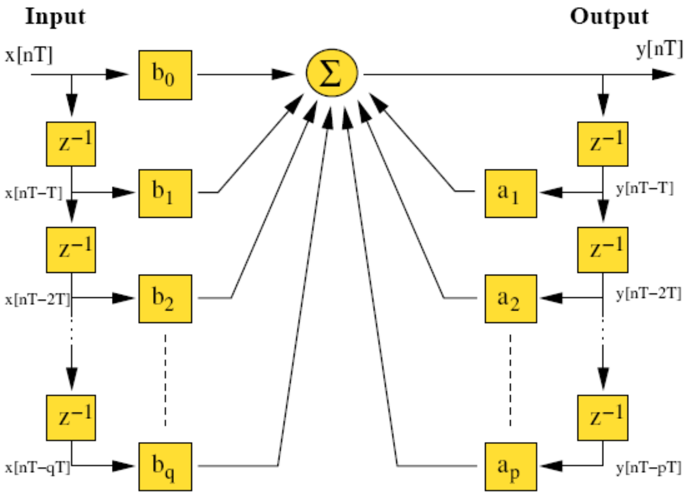
$$y[nT]=a_1y[(n-1)T]+a_2y[(n-2)T]+\ldots+a_py[(n-p)T]+$$
$$b_0x[nT]+b_1x[(n-1)T]+b_2x[(n-2)T]+\ldots+b_qx[(n-q)T]$$

- Taking Z transforms …

$$Y(Z)=Y(z)\sum_{i=1}^{p}a_iz^{-i}+X(z)\sum_{i=0}^{q}b_iz^{-i}$$

$$H(z)=\frac{\overbrace{\sum_{i=0}^{q}b_iz^{-i}}^{\text{zeros}}}{\underbrace{1-\sum_{i=1}^{p}a_iz^{-i}}_{\text{poles}}}$$

General Digital Filter



General Digital Filter

- A very wide range of transfer functions (*and associated frequency responses*) may be obtained by appropriate choice of the filter parameters
- p = number of poles, q = number of zeros
- For the case $p = 0$, the resulting non-recursive filter is said to be a 'finite impulse response - FIR' filter
- If $p > 0$, the resulting recursive filter is said to be an 'infinite impulse response - IIR' filter

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'FIR' Filters

- Finite impulse response (*FIR*) filters express each output sample as a weighted sum of the last N inputs (*where N is the order of the filter*)
- Advantages
  - inherently stable since they don't use feedback (*i.e. only zeros*)
  - the coefficients are usually symmetrical, hence the phase response is linear and signals of all frequencies are delayed equally
  - overflow is straightforward to avoid
  - generally easier to design than IIR filters
- Disadvantages
  - may require significantly more processing and memory resources than the equivalent IIR filter
  - often require a much higher filter order than IIR filters
  - delay can be much greater than for an equivalent IIR filter

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'IIR' Filters

- Infinite impulse response (*IIR*) filters are the digital counterpart to analog filters
  … because they contain an internal state, and the output and the next internal state are determined by the previous inputs and outputs
- Advantages
  - normally require less computing resources than an FIR filter of similar performance
- Disadvantages
  - due to the feedback, high-order IIR filters may have problems with instability, arithmetic overflow and limit cycles
  - careful design is required to avoid such pitfalls
  - the time delay through such a filter is frequency-dependent (*since the phase shift is inherently a non-linear function of frequency*)

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'IIR' Filters

- IIR filters all approximate the ideal *brickwall* filter …
  - Butterworth
  - Chebyshev (*types I and II*)
  - elliptic
  - Bessel
- High-order IIR filters can easily become unstable
- This is much less of a problem with first and second-order filters
- 2nd-order IIR filters are often called 'biquads'
- Higher-order filters are typically implemented as a cascade of 'biquad sections'

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'Biquad' Filters

- A '**biquadratic**' (*biquad*) filter is a 2nd-order recursive linear IIR filter, containing *two* poles and *two* zeros
- The name refers to the fact that its Z domain transfer function is the ratio of two *quadratic* functions …

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

- This is derived from the following difference equation …

$$y[kT] = b_0 x[kT] + b_1 x[(k-1)T] + b_2 x[(k-2)T] - a_1 y[(k-1)T] - a_2 y[(k-2)T]$$

