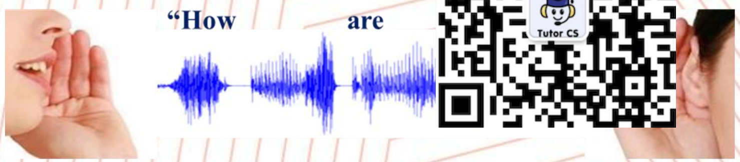
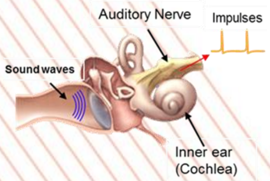
ELEC3104: Mini-Project – Cochlear Signal Processing





TLT – Level 2 (Pass Level): Implementation of an IIR cochlear parallel filter bank and then conversion to an equivalent parallel FIR filter bank model of the cochlea for analysis and synthesis purposes.

Note: This mini project involves a substantial time commitment to successfully complete all parts. It is suggested that you commence work on this project straight away.

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Term 3, 2024

Step 1: Bandpass Filter Design

Step 1: Design Criteria

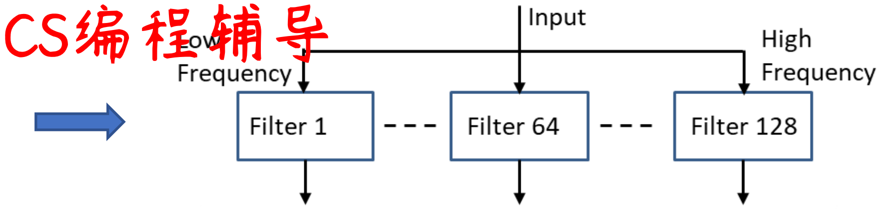
- ✓ The Cochlea can be modelled as a parallel filter bank consisting of overlapping bandpass filters covering the frequency range from 50Hz to 8kHz (Sampling frequency = 16 kHz).
- ✓ 128 bandpass filters (Resonant filters) with centre frequencies (f_p) and quality factors (Q_p) are used to model the cochlea. The bandwidth ($^{f_p}/_{Q_p}$) spanned by each of these filters increases with frequency.
- ✓ The centre frequency of the bandpass filters can be calculated using the following equation:

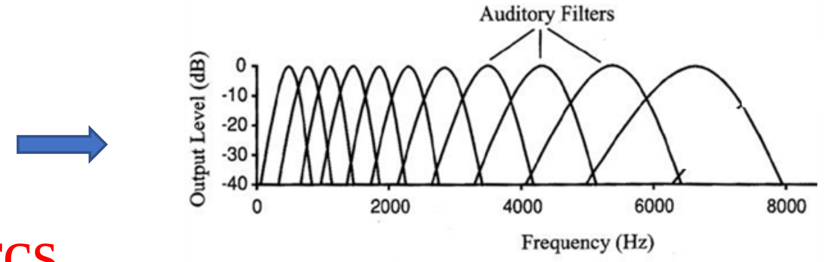
$$f_p(k) = (8000)\,10^{-0.667\,\mathrm{k}\Delta x}\ \mathrm{Hz} \qquad 0.0869 \le x \le 2.9985\ \mathrm{cm}$$

$$\Delta x = \frac{2.9985-0.0869}{N-1} \approx 0.02293\ \text{cm};\ N = 128; \quad \mathbf{k} = 1, 2, \ldots, N;$$

- ✓ From experiments, it is known that the Quality factors Q_p of the bandpass filters vary linearly from 5 (Low frequency filter, n=1) to 10 (High frequency filter, n=128)
- ✓ The bandwidth of the nth bandpass filter is given by:

$$BW_p(n) = \frac{f_p(n)}{Q_p(n)}.$$





Note: when **n**=1, **k**= 128; n =2, k =127 ….., n =128, k=1

Filter No. (n)	Resonant Frequency $f_p(n)$ in Hz	$Q_p(n)$	$BW_p(n)$ in Hz
1	$f_p(1)$ = 88	5	18
2	$f_p(2)$ =91	5.039	18
3	$f_p(3)$ =95	5.079	19
4	$f_p(4)$ =98	5.118	19
.	.	.	.
64	$f_p(64)$ =811	7.520	108
.	.	.	.
128	$f_p(128)$ =7723	10	772

Table 1

Step 1 (continued): Bandpass Filter Design - Pole-zero placement method

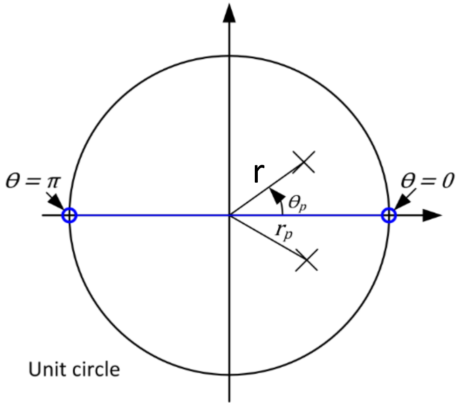
Step 1: Design Criteria (Continued)

