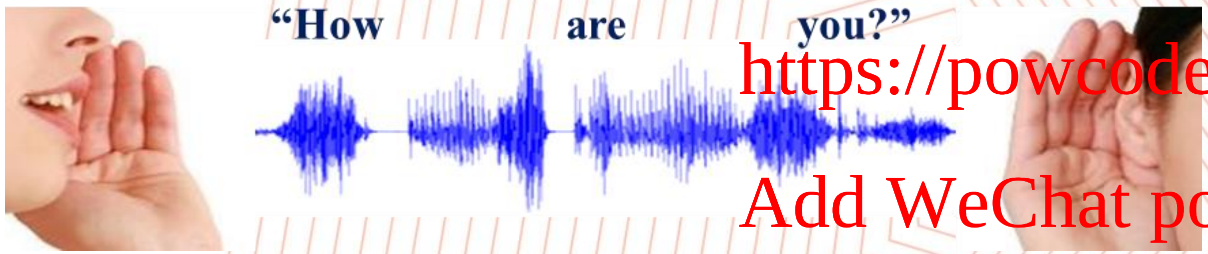
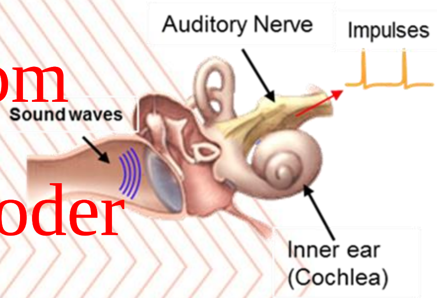
ELEC3104: Mini-Project – Cochlear Signal Processing





Prof. Eliathamby Ambikairajah, School of EE&T

Term 3, 2022

UNSW SYDNEY

ELEC3104: Project Outline

- ✓ This mini project (**individual**) will focus on understanding and modelling the spectral analyses carried out by the human cochlea.

TLT – Level 1

- Introduction to Human Auditory System and MATLAB coding fundamentals

TLT – Level 2 (Pass Level)

- Implementation of a cascaded filter bank model of the cochlea for analysis purposes.

TLT – Level 3 (Credit Level)

- Implementation of a cascaded filter bank model of the cochlea for spectral analysis.

TLT – Level 4 (Distinction Level)

- Implementation of a cascaded filter bank model of the cochlea for for pitch detection of a speech signal.

TLT – Level 5 (High Distinction Level)

- Incorporate mechanisms into the cascaded cochlear model that makes the cascaded filter bank adaptive.

Additional Information: In addition to the information provided to you in these slides, you are strongly encouraged to find and view animations and videos that describe the functioning of the peripheral auditory system and the cochlea in particular. Visualisation in the form of these animations will be very helpful in understanding cochlear signal processing.

Eg: Cochlear Animation - https://www.youtube.com/watch?v=dyenMluFaUw

ELEC3104: Mini-Project – Cochlear Signal Processing

TLT – Level 1: Introduction to Human Auditory System and MATLAB coding fundamentals

Prof. Eliathamby Ambikairajah, School of EE&T

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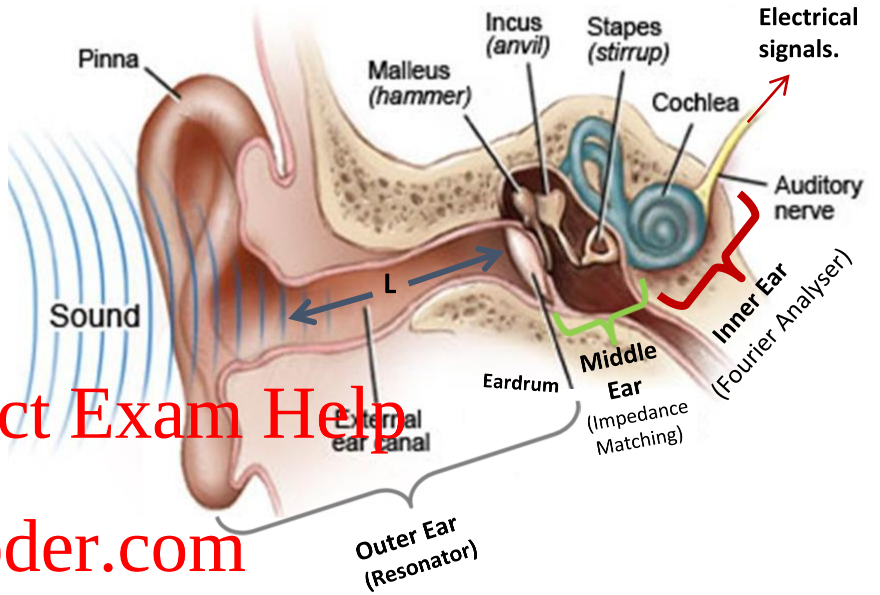
Introduction to the Human Auditory System

- ✓ The human auditory system is responsible for converting pressure variations caused by the sound waves that reach the ear into nerve impulses that are interpreted by the brain.
- ✓ The Human Auditory System is designed to assess frequency (pitch) and amplitude (loudness).
- ✓ The peripheral auditory system is divided into the Outer Ear, Middle Ear, and Inner Ear.
- ✓ The peripheral auditory system and in particular the cochlea can be viewed as a real-time spectrum analyser.
- ✓ The primary role of the cochlea is to transform the incoming complex sound wave at the ear drum into electrical signals.
- ✓ The human ear can respond to minute pressure variations in the air if they are in the audible frequency range, roughly 20 Hz - 20 kHz

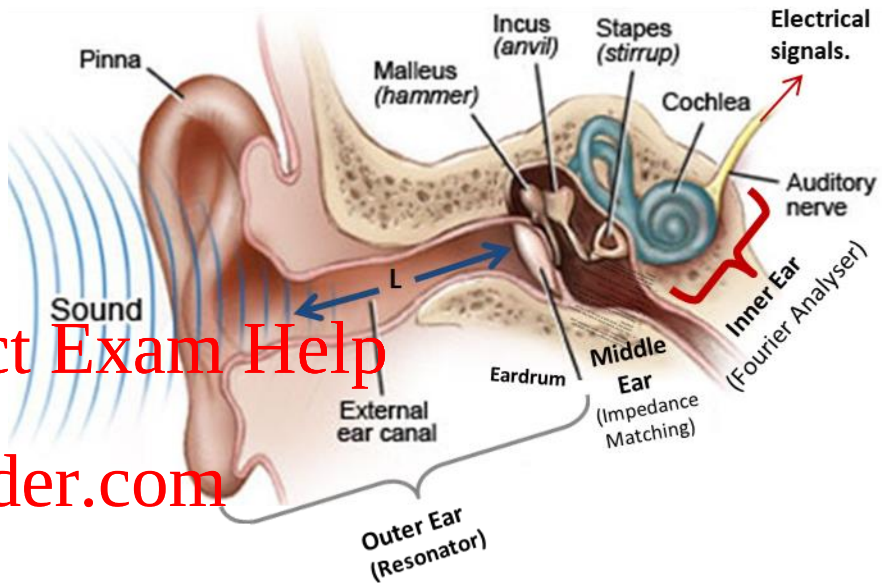


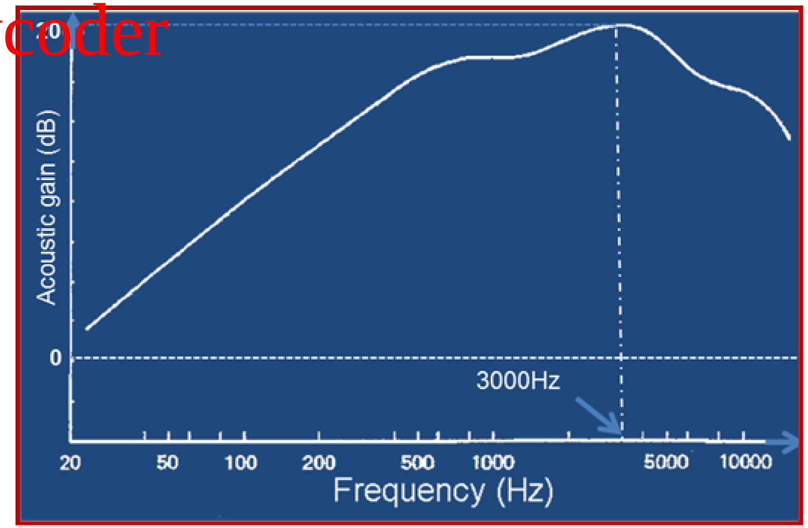
Sounds	Level
Faint	20dB (A faint Whisper is 30dB)
Soft (Quiet)	40dB
Moderate	60dB (normal conversation)
Loud	80dB (alarm clocks, vacuum cleaners)
Very Loud	90dB(Blenders);110dB (Concerts, car horns)
Uncomfortable	120dB (jet planes during take off)
Painful and dangerous	130dB(Jackhammers); 140dB(Gunshots) *Use hearing protection

- ✓ Over 85 dB for extended periods can cause permanent hearing loss
- ✓ Zero decibels (0 dB) represent the absolute threshold of human hearing, below which we cannot hear a sound.

Outer Ear (Air Vibration): A resonator

- ✓ The pinna surround the ear canal and functions as sound wave reflectors and attenuators .
- ✓ The sound waves enter a tube-like structure called ear canal and it serves as a sound amplifier.
- ✓ The sound waves travel through the canal and reach the eardrum and cause it to vibrate
- ✓ The length (L) of the human ear canal is 2.8 cm (and 7 mm in diameter)
- ✓ Speed of sound (c) = 340.3 m/sec ;
- ✓ The resonant frequency (f) of the canal is = $\frac{c}{4L}$ = 3,038Hz.
- ✓ The human outer ear is most sensitive at about 3kHz and provides about 20dB (decibels) of gain to the eardrum at around 3000Hz.

