Be Advised!

This part of the course will be covered from

Digital Signal Processing: A Practical Approach

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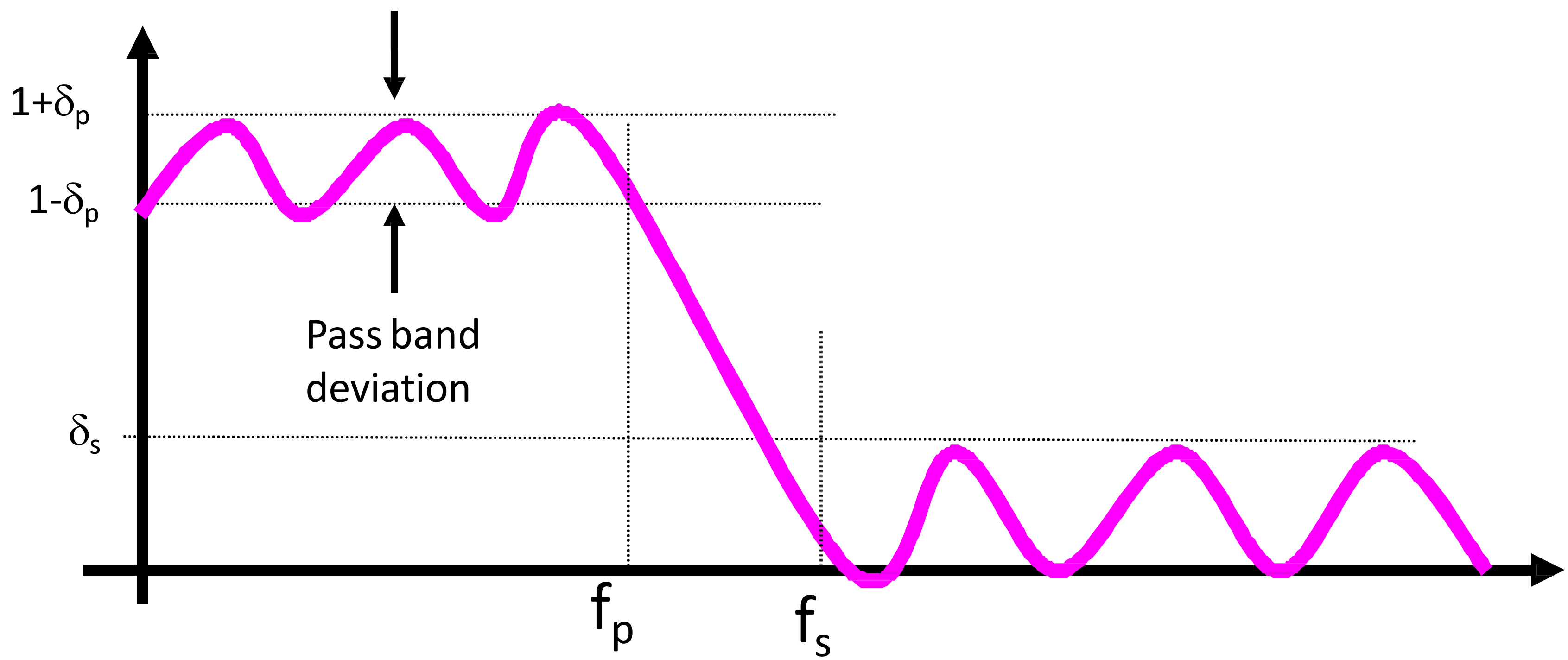
Filter

- A filter is essentially a *system* or network that selectively **changes** the **wave shape**, **amplitude-frequency** and/or **phase-frequency** characteristics of a signal in a desired manner

- In general, filtering is the processing of a time-domain signal, resulting in some change in the spectral content of the signal