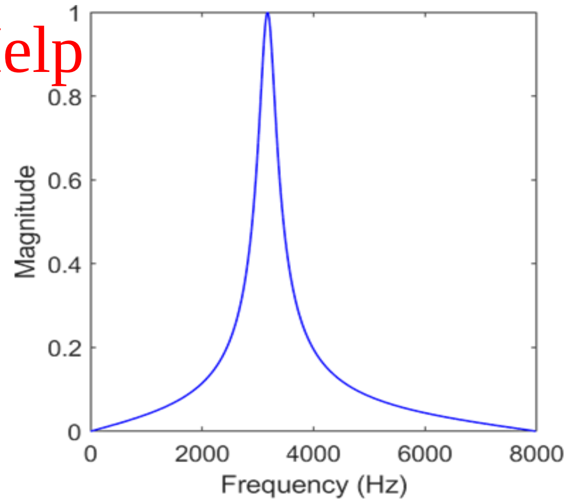
- ✓ Complete the table 1 using the equations given in the previous slide.
- ✓ Calculate the digital filter coefficients o b_1 & b_2) and zeros using pole-zero placement (e.g: see di
  - Pole frequency = θ_p; radius = r_p; ($= 16\ kHz$
  - $H_p(z) = \frac{z^2-1}{\left(z-r_p e^{j\theta_p}\right)\left(z-r_p e^{-j\theta_p}\right)} = \frac{1-z^{-2}}{1-2r_p\cos\theta_p z^{-1}+r_p^2 z^{-2}}$ ----- (1)
  - One section of the digital filter: $H_p(z) = \frac{1-z^{-2}}{1-b_1 z^{-1}+b_2 z^{-2}}$ ----- (2)
  - Equating the denominator of (1) and (2), we obtain
  - $b_1 = 2r_p\cos\theta_p$ and $b_2 = r_p^2$ ----- (3)
  - r_p and BW_p (bandwidth of H_p) can be calculated as follows:

$$r_p \approx 1 - \left(\frac{BW_p}{f_s}\right)\pi\ ; \qquad BW_p = \frac{f_p}{Q_p}\ ; \quad Q_p \text{ - Q factor}$$

- ✓ Calculate the coefficients b_1 & b_2 using Equation 3 above.
- ✓ Repeat the above calculations to obtain all 128 IIR filters $\{H_p(z)\}$.

Complex Pole pair and zeros placement





Magnitude response of a bandpass filter with centre frequency 3171 Hz

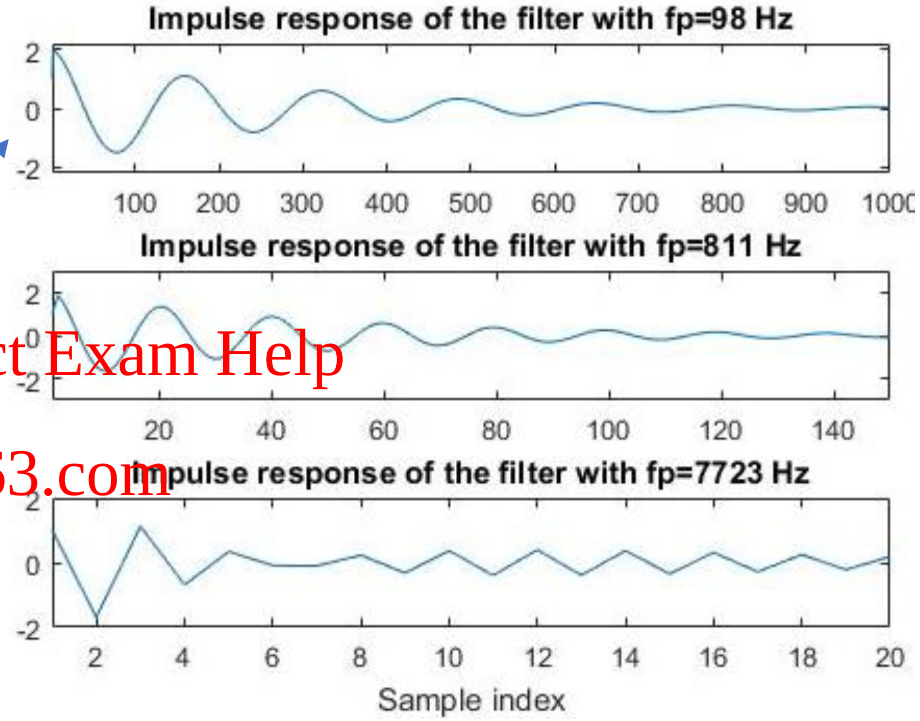
Are you on the right track?

Are you on the right track?

✓ If your implementation is on the right track, you should observe the following impulse responses (roughly) at different IIR filters. You must also check the coefficients (b_1 and b_2) that you have calculated against Table 2.