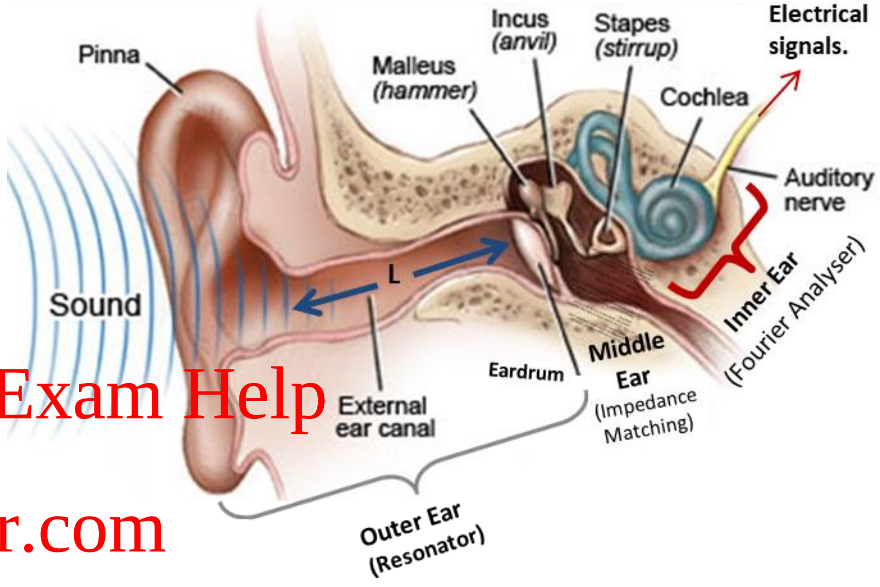


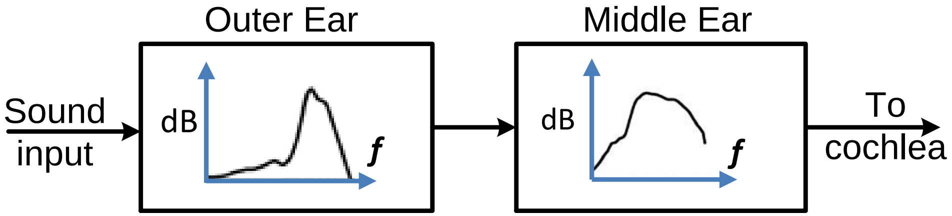


Outer ear is a low-Q bandpass filter (Representative figure only)

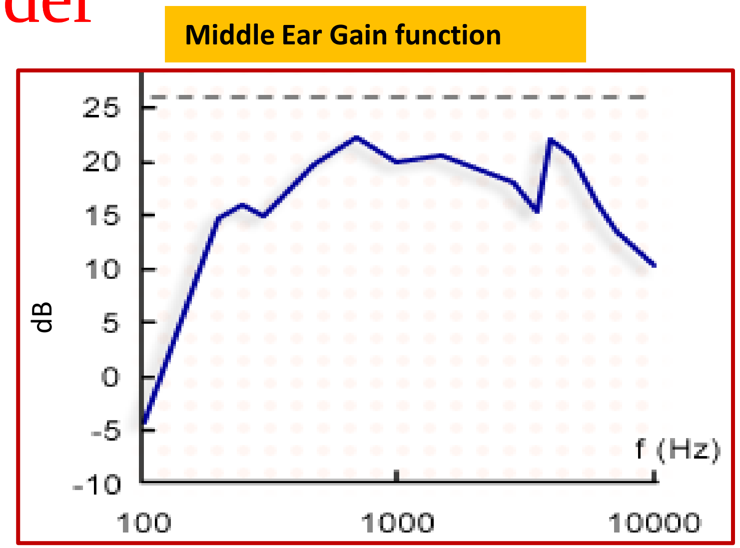
Middle Ear: An Impedance Matcher & an Amplifier

- ✓ Middle ear transforms the vibrating motion of the eardrum into motion of the stapes via the two tiny bones, the malleus and incus .
- ✓ The pressure of the sound waves on the oval window is around 25 times higher than on the eardrum.
- ✓ Since the sound Intensity (I) is proportional ($\propto$) to the square pressure (P^2) , the sound intensity increases 625 times (or 28dB)
- ✓ Middle ear converts acoustic energy to mechanical energy and mechanical energy to hydraulic energy



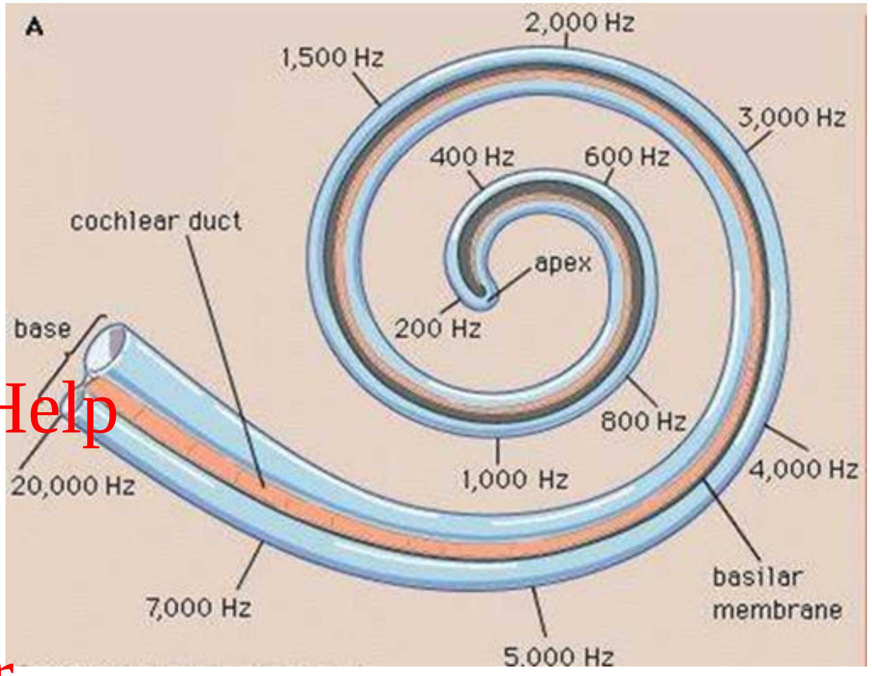


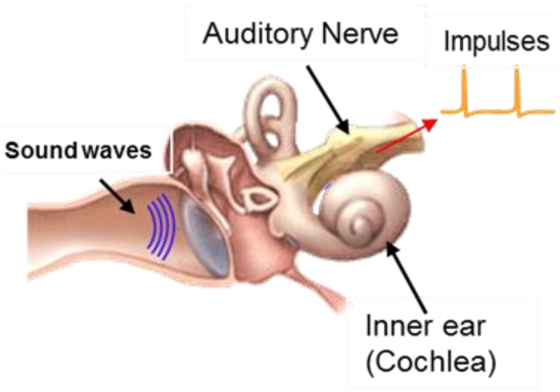
The combined frequency response of the outer and middle ear is a band-pass response, with its peak dominated near 3 kHz

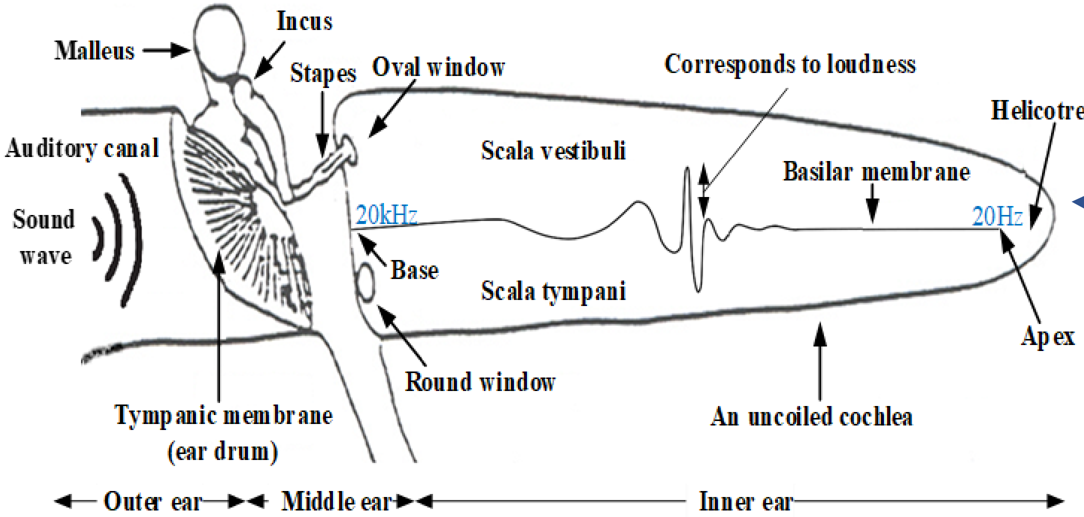


Inner Ear

- ✓ The **inner ear** consists of the cochlea responsible for converting the vibrations of sound waves into electrochemical impulses which are passed on to the brain via the auditory nerve.
- ✓ The cochlea is a spiral shaped structure which is about 3.5 cm in length if uncoiled.
- ✓ The cochlea is divided along its length by the basilar membrane (BM) which partitions the cochlear into two fluid canals (scala vestibuli and scala tympani).
- ✓ The BM terminates just reaching the helicotrema, so there is a passage way between the scala vistibuli and the scala tymapni equalising the difference in pressure at the ends of the two scalas.



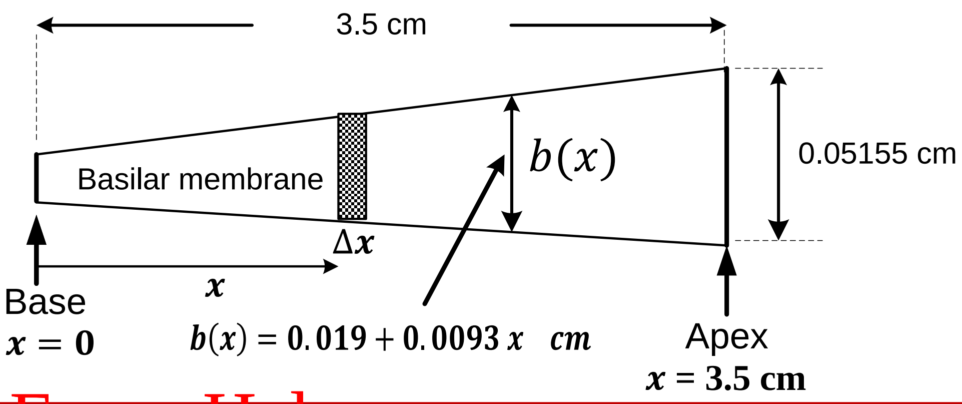




A longitudinal section of an uncoiled cochlea

Basilar Membrane (Hydro Dynamical process)

- ✓ The Basilar Membrane varies in width and stiffness along its length.
- ✓ At basal end it is narrow and stiff where as towards the apex it is wider and more flexible.
- ✓ Each point along the basilar membrane has a characteristic frequency, $f_p(x)$, to which it is most responsive.
- ✓ The maximum membrane displacement occurring at the basal end for high frequencies (20 kHz) and at the apical end for low frequencies (70Hz) .
- ✓ When the vibrations of the eardrum are transmitted by the middle ear into movement of the stapes, the resulting pressure differences between the cochlear fluid chambers, generate a travelling wave that propagates down the cochlea and reach maximum amplitude of displacement on the basilar membrane at a particular point before slowing down and decaying rapidly
- ✓ The location of the maximum amplitude of this travelling wave varies with the frequency of the eardrum vibrations