Filter bands

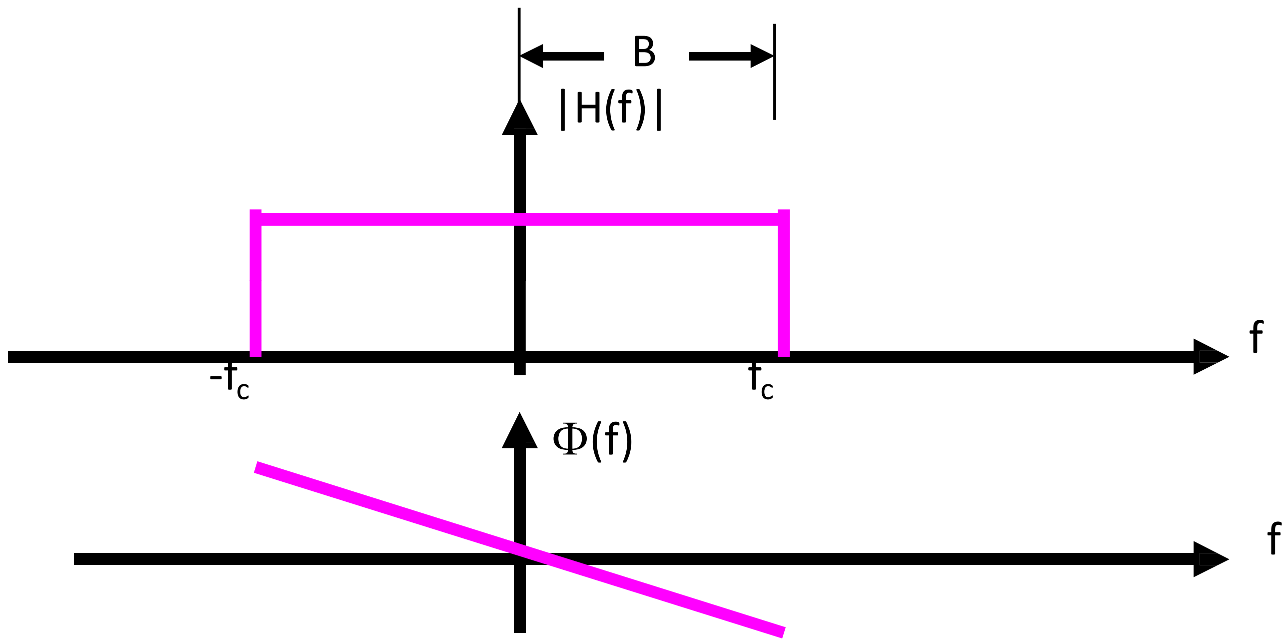
- Pass band
- Stop band
- Transition band



Filter Types

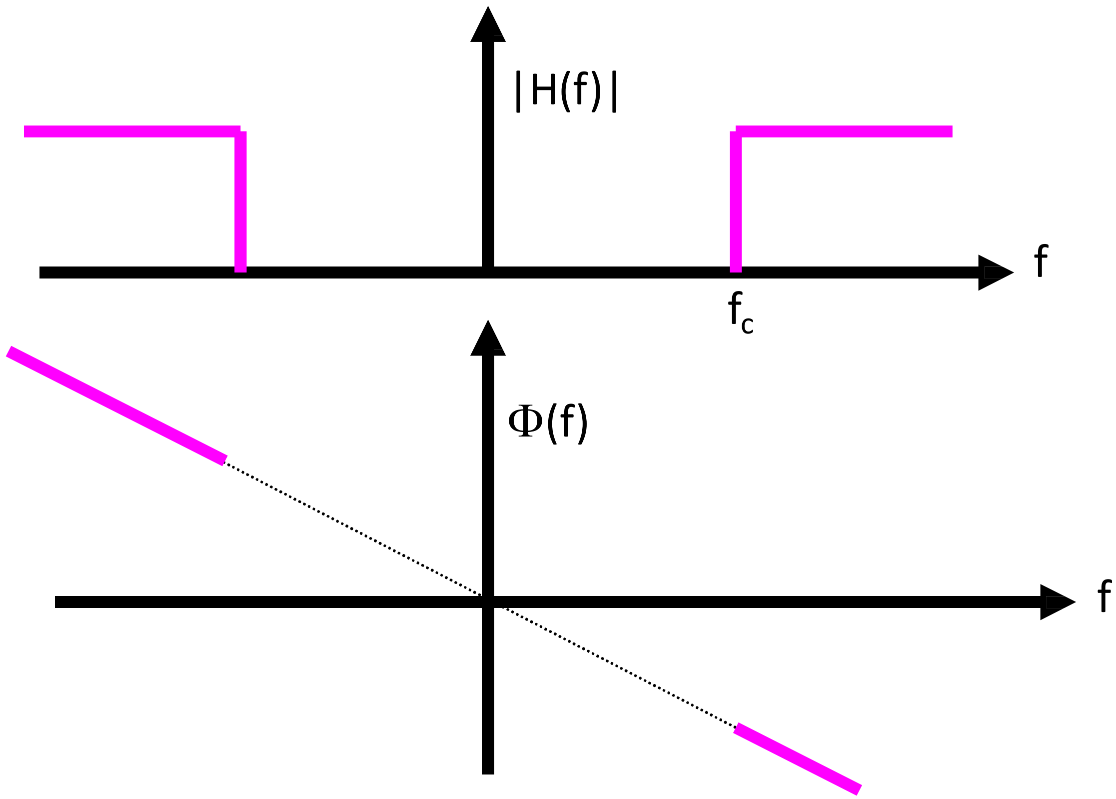
- Low Pass
- High Pass
- Band Pass
- Band Stop