Filter No. (n)	Resonant Frequency $f_p(n)$ in Hz	$Q_p(n)$	$BW_p(n)$ in Hz	$b_1 =$ [illegible]	r_p^2
1	$f_p(1)$ = 88	5	18	1.9[illegible]	[illegible]93 (3DP)
2	$f_p(2)$ =91	5.039	18	1.9916	0.993
3	$f_p(3)$ =95	5.079	19	1.9913	0.993
4	$f_p(4)$ =98	5.118	19	1.9910	0.992
.	.	.	.		
64	$f_p(64)$ =811	7.520	108	1.8589	0.958
.	.	.	.	.	.
128	$f_p(128)$ =7723	10	772	-1.6867	0.720

Table 2

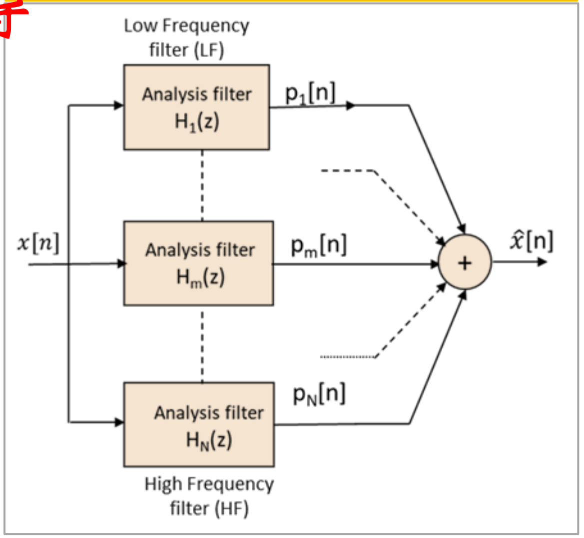


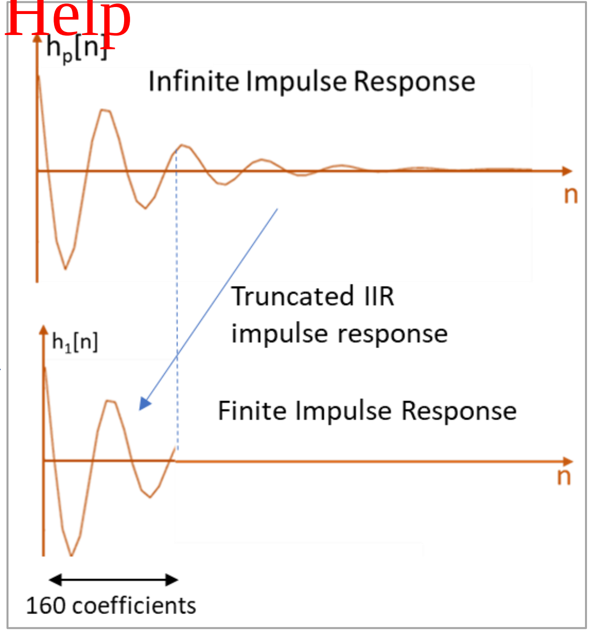
Step 2: Analysis Filter Bank

Step 2: FIR Analysis Filters

- ✓ In a filter bank analysis, an input signal is fed into a series of analysis filters (in this project we use $N = 128$ filters), decomposing the signal into subbands.
- ✓ This means that the output signal ($p_m[n]$) f[illegible]ysis filter represents different frequency component[illegible]signal.
- ✓ The analysis filters $H_m(z)$ (see diagram on the right) comprise a set of overlapping N bandpass filters used to model the human auditory filters as explained in the previous slides.
- ✓ The analysis filter $H_m(z)$ (impulse response h[n]) can be implemented in practice as an FIR filter.
- ✓ Once you have calculated the filter coefficients of the 128 IIR filters, use MATLAB to calculate the impulse responses of your IIR filters (theoretically the IIR impulse response should be of infinite length but in MATLAB, it is truncated between 60 and 2852 samples depending on centre frequency of the IIR bandpass filter)
- ✓ Each IIR filter's impulse response should be truncated to 160 samples, in order to obtain the FIR filter coefficients of your analysis filters $\{H_1(z)$ to $H_N(z)\}$ {see an example shown on the right). In the case where the IIR filter impulse response is less than 160 samples, padding with zeros is required.
- ✓ Using these coefficients, the filters $\{H_1(z)$ to $H_N(z)\}$ can now be implemented as a bank of Parallel FIR filters.