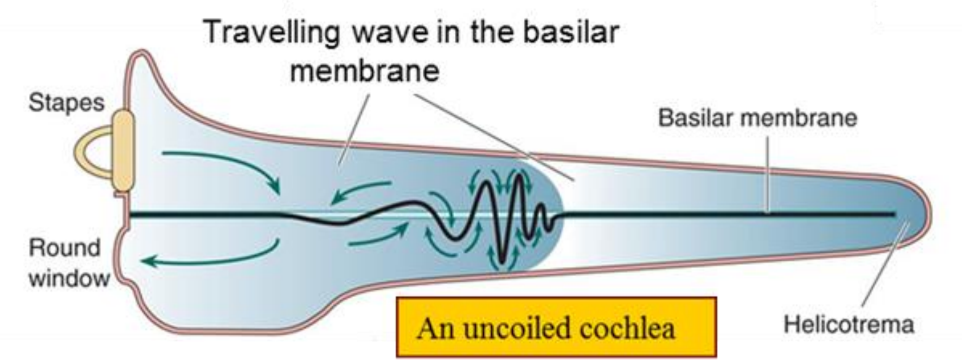


Unrolled basilar membrane

If x is the distance of a point on the basilar membrane from the stapes, then the frequency, $f_p(x)$, that produces a peak at this point is given by:

$$f_p(x) = (20000.0)\,10^{-0.667\,x}\ Hz \qquad 0 \le x \le 3.5\ cm$$

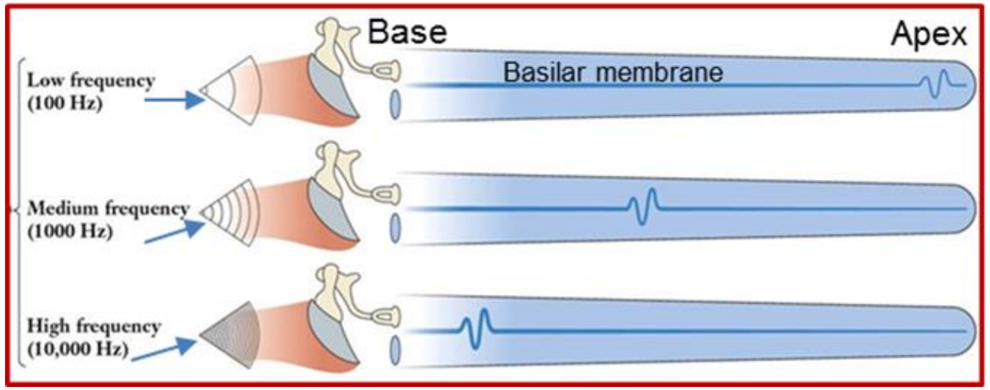
- It is evident that a 20 kHz tone at the stapes will cause the BM to vibrate at a point $x\ = 0$.
- A 70 Hz tone will excite the BM at a point x = 3.5 cm (i.e. at the apex)

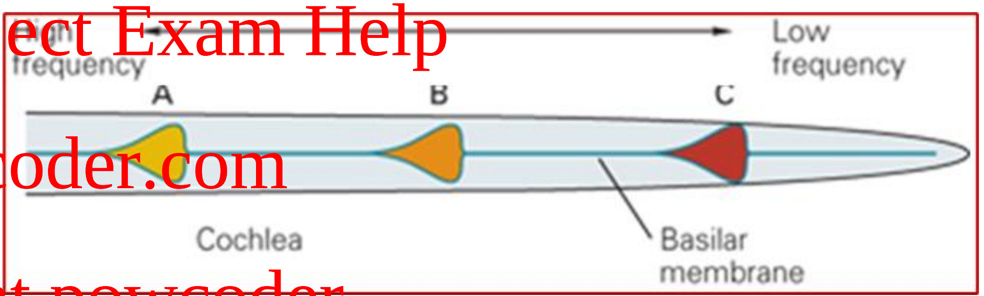


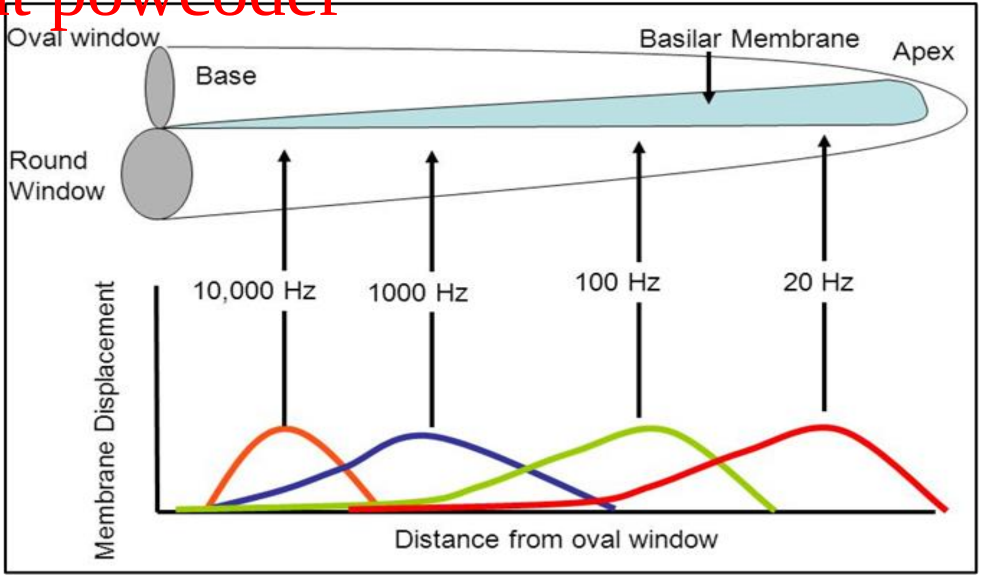
The basilar membrane is a resonant structure that vibrates, vertically in sympathy with pressure variations in the cochlear fluid.

Basilar Membrane

- ✓ Different frequencies stimulate different areas of the basilar membrane
- ✓ When a tone (single sinusoid) is applied, the cochlear fluid oscillates in phase with the stimulating frequency causing a travelling wave pattern of the vibration on the basilar membrane
- ✓ There will be one place where the resonant frequency of the membrane matches the stimulus frequency and this place will show the maximum amount of vibration
- ✓ By measuring vibration at particular points on the membrane for a range of stimulus frequencies we can plot the frequency response of each place on the membrane
- ✓ The essential function of the basilar membrane is to act as a frequency analyser (a set of band-pass filters each responding to a different frequency region) resolving an input sound at the eardrum into its constituent frequencies

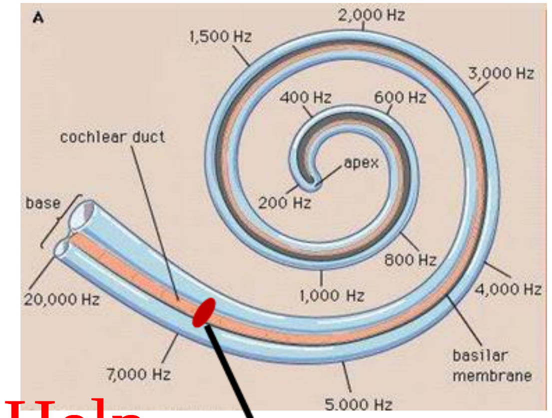




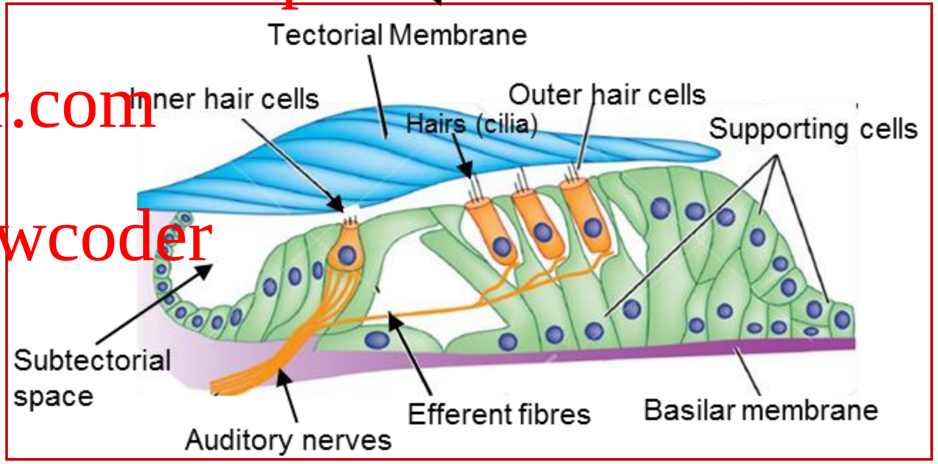


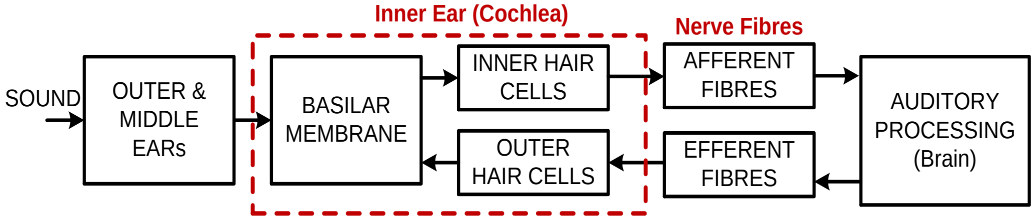
Organ of Corti

- ✓ Attached to the basilar membrane and running its entire length is the organ of corti containing some 30,000 sensory hair cells.
- ✓ The hairs (cilia) of these cells stick up from the organ of corti and are in contact with overlying Tectorial Membrane
- ✓ There are two types of sensory hair cells:
  - One row of inner hair cells, whose cilia float freely in the fluid-filled region called subtectorial space
  - Three rows of outer hair cells whose cilia are attached to the tectorial membrane
- ✓ Most of the afferent fibres (neurons which carry signals to the brain) come from inner hair cells,
- ✓ The efferent fibres (which receive signals from the brain) go mainly to outer hair cells.
- ✓ When the basilar membrane deflects, due to pressure wave in the cochlear fluid, the tectorial membrane move and shear which causes the hairs of the outer hair cells to bend and also cause the fluid flow in the subtectorial space.
- ✓ This in turn triggers the inner hair cells to transmit nerve impulses along the afferent fibres and eventually to brain.
- ✓ The motion of each part of the basilar membrane as detected by the inner hair cells is transmitted as neural description to the brain.