Low Pass Filter



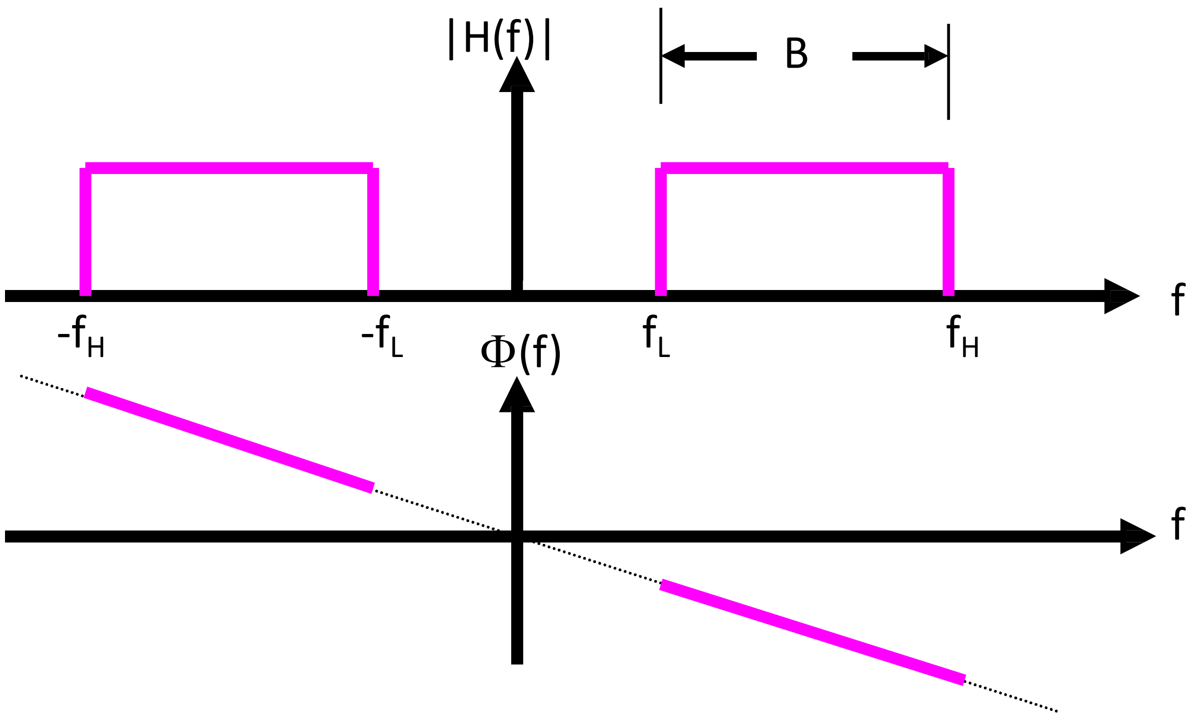
Passes frequencies smaller than a cut-off