Block Diagram of Analysis Filter Bank



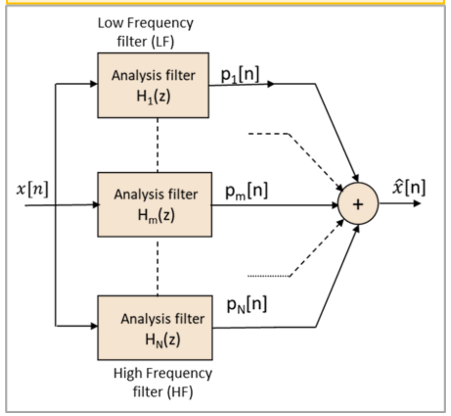


Step 2 (continued) : Analysis Filter Bank

Step 2: FIR Analysis Filters

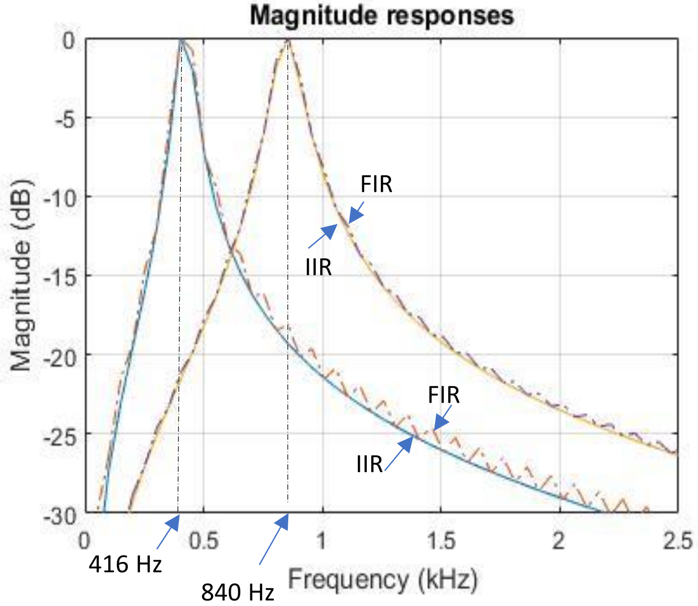
- ✓ In a filter bank analysis, an input signal is fed into a number of analysis filters (in this project we use $N = 128$ FIR filters, see the diagram on the right), decomposing the signal into subbands.
- ✓ This means that the output signal from each an $\ldots_1[n], p_2[n], \ldots, p_N[n]\}$, represents different frequency components of the input si

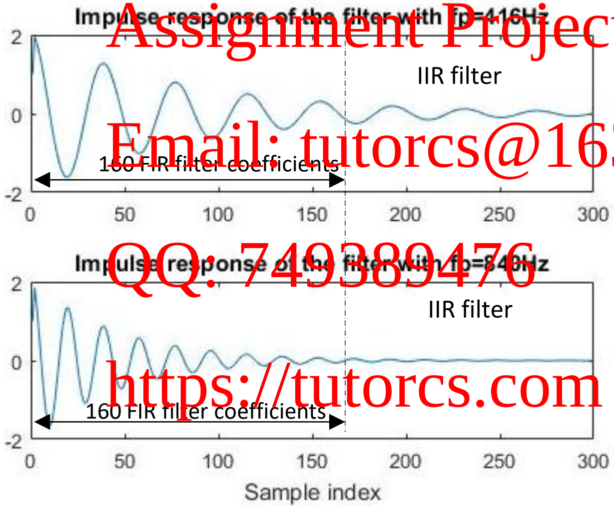
Block Diagram of Analysis Filter Bank

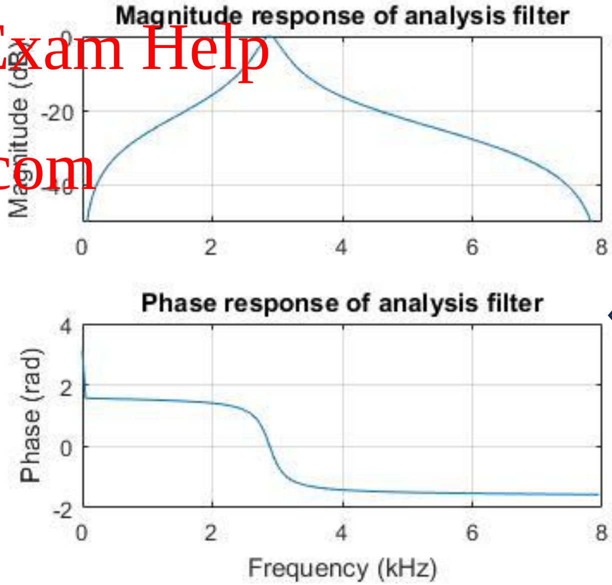


Are you on the right track?

- ✓ If your implementation is on the right track, you should observe the magnitude responses (roughly) for the IIR filters 45 (416Hz)and 65 (840Hz) as in the diagram below. The corresponding FIR filter magnitude responses are also shown below.







Magnitude & phase responses of the analysis filter number 100 corresponding to the centre frequency of 2882 Hz.

Note: The phase response is non-linear.

Step 3: Analysis Filter Bank Implementation

Step 3: Implementation

- ✓ For the filter with a centre frequency of 1 kHz, calculate the IIR filter impulse response h[n], truncate it to M (=160) coefficients and plot the magnitude response of the filter using these coefficien[illegible]

- ✓ On the same figure, plot the magnitude res[illegible]corresponding 1kHz IIR filter. You may notice that the magnitud[illegible]r the filters are not exactly the same. Discuss the reasons for th[illegible]ould be given with a centre frequency of the filter normalised to 0dB.