Organ of Corti

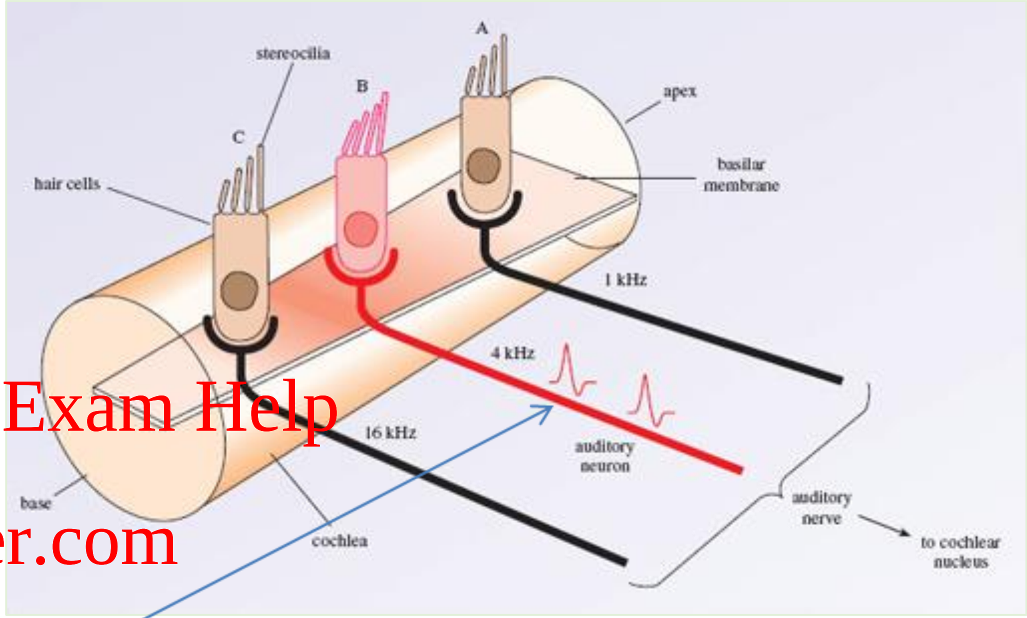




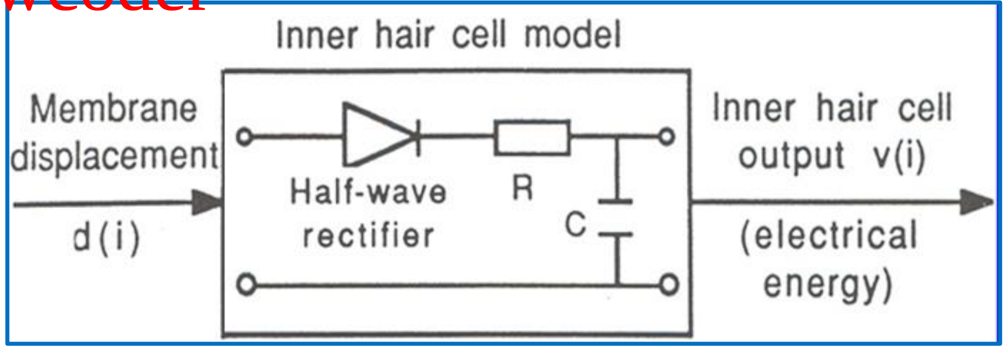
A simplified Diagram of a Human Auditory System

Mechanical to Neural Transduction (Electro Chemical)

- ✓ The mechanical displacement to electrical energy transduction process takes place in the inner hair cells
- ✓ Bending of the inner hair cell cilia due to basilar membrane displacement produces a change in the overall resistance (reduces it) of the inner hair cell, thus modulating current flow through the hair cell.
- ✓ The modulation being directly proportional to the degree of bending of the cilia and the bending of the cilia is one direction only; in effect a half wave rectification of the basilar membrane displacement takes place.
- ✓ Bending of the cilia releases neurotransmitter which passes into synapses of one or more nerve cells which fire to indicate vibration
- ✓ The amount of firing is thus related to the amount of vibration
- ✓ Since the neurotransmitter is only released when the cilia are bent in one direction, firing tends to be in phase with basilar membrane movement



Inner Hair Cell model



Here bending the inner hair cell cilia is simulated by charging of the capacitor and returning to the initial position of the cilia is equivalent to discharging the capacitor.

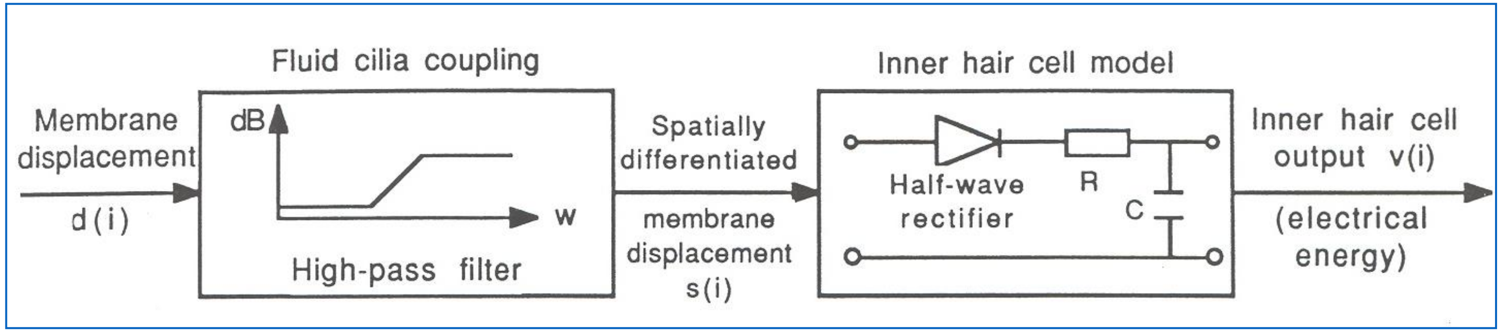
Mechanical to Neural Transduction (Electro Chemical)

- ✓ The model inner hair cell is a capacitor model, in which the input voltage corresponds to the spatially differentiated membrane displacement output of the auditory model. (Second part of Figure below). Here bending the inner hair cell cilia is simulated by charging of the capacitor and returning to the initial position of the cilia is equivalent to discharging the capacitor.
- ✓ Spatially differentiation refers to taking the derivative with respect to position (along the basilar membrane) and a discrete model is given by:

$$s[i] = \frac{d[i+1] - d[i]}{\Delta x_i}$$

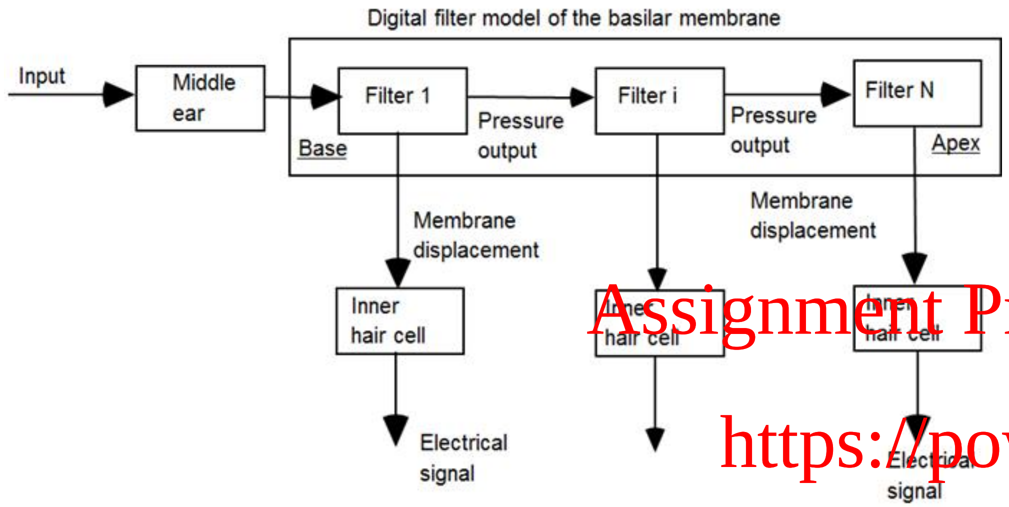
where, $d[i]$ is the displacement at the i^{th} section along the membrane and Δx_i is the width of the i^{th} section

- ✓ Spatial differentiation of the membrane displacement represents coupling between the cilia of the inner hair cells , through the fluid in the subtectorial space (high-pass filter effect, first part of Figure below)
- ✓ You will implement a digital model for neural transduction in **TLT Level 2** in addition to the transmission line auditory model

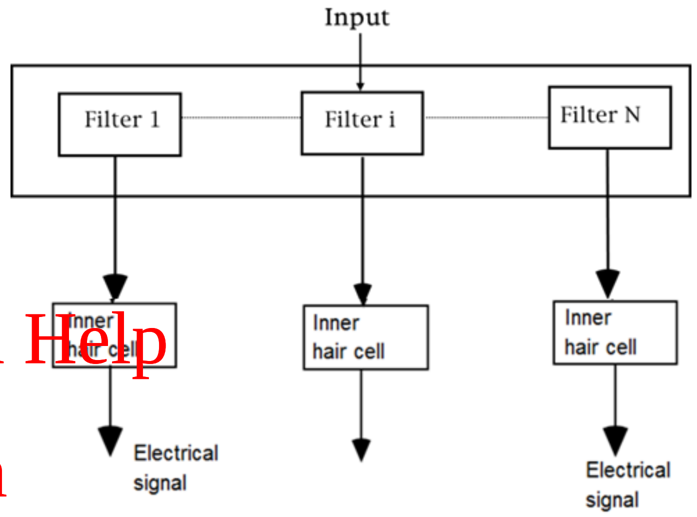


Cochlear Modelling: Cascade and Parallel Models

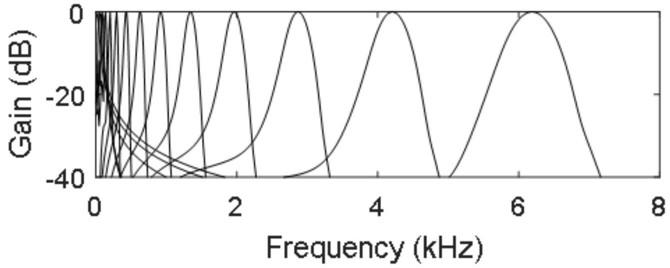
Transmission Line Model



Parallel Filter Bank Model



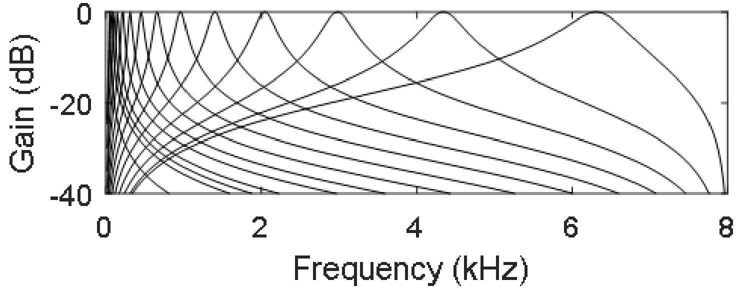
- ✓ The basic model of the cochlea is a transmission line model (cascade model) in which the basilar membrane is modelled as a cascade of 128 low pass filters, notch filters and resonators as shown above.
- ✓ Each digital filter section in the model above represents a section of the basilar membrane (tuned to a specific frequency) with 128 sections representing the entire basilar membrane



Magnitude response of the cascaded filter bank model

- ✓ The peripheral auditory system is often modelled as a bank of 128 bandpass filters (***auditory filters***) with overlapping passbands.
- ✓ Typically modelled using a finite number of bandpass filters, equally spaced along the Basilar Membrane.