High Pass Filter



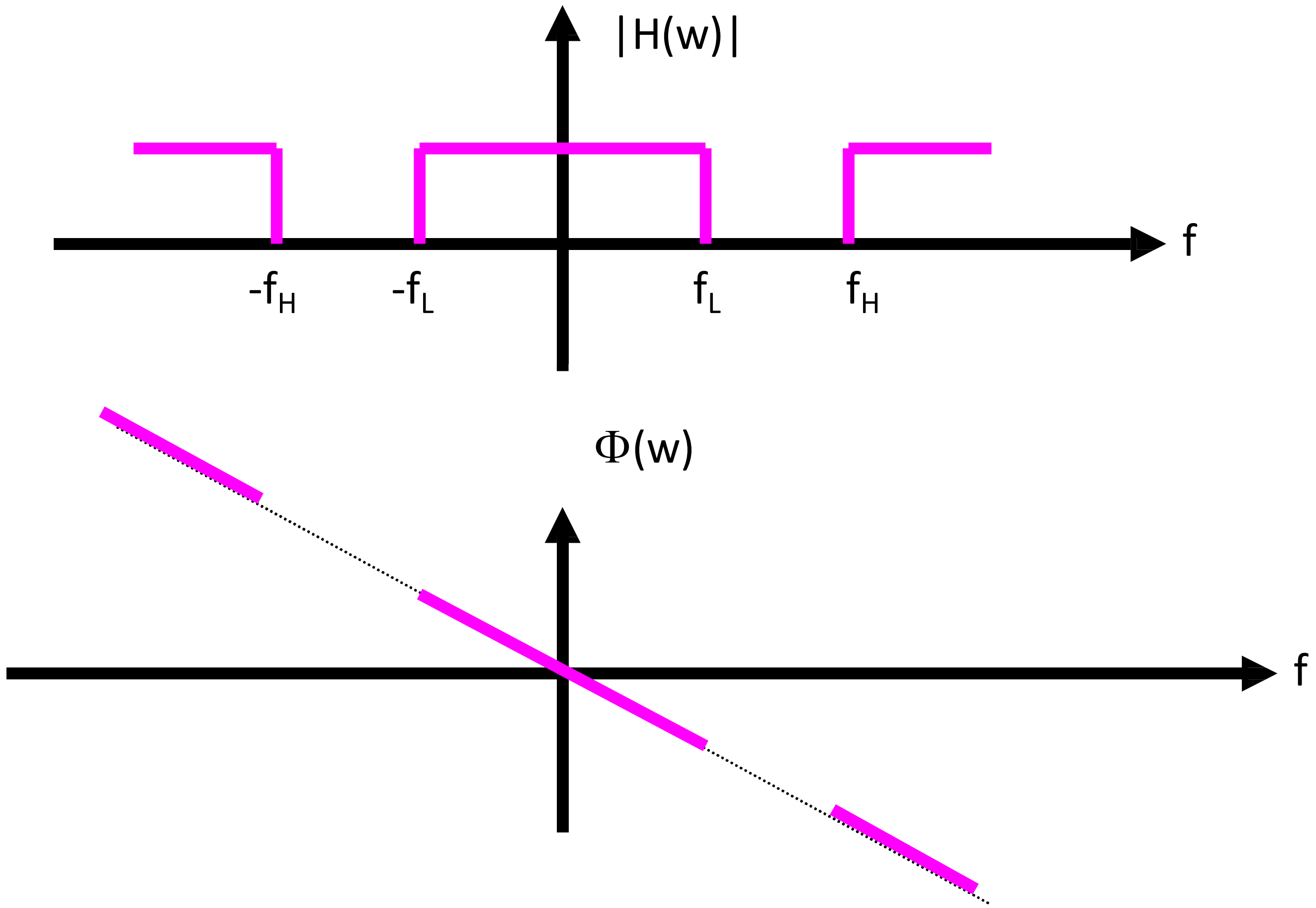
Passes frequencies higher than a cut-off

Band Pass Filter



Passes frequencies between two cut-offs

Band Stop Filter



Passes frequencies other than those between two cut-offs

Digital Filters

- Where an analog filter operates on a continuous signal, a digital filter processes a sequence of discrete sample values