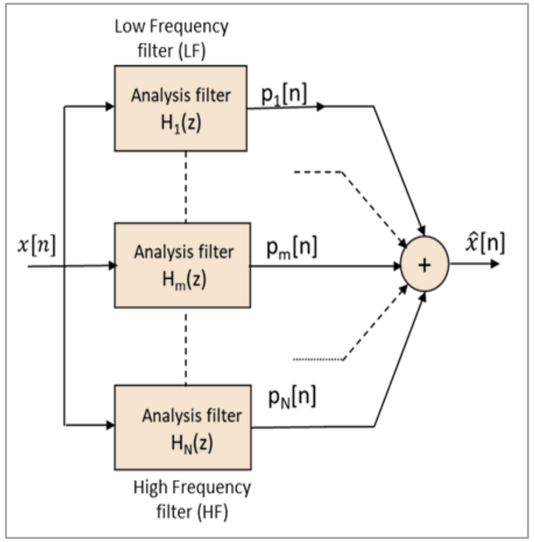
- ✓ Using MATLAB, plot the magnitude responses of the analysis filters (you may select 10 filters). The plot should be given with a centre frequency of each filter normalised to 0dB.

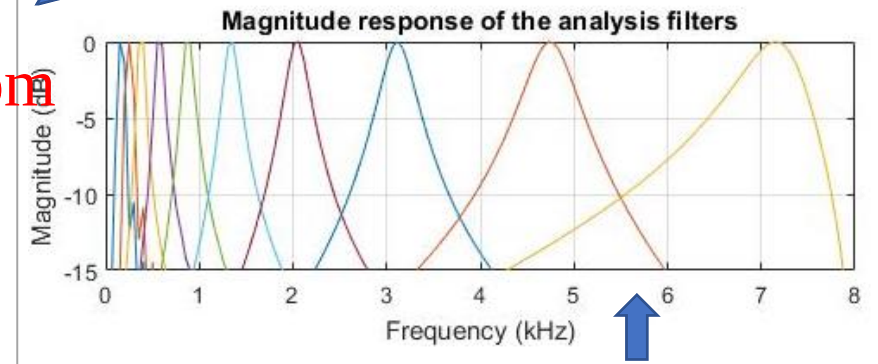
- ✓ Using MATLAB, filter the speech signal (saved in **speech.wav** and **music.wav** from 'Wav files for DSP Labs' on Moodle) sampled at 16 kHz using the analysis filter bank. The outputs of these analysis filters can then be added ($\sum_{m=1}^{N} p_m$[n]) to produce synthesised speech/music

- ✓ Using the *soundsc* command in MATLAB, **compare** the quality of the reconstructed speech/music $\hat{x}$[n] (=$\sum_{m=1}^{N} p_m$[n]) with that of the original $x[n]$. **Discuss** any noticeable differences. You can also use **PESQ** software to compare the quality of the reconstructed waveform with the original.

- ✓ Now increase the FIR filter coefficients to 300 and repeat the above two steps to produce synthesised speech/music . **Discuss** any noticeable differences.

Block Diagram of the Analysis Filter Bank





Centre frequencies of the above 10 filters:
161 Hz 245 374 570 870 1328
2026 3092 4718 7198 Hz

Step 4: Synthesis Filter Bank

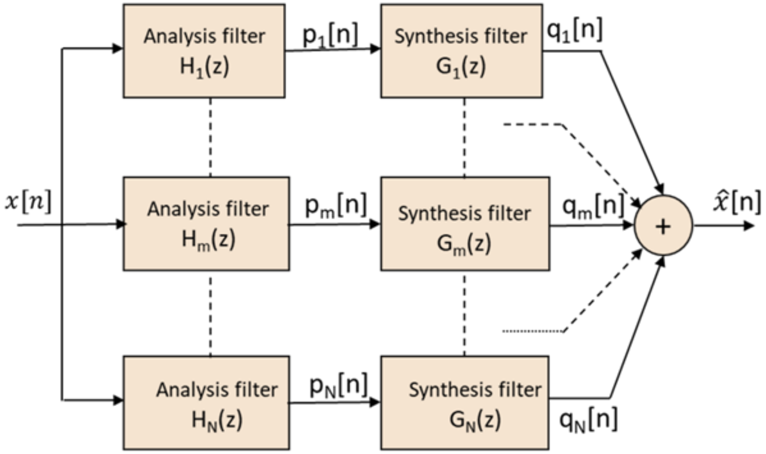
Step 4: FIR Synthesis Filters

- From step 3, you may have noticed that the reconstruction of the input signal $x[n]$ is achievable, by simply summing the analysis filter outputs ($\sum_{m=1}^{N} p_m$[n]).
- If a linear phase property is required, we ne ce synthesis filters in each channel (see figure on the rig
- The analysis/synthesis filter pair for each ch means independent. In fact, by specifying either the analysis or the synthesis filter response, the linear phase requirement implicitly determines the other.
- For a linear phase system, we need a synthesis filter $G_m(z)$, in each channel. The synthesis filter is a time reversed impulse response of the analysis filter, $H_m(z)$.
- By time reversing the FIR analysis filters' impulse responses, we can form the synthesis filters (Impulse response $h[-n]$).
- In this project, we choose the synthesis filters {G_1(z) to G_N(z)} to be the time-reversed version of the corresponding analysis filters i.e.