Magnitude response of the parallel filter bank model



ELEC3104: Mini-Project – Cochlear Signal Processing

TLT – Level 1: Learning Activities (MATLAB Coding)

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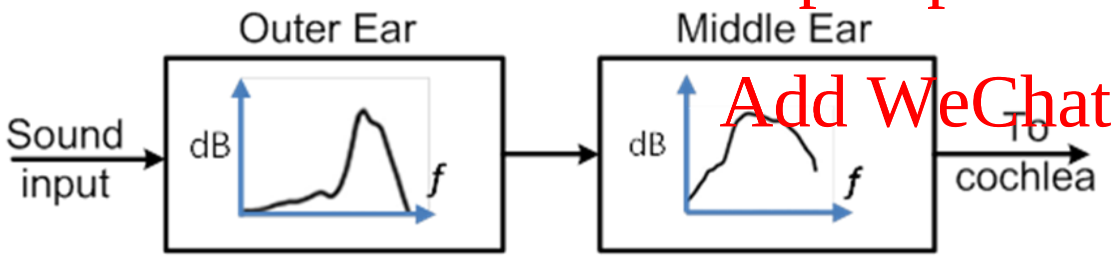
Learning Activity 1: Modelling the Outer Ear and the Middle Ear

- ✓ The middle ear may be modelled as a cascade of two complex pairs of zeros (to remove very high and very low frequencies) and one complex pair of poles (to provide low-Q gain at the middle frequencies). The approximate frequency response of the middle ear can be seen in the figure below.

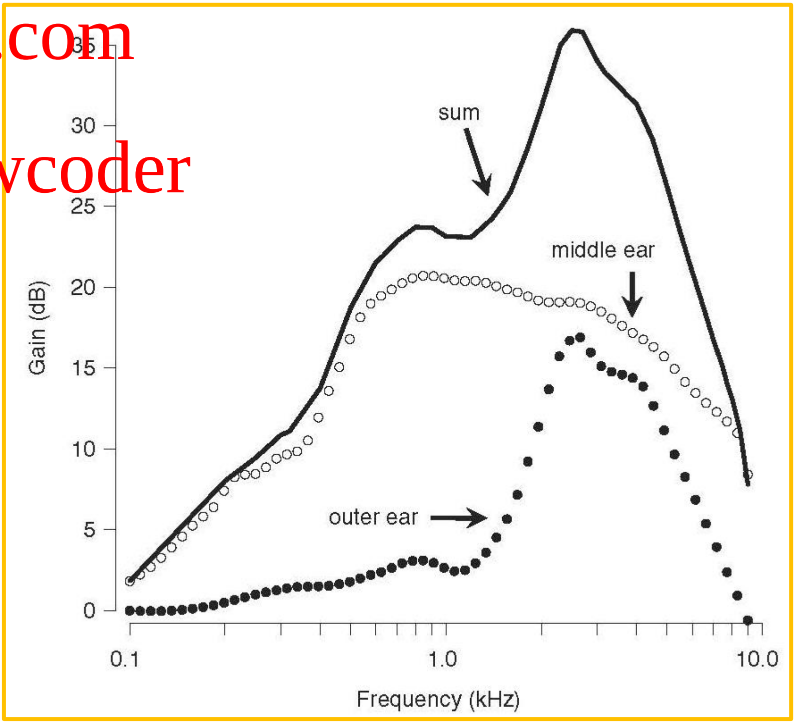
Assuming a sampling frequency of 16kHz:

(a) Obtain the transfer function of the middle ear filter, by suitably placing poles and zeros on the z-plane. Verify your results in MATLAB.

(b) Using placement of poles and zeros, estimate a model for the outer ear and cascade it with your previous model of the middle ear and show using MATLAB that the overall response matches the one shown in this figure.



The combined frequency response of the outer and middle ear is a band-pass response (sum – see the adjacent magnitude response diagram), with its peak dominated near 3 kHz



Filter Design: Pole zero placement

- ✓ Calculate the digital filter coefficients of the resonant pole and resonant zeros using pole zero placement (e.g. : see diagram below)
- ✓ Resonant pole frequency = θ_p; radius = r_p; $\theta_p = \frac{2\pi f_p}{f_s}$; $f_s = 16kHz$ (or higher)
- ✓ Resonant zero frequency = θ_z; radius = r_z ($r_z > r_p$ and closer to unit circle); $\theta_z = \frac{2\pi f_z}{f_s}$
- ✓ $H_p(z) = \frac{z^2}{(z-r_p e^{j\theta_p})(z-r_p e^{-j\theta_p})} = \frac{z^2}{z^2 - r_p(e^{j\theta_p}+e^{-j\theta_p})z + r_p^2} = \frac{z^2}{z^2 - r_p(2\cos\theta_p)z + r_p^2} = \frac{1}{1-2r_p\cos\theta_p z^{-1} + r_p^2 z^{-2}}$
- ✓ $H_p(z) = \frac{1}{1-b_1 z^{-1}+b_2 z^{-2}}$ (from one section of the digital filter) – Equating to above, we obtain

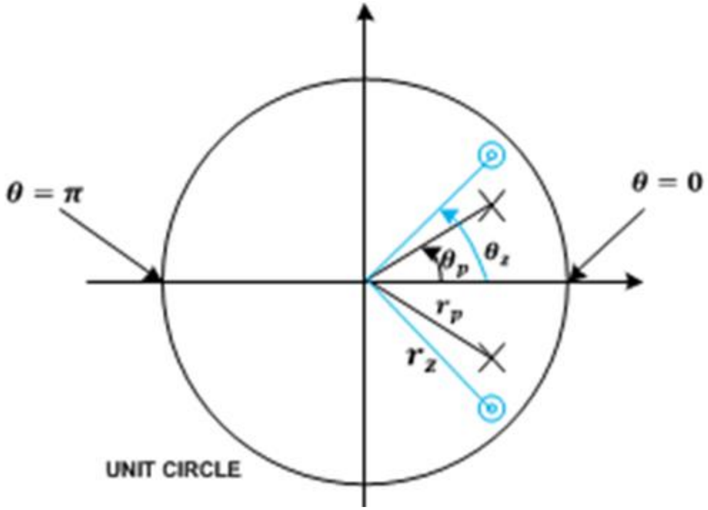
  $b_1 = 2r_p\cos\theta_p$ and $b_2 = r_p^2$.
- ✓ Similarly, $a_1 = 2r_z\cos\theta_z$ and $a_2 = r_z^2$ for $H_z(z) = 1 - a_1 z^{-1} + a_2 z^{-2}$
- ✓ Both transfer functions can be normalised such that DC gain = 1 as follows:

$$H_p(z) = \frac{1-b_1+b_2}{1-b_1 z^{-1}+b_2 z^{-2}} \quad \text{and} \quad H_z(z) = \frac{1-a_1 z^{-1}+a_2 z^{-2}}{1-a_1+a_2}$$

- ✓ r_p and r_z can be calculated approximately as follows:

$$r_p \approx 1 - \left(\frac{BW_p}{f_s}\right)\pi \; ; r_z \approx 1 - \left(\frac{BW_z}{f_s}\right)\pi$$

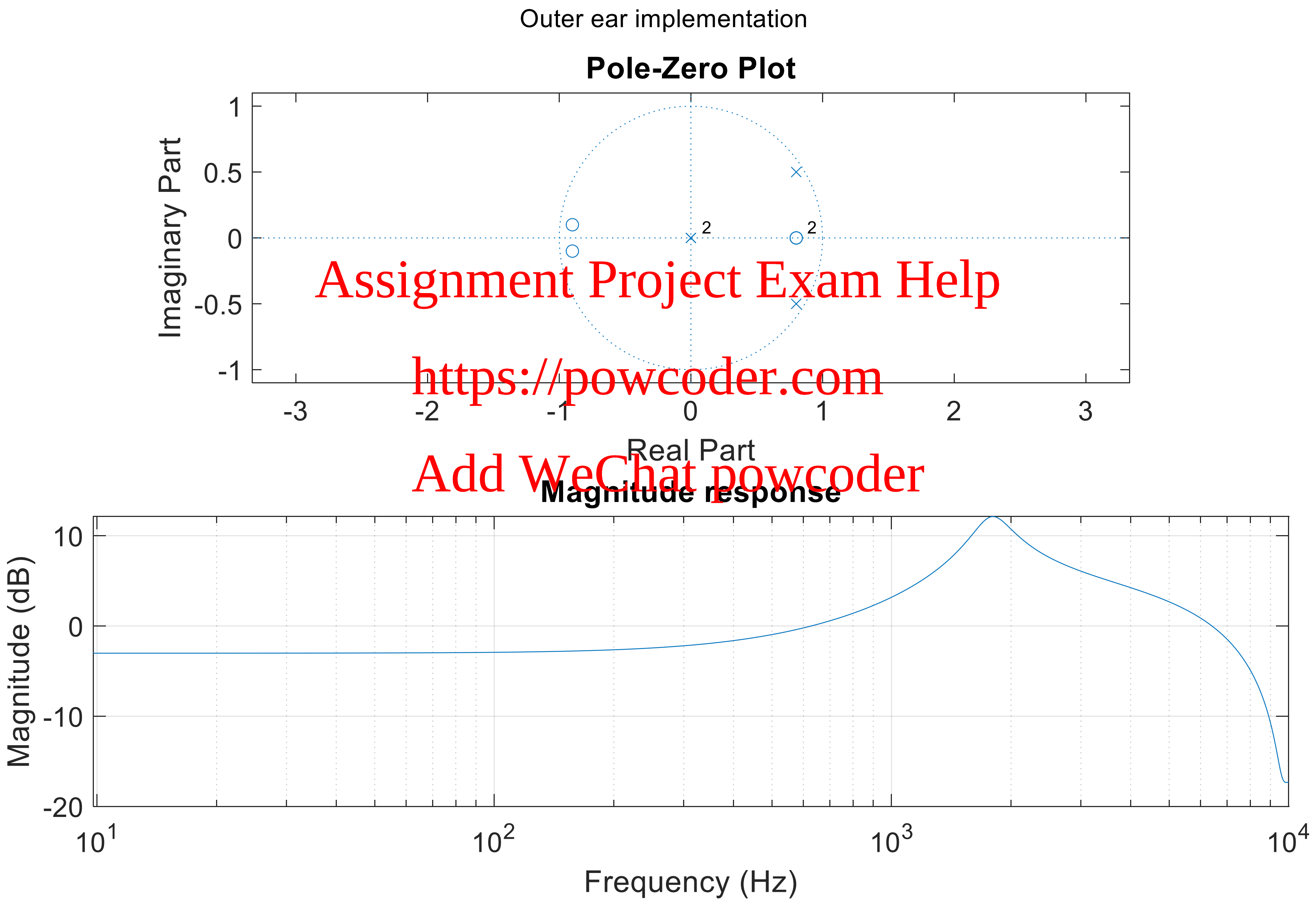
Q-factors: $Q_p = \frac{f_p}{BW_p}$; $Q_z = \frac{f_z}{BW_z}$