- A digital filter can be a software program, a programmable hardware processor or a dedicated integrated circuit

Digital Filters

- How can Digital Filters provide frequency selectivity?
- The digital signal we get, still has a frequency definition into it
  - In terms of sampling instant
- And a digital system has a frequency response
- So, the output of that system with the input gives us the required frequency selectivity

Digital Filters vs Analog Filters

- Digital filters can have a linear phase response in the band of interest as compared to analog filters, which have non-linear phase response

- The frequency response of a Digital filter can be easily adjusted (since it is a program), however, this is not possible for an Analog filter (create a new circuit)

- The performance of Digital filters is not affected by environmental effects (heat), while it does affect the performance of Analog filters
  - Re-calibration
- Both filtered and un-filtered data can be stored
- Digital filters can work for low frequency signals unlike Analog filters

- Digital filters are affected by the finite word length effects

- The Processing capability of filter depends on the speed of the Processor running the program

Types of Digital filters

- Like any LTI system, a digital filter can be characterized by its Impulse Response (I.R.)

- Based on the IR, digital filters can be classified as
  - Finite Impulse Response (FIR) filters
    - Impulse Response becomes zero with time
    - Time limited
  - Infinite Impulse Response (IIR) filters
    - Impulse Response *does not* become zero with time
    - Time un-limited

Comparison between FIR & IIR

Finite Impulse Response	Infinite Impulse Response
Implemented non-recursively - No Feedback	Implement recursively - With Feedback
Always stable	Stability not guaranteed
Simple to implement	Difficult to implement
Linear phase response in pass-band	Non-linear phase response in pass-band

Finite Impulse Response

- Requires more filter co-effs for a specified filter than IIR
  - More memory
  - More processing power
- Does not have an analog counterpart
- Less susceptible to the effect of using a limited number of bits to implement filters such as round off noise, and coefficients quantization error

Infinite Impulse Response

- Requires less filter co-effs for a specified filter than FIR
  - Less memory
  - Less processing power
- Analog filters can be converted into IIR filters
- More susceptible to the effect of using a limited number of bits to implement filters such as round off noise, and coefficients quantization error

Finite Impulse Response filters

Use FIR if the number of filter coefficients is not too large and in particular if little or no phase distortion is desired

Infinite Impulse Response filters

Use IIR when there are requirements such as sharp cut off and high throughput

FIR Filters

- No feedback
- Output is function of the present input and the past inputs
- Output does not depend on the previous outputs

$$y[n]=b_0x[n]+b_1x[n-1]+....+b_Nx[n-N] \qquad (1)$$

$$y[n]=\sum_{i=0}^{L}b_ix[n-i]$$

- L+1 is said to be the filter length

IIR Filters

- Requires feedback
- Output is the function of the present input, the past inputs and also the past outputs

$$y[n] = b_0 x[n] + b_1 x[n-1] + \dots + b_L x[n-L] + a_1 y[n-1] + a_2 y[n-2] + \dots + a_M y[n-M] \quad (2)$$