$$g_m[n] = h_m[-n] \quad \text{for m = 1, 2....N}$$

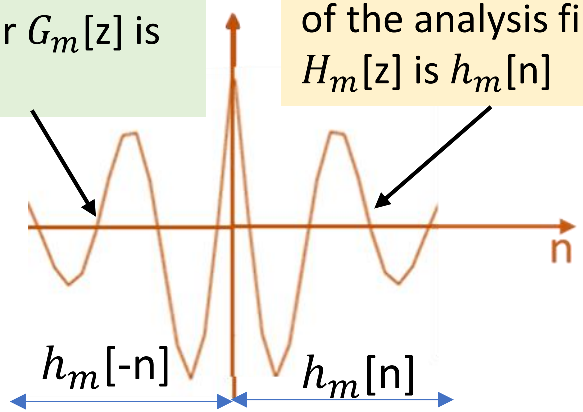
- Each analysis filter bank output is fed to the input of the corresponding synthesis filter. The outputs of these synthesis filters can then be added to produce the synthesised speech i.e. $\hat{x}$[n] (=$\sum_{m=1}^{N} q_m$[n]).

Analysis/Synthesis Filter Bank



Impulse response of the Synthesis filter G_m[z] is h_m[-n]

Impulse response of the analysis filter H_m[z] is h_m[n]



Step 4 (continued): Synthesis Filter Bank

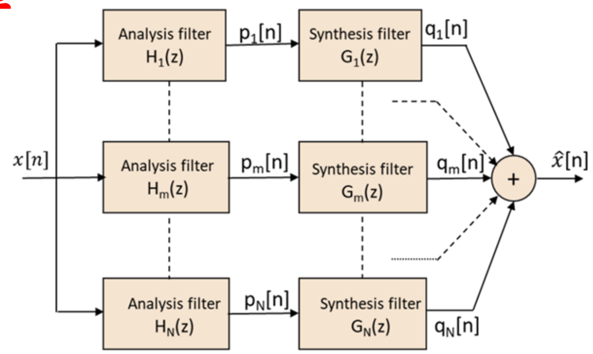
Step 4: FIR Synthesis Filters

- ✓ The impulse responses of the synthesis filters are a time-reversed version of the corresponding analysis filters, therefore the filters are non-causal. To make the synthesis filters causal, a delay is e.

$$g_m[n] = h_m[L-n]$$ for m = 1, 2... e delay

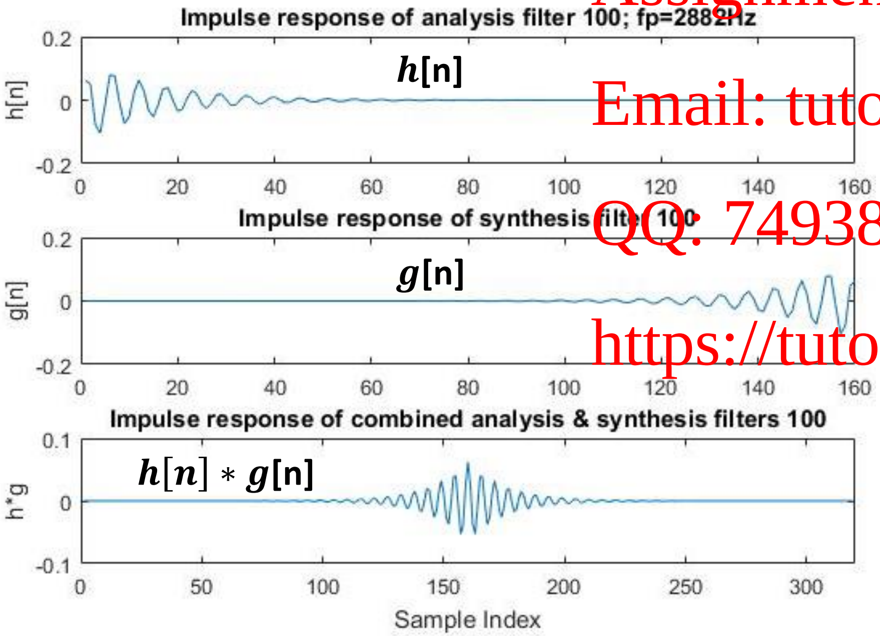
- ✓ In our example, we use 160 coefficients, th 0.