Pole – zero plots and magnitude responses of the outer ear

```
% Outer ear implementation - Learning activity 1
% Using the magnitude response (dotted line) given in slide 15
% You can observe that the magnitudes at 0 Hz and 10 kHz are
closer to zero.
% Hence, we need to place zeros at real axis and complex zeros
closer to 10 kHz.
% Therefore we choose the sampling frequency at 20 kHz.
% You may notice that there is a peak around 2 kHz.
% Therefore, we need to place a complex conjigate pair
(causing a peak) -
% - at 2 kHz (approximately 0.8 + 0.5i and 0.8 - 0.5i).
fs = 20*10^3; % sampling frequency
zero_1 = 0.8;
zero_2 = 0.8;
zero_3 = -0.9 + 0.1i;
zero_4 = -0.9 - 0.1i;

pole_1 = 0;
pole_2 = 0;
pole_3 = 0.8 + 0.5i;
pole_4 = 0.8 - 0.5i;
% convert the poles and zeros to numerator and
denominator polynomials
zeros_outer_ear = [zero_1 zero_2 zero_3 zero_4];
poles_outer_ear = [pole_1 pole_2 pole_3 pole_4];
b = poly([zero_1 zero_2 zero_3 zero_4]);
a = poly([pole_1 pole_2 pole_3 pole_4]);
% pole-zero plot and magnitude response
figure
sgtitle('Outer ear implementation');
subplot 211
zplane(b,a);
title('Pole-Zero Plot');
subplot 212
n = 1024; % FFT points
[H,w] = freqz(b,a,n);
mag_db = 10*log10(abs(H));
% plot the x axis in log scale
semilogx(fs/2*(w/w(end)),mag_db);
grid on;
title('Magnitude response');
ylabel('Magnitude (dB)');
xlabel('Frequency (Hz)');
```

Pole – zero plots and magnitude responses of the outer ear

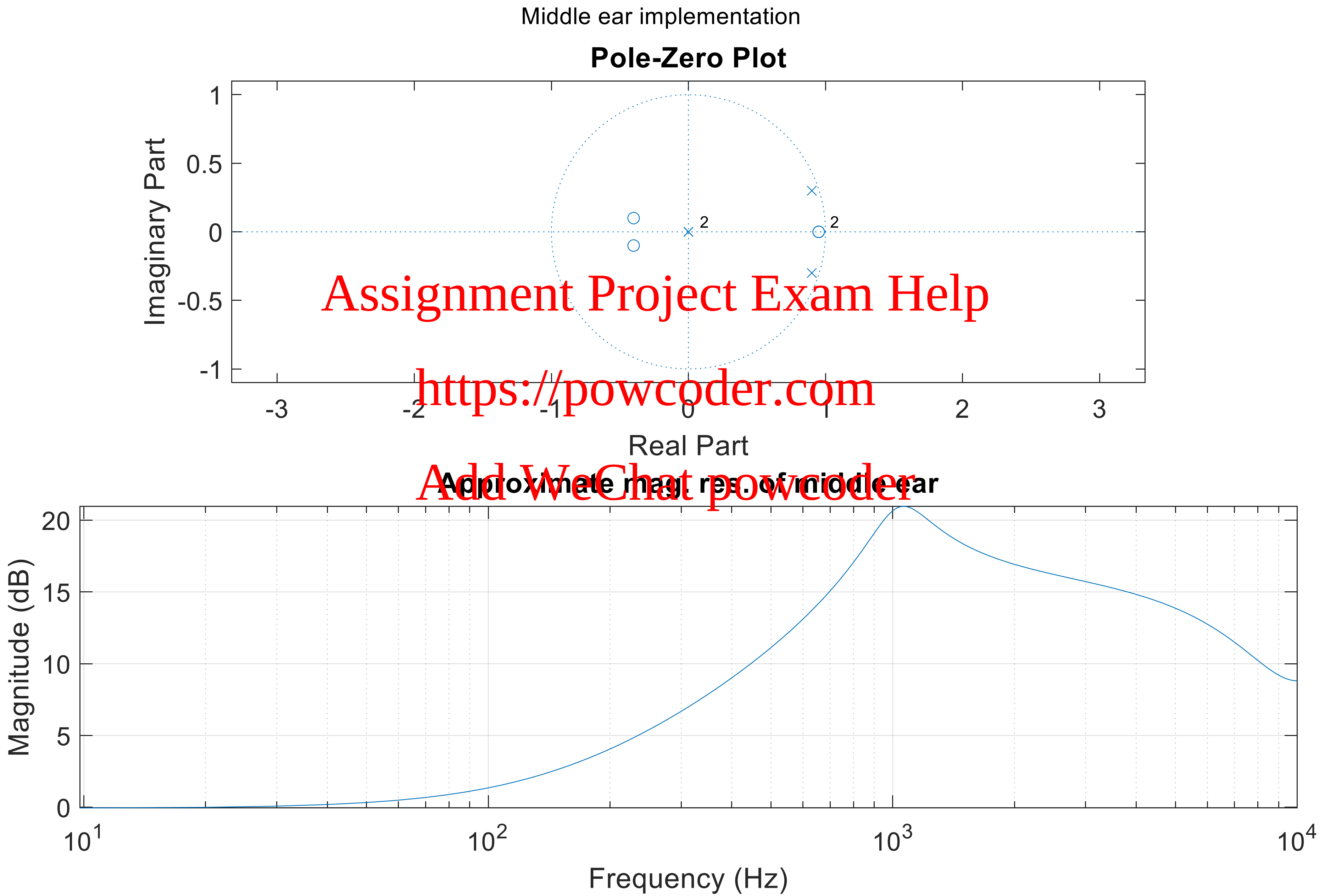


Pole – zero plots and magnitude responses of the middle ear

```
% Middle ear implementation - Learning activity 1

zero_1 = 0.95;
zero_2 = 0.95;
zero_3 = -0.4+0.1i;
zero_4 = -0.4-0.1i;

pole_1 = 0;
pole_2 = 0;
pole_3 = 0.9+0.3i;
pole_4 = 0.9-0.3i;

% convert the poles and zeros to numerator and -
% -denominator polynomials

zeros_middle_ear = [zero_1 zero_2 zero_3 zero_4];
poles_middle_ear = [pole_1 pole_2 pole_3 pole_4];
b = poly(zeros_middle_ear);
a = poly(poles_middle_ear);
```

```
figure
sgtitle("Middle ear implementation");
subplot 211
zplane(b,a) % plot zplane of middle ear model
title('Pole-Zero Plot');
% compute freq. resp. of middle ear model
n = 1024; % FFT points
k0 = 20; % gain factor


[H, w] = freqz(k0*b,a,n);
subplot 212
% plot the x axis in log scale
semilogx(fs/2*w/w(end),10*log10(abs(H)));
grid on
xlabel('Frequency (Hz)')
ylabel('Magnitude (dB)')
title('Approximate mag. res. of middle ear');
```

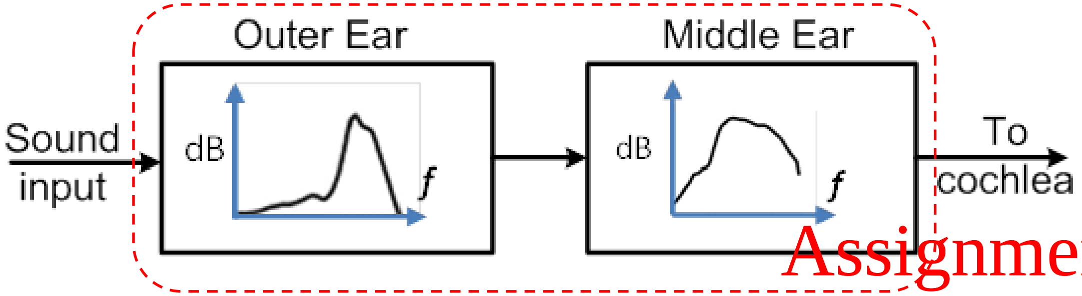
Pole – zero plots and magnitude responses of the middle ear