$$y[n] = \sum_{i=0}^{L} b_i x[n-i] + \sum_{j=1}^{M} a_j y[n-j]$$

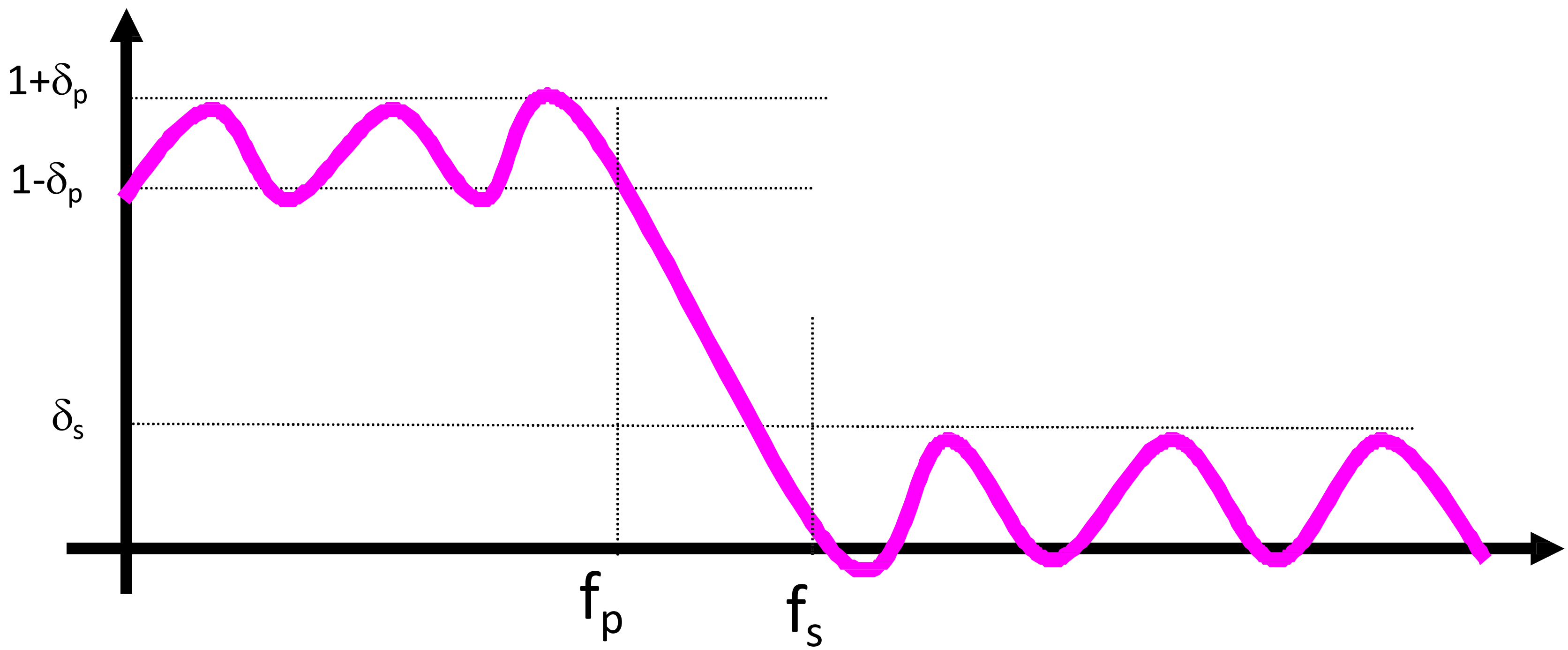
Digital Filter Design

1. Specification of the filter requirements
2. Calculation of suitable filter co-efficients
3. Representation of the filter by a suitable structure (realization)
4. Analysis of the effects of the finite word-length on filter performance
5. Implementation of filter in software/hardware

Specification of filter requirements

- Signal Characteristics
- Sampling Frequency F_s
- Highest Frequency of interest

- Characteristics of the filter (Frequency domain)
  - Magnitude Response
    - Pass band deviation (δ_p)
    - Stop band deviation (δ_s)
    - Pass band edge frequency (f_p)
    - Stop band edge frequency (f_s)



- Stop band attenuation

$$A_s = -20\log_{10}(\delta_s)$$

- Pass band ripple

$$A_p = 20\log_{10}(1+\delta_p)$$

- Transition width

- Phase Response
  - Strictly Linear
    - Which implementation would you choose?
  - Loosely Linear

A filter has to be designed to meet the following specifications

Pass band	:	0.18 – 0.33
Transition width	:	0.04
Stop band deviation	:	0.001
Pass band deviation	:	0.05
Sampling frequency	:	10 KHz
Phase requirements	:	Linear in pass band

The cutoff frequencies are given in normalized form, i.e. f/Fs, so the un-normalized frequencies are:

Pass band = 0.18x10e3 to 0.33x10e3 i.e. from 1800 to 3300 Hz

Stop band 1 = 0 to 1800 - 400 = 0 to 1400 Hz

Stop band 2 = 3300 + 400 to 5000 = 3700 to 5000 Hz

Pass band ripple = $20\log_{10}(1+0.05) = 0.42$ dB

Stop band attenuation = $-20\log_{10}(0.001) = 60$ dB

I’ve implemented the specification as an FIR filter