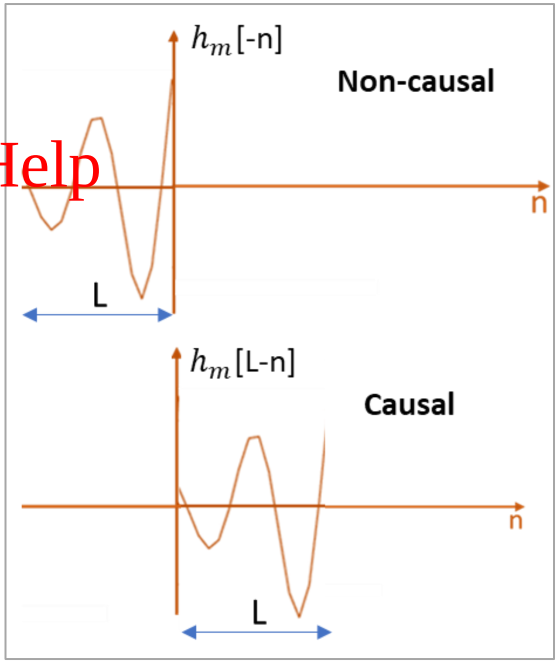
Analysis/Synthesis Filter Bank



Are you on the right track?

- ✓ If your implementation is on the right track, you should obtain the following impulse responses (roughly) for the analysis, synthesis, and the combination of analysis & synthesis FIR filters for subband 100 (f_p=2882Hz):





Are you on the right track?

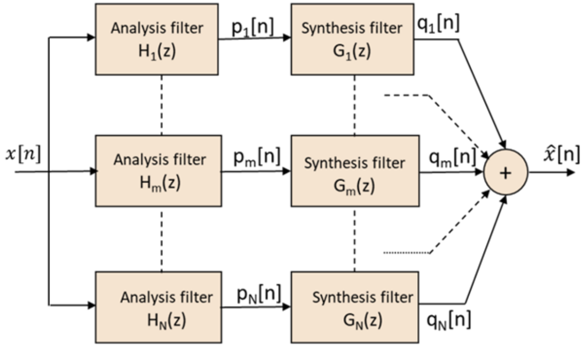
Are you on the right track?

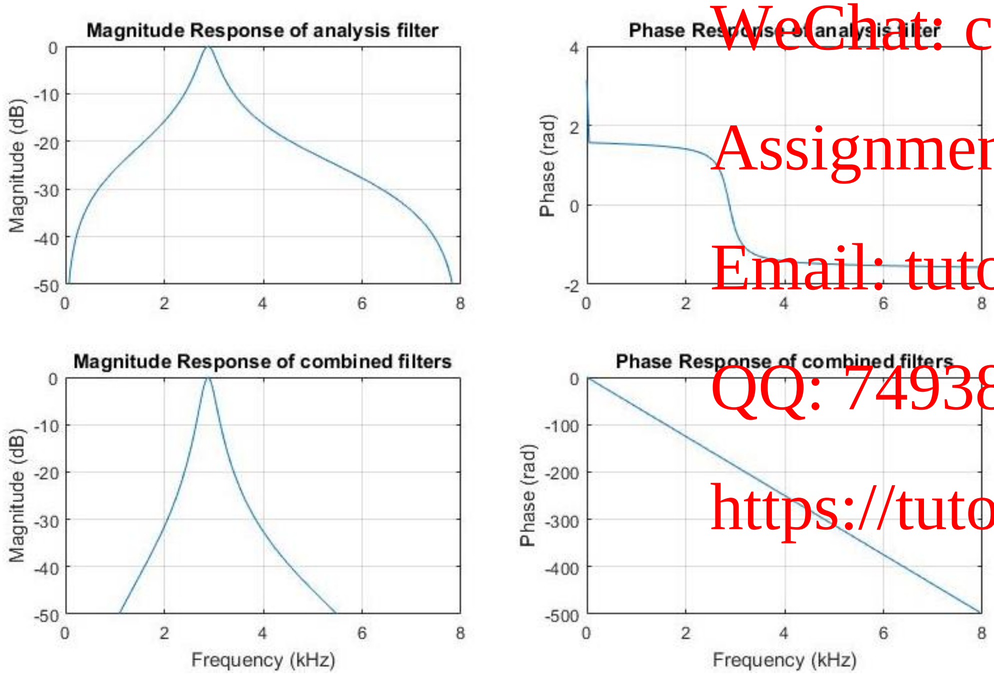
✓ If your implementation is on the right track, you should obtain the following:

(a) Magnitude and phase responses (roughly) for the analysis, synthesis, and the combination of analysis & synthesis FIR fi[illegible]nd 100 (f_p=2882 Hz)