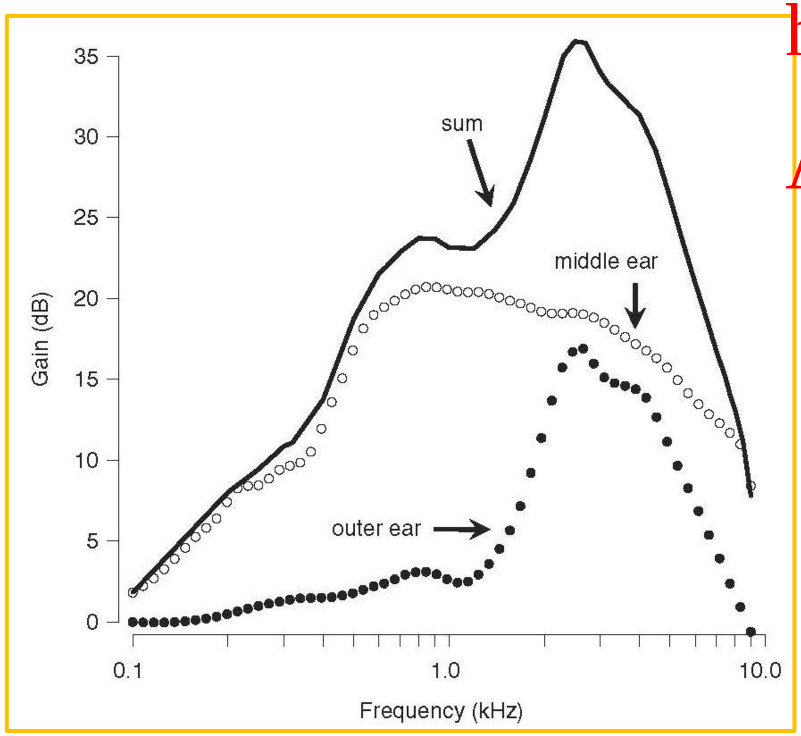


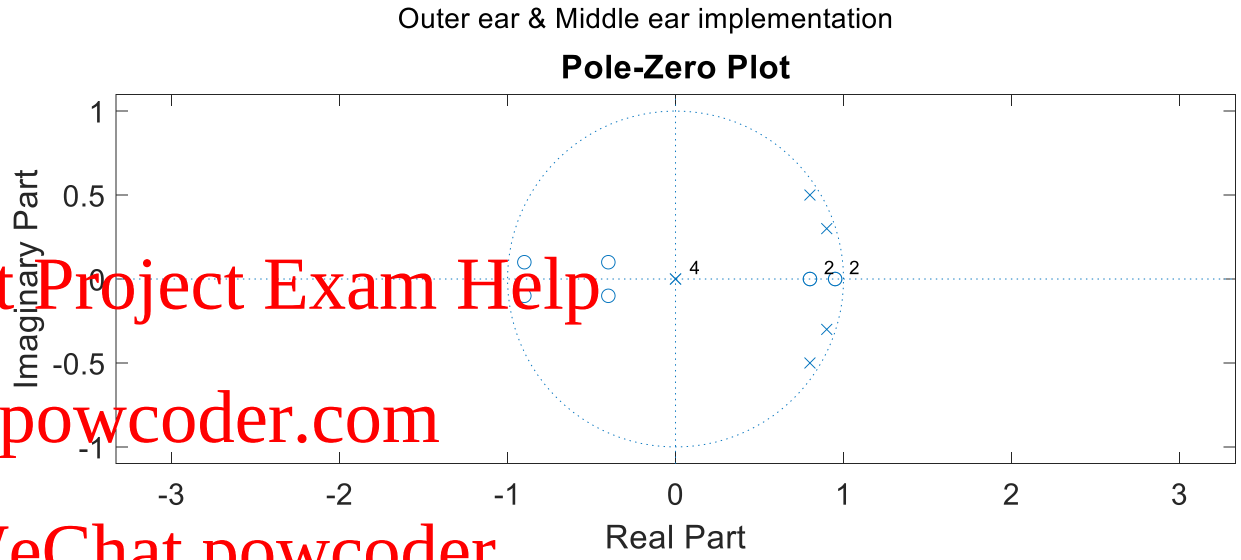
Combined magnitude responses of the outer & middle ear

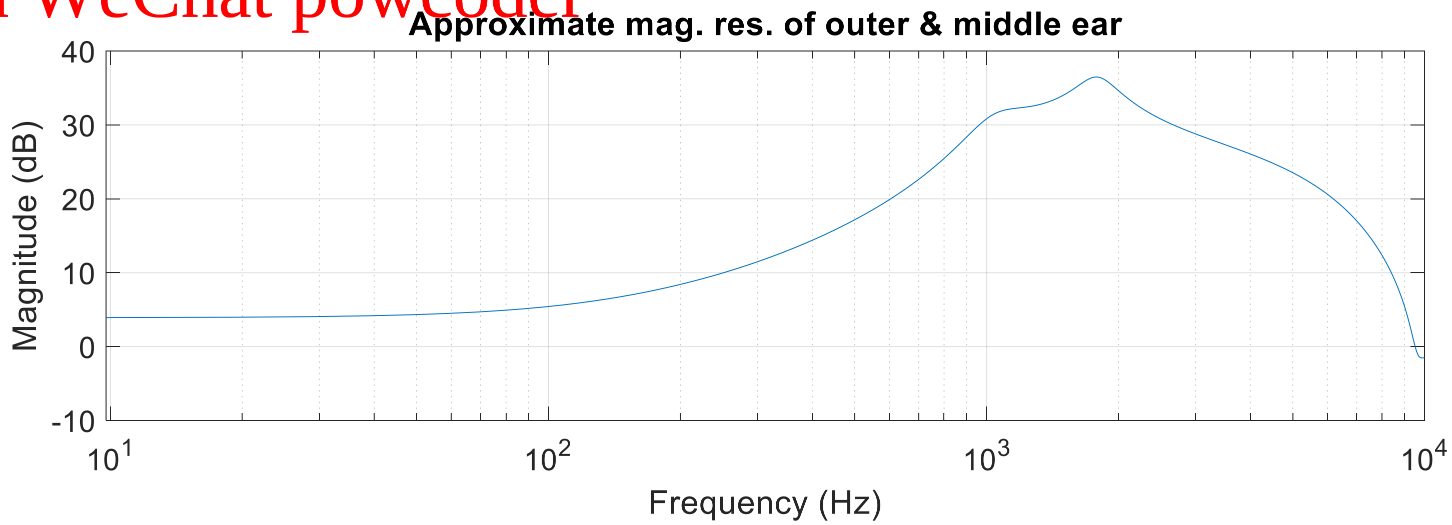
```
% Combine Outer ear and middle ear implementation -
Learning activity 1

zeros_combined = [zeros_outer_ear zeros_middle_ear];
poles_combined = [poles_outer_ear poles_middle_ear];

b = poly(zeros_combined);
a = poly(poles_combined);

figure
sgtitle("Outer ear & Middle ear implementation");
subplot 211
zplane(b,a) ;% plot zplane of middle ear model
title('Pole-Zero Plot');
% compute freq. resp. of middle ear model
n = 1024; % FFT points
k0 = 100; % gain factor
[H, w] = freqz(k0*b,a,n);
subplot 212
% plot the x axis in log scale
semilogx(fs/2*w/w(end),10*log10(abs(H)));
grid on
xlabel('Frequency (Hz)')
ylabel('Magnitude (dB)')
title('Approximate mag. res. of outer & middle ear');
```

Combined magnitude responses of the outer & middle ear









Learning Activity 2 : Impulse and Magnitude Responses

- ✓ The **impulse response** of an auditory filter can be modelled by:

$$g[n] = k\ (nT)^{N-1} e^{-2\pi b(24.7+0.108 f_p)nT} cos(2\pi f_p nT)$$

where, f_p is the centre frequency, T is the sampling period (f_s=$^1/_T$) , n is the discrete time sample index, N is the order of the filter ($N = 4$) and a is a constant chosen such that the filter gain at the centre frequency is 0dB ; $b = 1.14$; f_s= 16,000 Hz.; [Initially, you may choose a=1 and then change the value such that the gain of the filter is normalised to 0dB at the centre frequency, f_p .

- ✓ You are required to **calculate** the impulse response, $g(n)$, for **four** auditory filters of your choice from the low, mid and high frequency regions of the basilar membrane using the equation {f_p(x)} given below in MATLAB.

$$f_p(x) = (8000.0)\ 10^{-0.667\ x}\ Hz \qquad 0.0869\ cm\,(80\ Hz) \leq x \leq 2.9985\ cm\,(7\ kHz)$$

You will notice that the impulse responses have infinite duration, and thus each impulse response will need to be truncated to, say, 150 to 200 coefficients (i.e., $0 \leq n < 200$).

- ✓ Plot the impulse responses of all **four** filters.

- ✓ Plot the magnitude responses of all four filers.

- ✓ Plot the centre frequency, bandwidth and Q factor for all filters.

Discuss your plots with your lab demonstrator.