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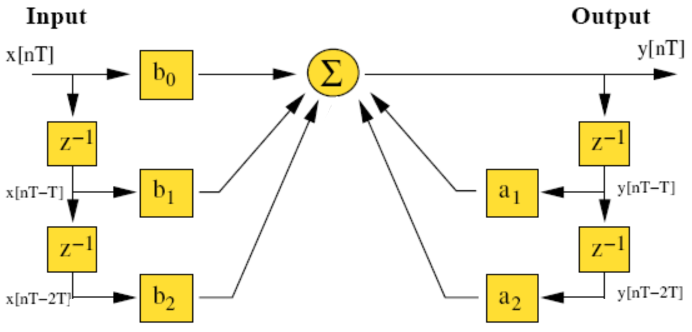
'Biquad' Filters



$$y[kT] = b_0 x[kT] + b_1 x[(k-1)T] + b_2 x[(k-2)T] - a_1 y[(k-1)T] - a_2 y[(k-2)T]$$

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The Biquad Filter in Pd

- Pd provides a raw biquadratic filter object [biquad~]
- The five coefficients (b_0, b_1, b_2, a_1, a_2) can be sent as a single message
- For a bandpass filter, the coefficients can be calculated using the [bandpass] object
- Pd provides the following objects for calculating filter coefficients for [biquad~] ...
  - [lowpass] [highpass]
  - [bandpass] [notch]
  - [lowshelf] [highshelf] [hlshelf]
  - [equalizer]

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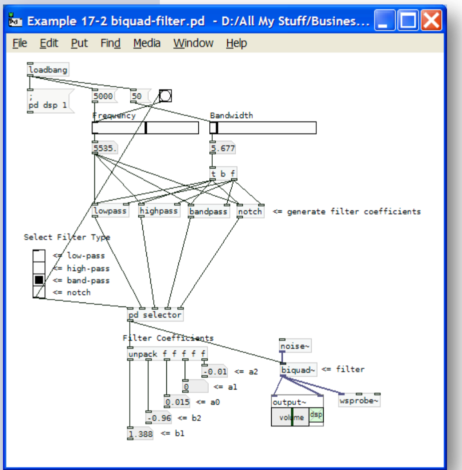
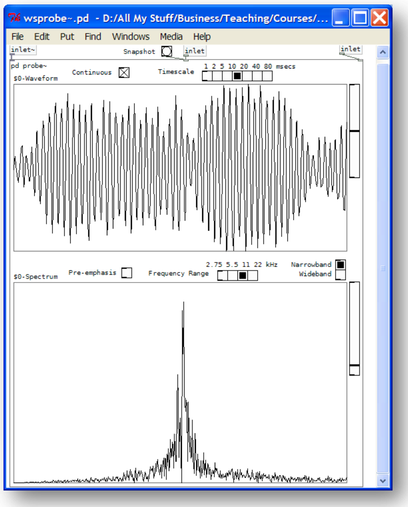
'Biquad' Filters





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Filters in Speech Processing

- General filtering
  – notch filtering
  (*to remove interference and noise*)
  – pre-emphasis
  (*to equalise the frequency response*)
- Modelling speech production
  – the source-filter model (*Lecture 3*)
- Modelling the auditory system
  – filter-bank analysis (*Lecture 4*)

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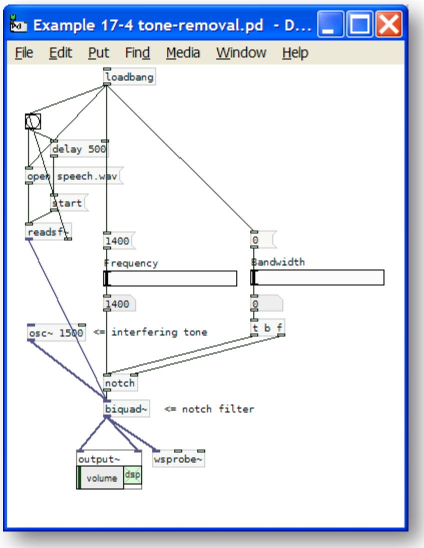
Tone Removal by Notch Filtering