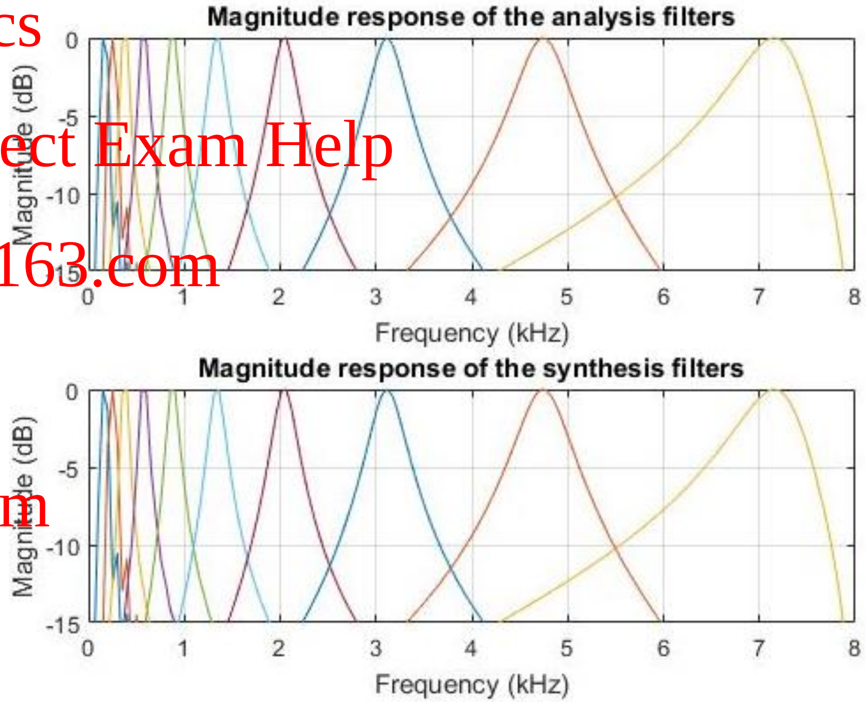
(b) Magnitude responses of the analysis and [illegible]ers (10 filters). The centre frequency of each filter is normali[illegible]ntre Frequencies are:

161 Hz 245 Hz 374 570 870 13[illegible] [illegible]092 4718 7198 Hz





(a)



(b)

Step 5: Analysis/Synthesis Filter Bank Implementation

Step 5: Implementation

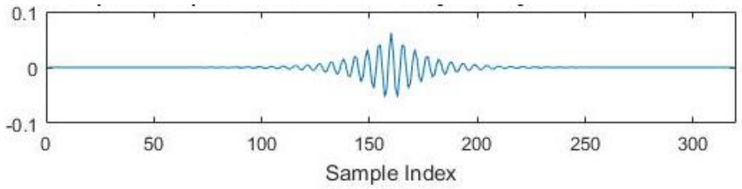
- ✓ For the synthesis filter with a centre frequency of 1 kHz, plot the magnitude response and on the same figure, plot the magnitude response of the corresponding analysis filter. Do you notice any difference? Explain what you no should be given with a centre frequency of a filter normalised to 0dB.
- ✓ Using MATLAB, filter the speech signal (save av and music.wav from 'Wav files for DSP Labs' on Moodle) sampled at 16 k alysis filter bank. The outputs of these synthesis filters can then be added ($\sum_{m=1}^{N} q_m$[n]) to produce synthesised speech/music.

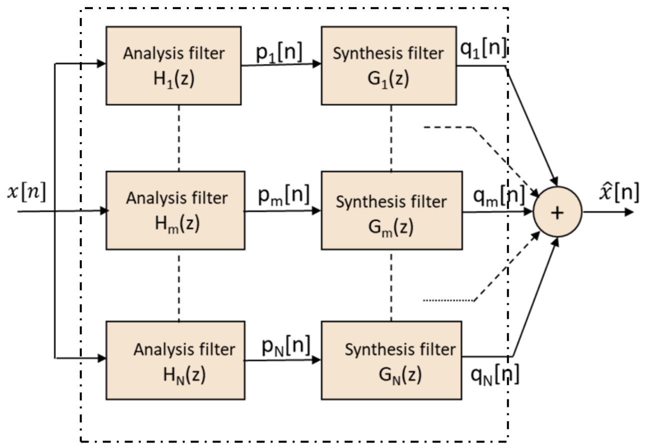
- ✓ Using the *soundsc* command in MATLAB, **compare** the quality of the reconstructed speech/music $\hat{x}$[n] (=$\sum_{m=1}^{N} q_m$[n]) with that of the original $x[n]$. **Discuss** any noticeable differences. Explain what you notice? You can also use **PESQ** software to compare the quality of the reconstructed waveform with the original.

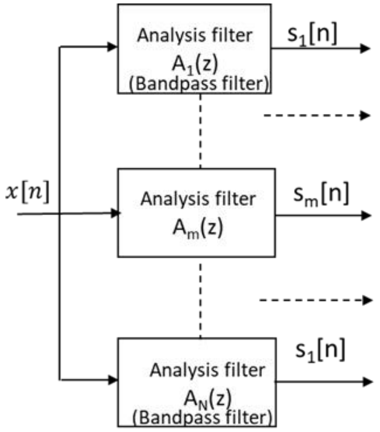
- ✓ Now increase the FIR filter coefficients to 300 and repeat the above steps to produce synthesised speech/music. Discuss any noticeable differences.

Note:

- ✓ By convolving $h_m[n] * g_{m[n]}$, we can form a new analysis FIR filter bank (see diagram on the right) with a linear phase characteristic. As discussed before, the FIR filter coefficients of all the filters ($A_1[z]$, $A_1[z]$, … $A_m[z]$ .. $A_N[z]$) are symmetrical around n= 160, as given below:







A new analysis filter bank (linear phase FIR filter)