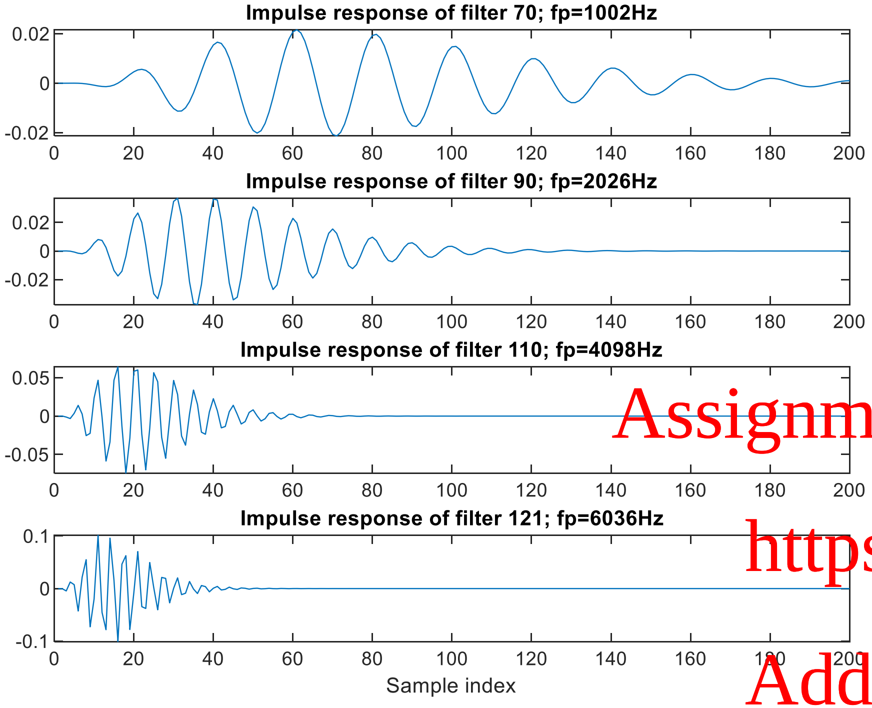
Learning Activity 2 – MATLAB code

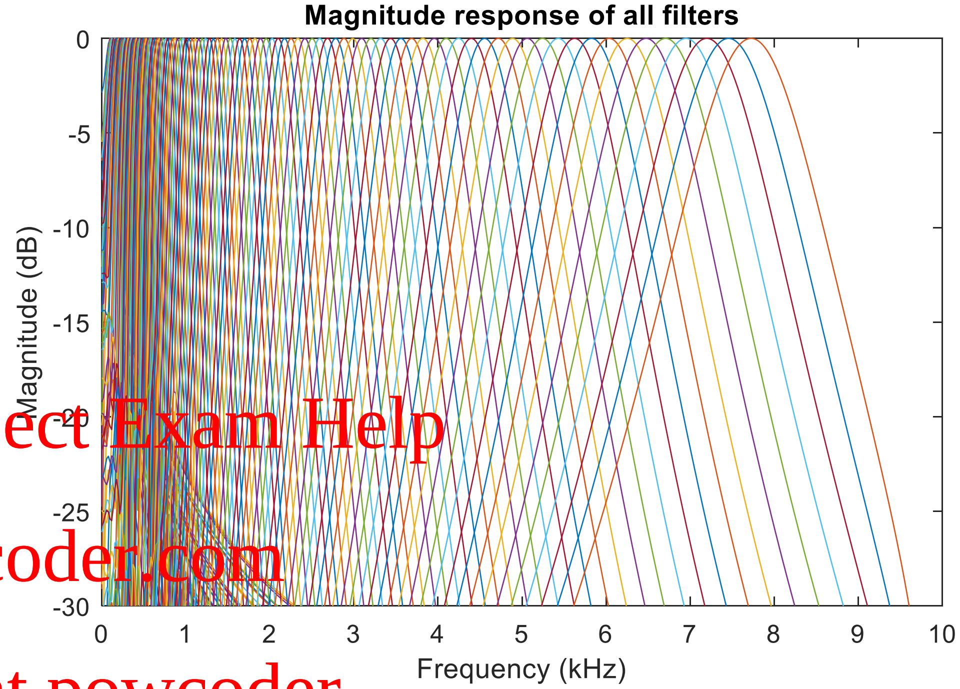
```
% Impulse and magnitude response calculation of an auditory filter
clc; clear all; close all
fs=20000;
num_filter = 128; % nubmer of filters
NFFT=1024; % number of FFT points
% 2.9985 and 0.0869 are xmax and xmin
delta_x = (2.9985-0.0869)/(num_filter-1);
k = 128:-1:1; % filter index
fp = 8000*10.^(-0.667*k*delta_x); % centre frequencies
% Auditory filter parameters
b=1.14;
T=1/fs; % sampling period
N=4; % order of filter
n=0:199; % sample index

% filter's impulse response
for i=1:num_filter
    g(:,i)=((n*T).^(N-1)).*exp(2*pi*b*(24.7+0.108*fp(i))*n*T).*...
cos(2*pi*fp(i)*n*T);
end
G=fft(g, NFFT); % filter's frequency response in [0 fs]
G=abs(G(1:NFFT/2,:)); % filter's magnitude response in [0 fs/2]
for i=1:num_filter
% normalize all the impulse response max to 1
g(:,i)=g(:,i)/max(abs(G(:,i)));
end
G=fft(g, NFFT); % normalised filter's frequency response in [0 fs]
% normalised filter's magnitude response [0 to fs/2] in dB
G = 20*log10(abs(G(1:NFFT/2,:)));
% est. filter's bandwidth
% i.e. width of freq. region where filter's gain > -3dB
freqHz = fs*(0:NFFT-1)/NFFT; % frequency axis [0 fs]
freqHz = freqHz(1:NFFT/2); % frequency axis [0 fs/2]
BW = [];
for i=1:num_filter
% find frequency index in passband region of filter
pass_band_freqID = find(G(:,i)>=-3);
% Bandwidth of filter
BW = [BW freqHz(pass_band_freqID(end))-freqHz(pass_band_freqID(1))];
end
```

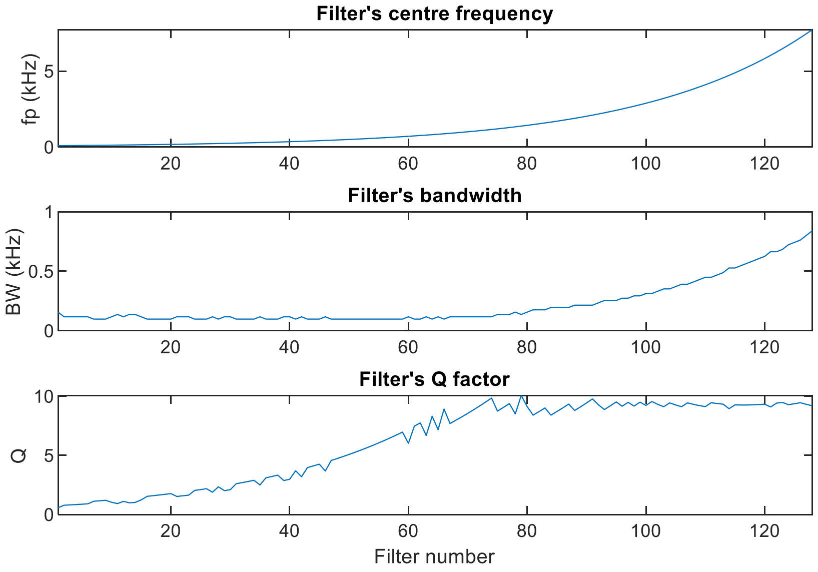
Learning Activity 2 - MATLAB code

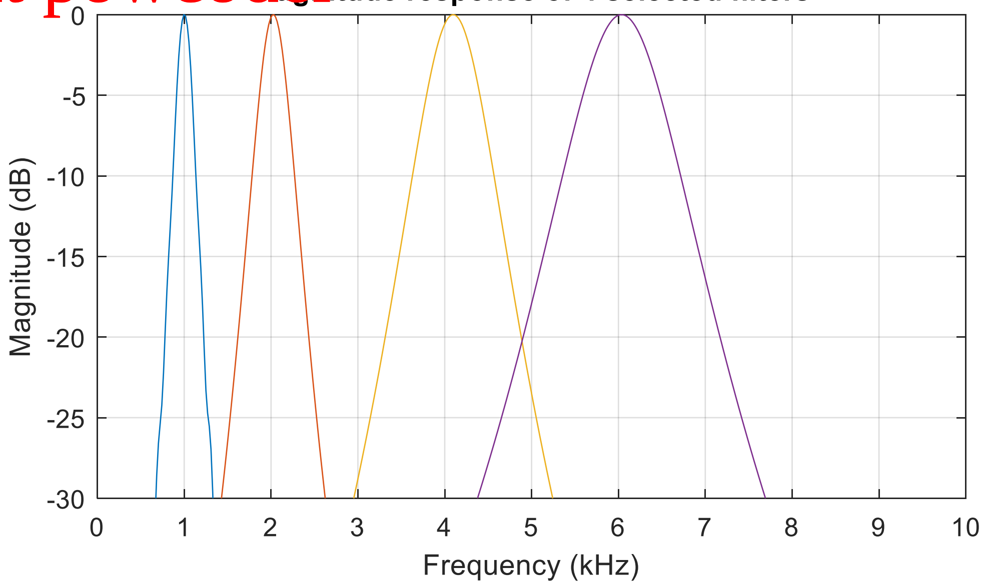
```
Q = fp./BW; % Q factor of filter (Selectivity)
figure
subplot 311
plot(fp/10^3) % plot centre frequencies
xlim([1 num_filter])
ylabel('fp (kHz)')
title('Filter''s centre frequency')
subplot 312
plot(BW/10^3) % plot bandwidth
ylabel('BW (kHz)')
xlim([1 num_filter])
title('Filter''s bandwidth')
subplot 313
plot(Q) % plot Q factor
xlim([1 num_filter])
ylabel('Q')
xlabel('Filter number')
title('Filter''s Q factor’)

figure
vlimit=[0.02 0.05 0.1 0.5];
checked_filter_index = [70, 90, 110, 121];
for i = 1:length(checked_filter_index)
subplot(4,1,i)
plot(g(:,checked_filter_index(i)))
axis([0 200 min(g(:,checked_filter_index(i)))...
max(g(:,checked_filter_index(i)))])
title(['Impulse response of filter ' num2str(i) ...
'; fp=' num2str(round(fp(i))) 'Hz'])
end
xlabel('Sample index')
figure
plot(freqHz/10^3,G(:,checked_filter_index));
hold on
xlabel('Frequency (kHz)');
ylabel('Magnitude (dB)');
axis([0, fs/2/10^3 -50 0]);
ylim([-30 0]);
title('Magnitude response of 4 selected filters');
grid on;
figure
plot(freqHz/10^3,G(:,:));
xlabel('Frequency (kHz)');
ylabel('Magnitude (dB)');
axis([0, fs/2/10^3 -50 0]);
ylim([-30 0]);
title('Magnitude response of all filters');
```

Learning Activity 2 – Filter responses









Learning Activity 2

Reflections

(a) What major differences do you see between the impulse responses you have plotted and why?

(b) Using these impulses responses, find the magnitude responses of all four filters and plot them (frequency vs magnitude in dB) on the same figure so you can compare them.

(c) The gains at the centre frequencies of the filters may not be equal, choose the scaling factor $'k'$ for each filter (see equation of g[n]) such that the gain of each filter is normalised to 0dB at the centre frequency.

(d) Approximately estimate the 3dB bandwidths of all four filters from your plots. Do they vary with the centre frequency? If so, how do you think they are related?

(e) Explain your understanding of constant-Q filters and constant-Bandwidth filters.

Assignment Project Exam Help

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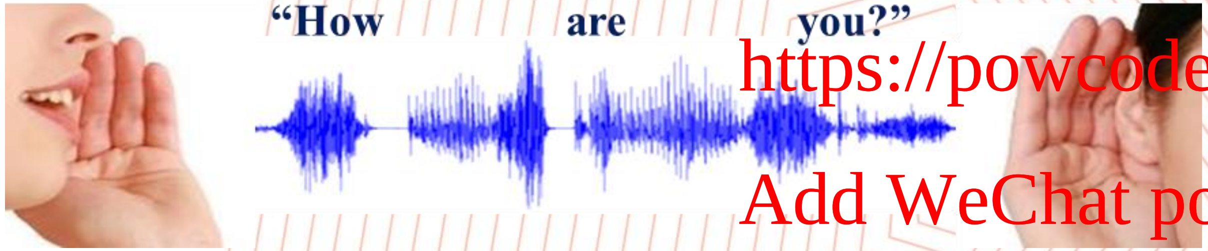
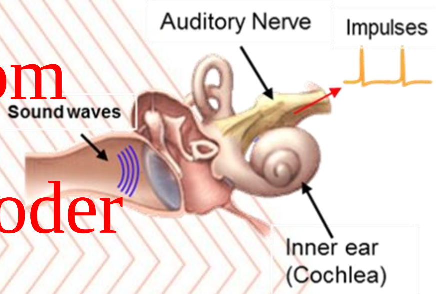
ELEC3104: Mini-Project – Cochlear Signal Processing





Prof. Eliathamby Ambikairajah, School of EE&T

Term 3, 2022

ELEC3104: Mini-Project – Cochlear Signal Processing

TLT – Level 2 (Pass Level): Implementation of a cascaded filter bank model of the cochlea for analysis purposes.

Prof. Eliathamby Ambikairajah, School of EE&T

Term 3, 2022

Learning Activity 2 - MATLAB code