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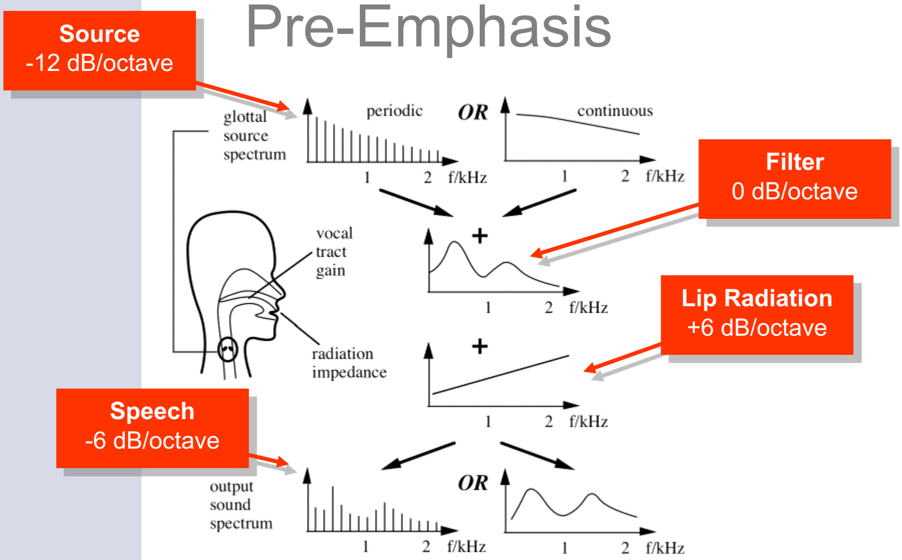
Pre-Emphasis



http://www.phys.unsw.edu.au/jw/voice.html

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Pre-Emphasis

- Averaged over time, speech has an overall spectral slope of ~ -6 dB/octave
- This means that there can be a large amplitude difference between the lowest *(~50 Hz)* and highest *(~8 kHz)* frequency components
- E.g. over 8 octaves there will be a (6x8) 48 dB difference (*which is comparable to the dynamic range from the quietest to the loudest speech sounds*)
- This can make processing difficult (*especially for functions such as spectral peak picking, i.e. formant tracking*)
- It is therefore usual to 'pre-emphasise' a speech signal by giving it a +6 dB/octave lift
- In Pd this can be achieved by placing a 'real zero' near the origin, e.g. `[rzero~ 1]`

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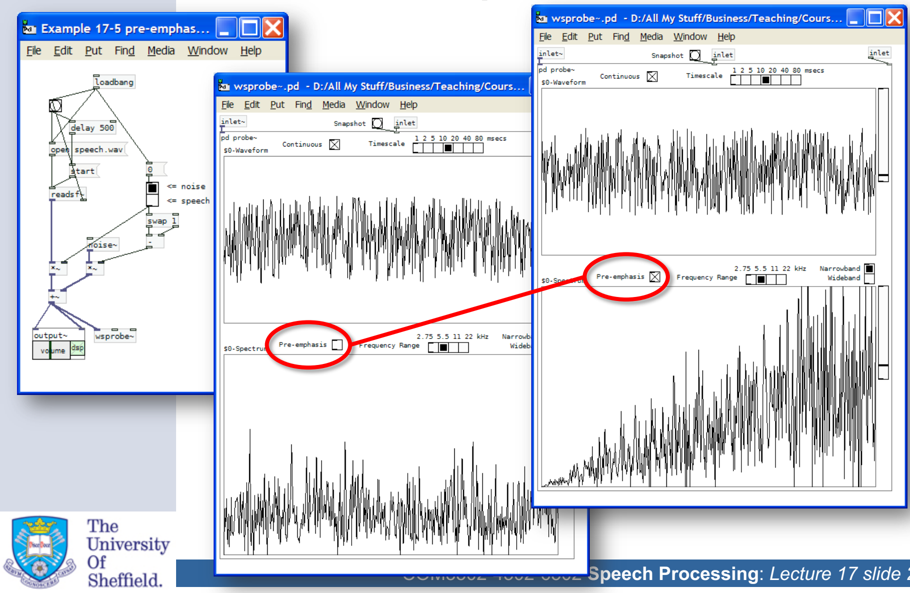
Pre-Emphasis



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Auditory Filters

- Filterbanks associated with the DFT/FFT have constant bandwidths and are centered at uniformly spaced locations along the frequency axis
- To better model the frequency response characteristics of the human ear, many researchers use filters inspired by the auditory system
- These typically have *non-uniform* bandwidths and *non-uniform* spacing of center frequencies
- The most well-known auditory filter is the 'gammatone'
- This filter has an impulse response that is the product of a gamma distribution and sinusoidal tone

Patterson, R. D., & Moore, B. C. J. (1986). Auditory filters and excitation patterns as representations of frequency resolution. In B. C. J. Moore (Ed.), *Frequency Selectivity in Hearing* (pp. 123-177). London: Academic Press Ltd.

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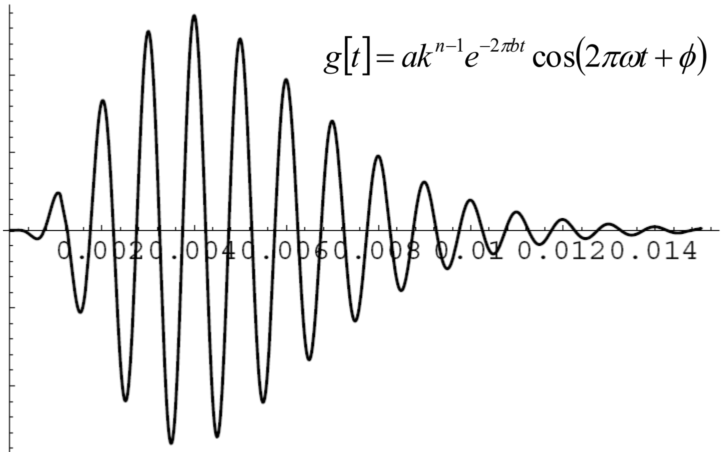
‘Gammatone’ Filter

$$g[t] = ak^{n-1}e^{-2\pi bt}\cos(2\pi\omega t + \phi)$$



Slaney, M. (1993). ***An efficient implementation of the Patterson-Holdsworth auditory filter bank*****: Apple Computer.**

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The Gammatone in Pure Data

- Daniel Pressnitzer and Dan Gnansia have implemented a gammatone filterbank in Pd: [gammabank_dm~]
- It is part of an ‘Audition Library’ which contains other useful functions, such as …
  - gammatone re-synthesis
  - multiplexers
  - de-multiplexers
  - outer and middle ear filters
- The code is based on:
  - Hohmann, V. (2002). Frequency analysis and synthesis using a Gammatone filterbank. *Acta Acustica*, 88, 433-442.
- The Audition toolkit is *not* available on the University machines, but is downloadable (*for Windows*) from:
  http://lumiere.ens.fr/Audition/tools/realtime/

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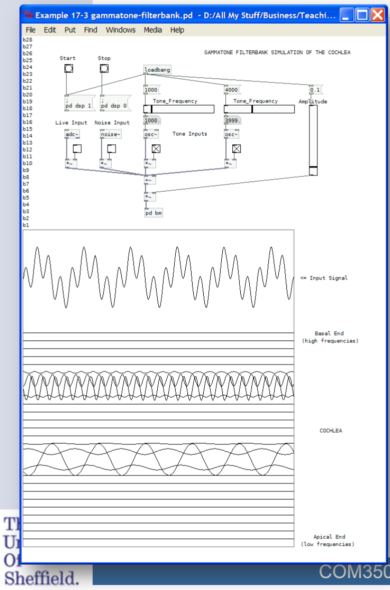
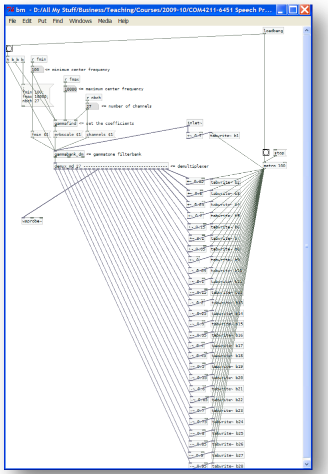
Gammatone Filterbank





Improved version of
'Example 4-1 cochlea-simulation.pd'

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This lecture has covered …

- Idealised *vs.* practical filters
- Filter design
- Second-order filters
- The general difference equation
- FIR and IIR filters
- The 'biquad' filter
- Filters in speech processing
- The 'gammatone' filter

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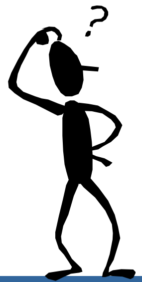
Any Questions ?



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Next time …

Linear Prediction

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