M.K.S.S.S's Cummins College of Engineering for Women
Department of Electronics & Telecommunication Engineering

Course: Digital Signal Processing (20EC501)
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In-Semester- T1
PROJECT BASED LEARNING

Digital Hearing Aid System for Hearing Impaired

Presented by: GROUP No. 2 (A4)

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PHASE- I EVALUATION

COURSE: DSP (PROJECT BASED LEARNING)

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Problem statement

According to biological concepts, Hearing process is very much important for human being for every second. After hearing sound our brain process that information and then we start doing activities with the help of our body. When we hear very low volume sound or information at that time we aren’t able to understand what should we do next. Low volume sound can be originally low sound but sometimes it is loud and we can’t hear properly so this problem is known as hearing loss.

Hearing loss is a measure of shift in the auditory system compared to that of normal ear for detection of a pure tone. The huge scale of the human population suffers from hearing loss.

Abstract

- With the availability of modern day technologies and the recent developments in signal processing area, sophisticated artificial hearing aid systems can be designed that relax the job of damaged auditory systems to a great extent and make much of the sound available to the hearing impaired.
- In this project, the simulation of the simple digital hearing aid will be developed in 2nd phase using MATLAB programming language.
- The implementation of this configurable digital hearing aid (DHA) system includes noise reduction filter, frequency-dependent amplification and amplitude compression.
- We will test our filters on a mock patient and successfully reduced white Gaussian noise, increased the gain for frequencies which were difficult to hear, and shaped the amplitude to prevent any of the frequencies from becoming too loud.

Literature Survey

S. No.	Title of Paper	Name of Conference/Journal	Year of Publication	Summary and Results
1.	Digital filtering in hearing aid system for the hearing impaired	IEEE, ICEEOT	2016	Block diagram representation of the hearing aid system.
2.	Enhancement of speech signals for Hearing Aid Devices using digital Signal processing	IEEE, 4^{th} ISMSIT	2013	The digital hearing aids can be programmed to match patents hearing individually according a specific Frequency. The aids are programmed using the human audiogram. Digital hearing aid work with very low power battery, approximately in mW.

Literature Survey

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3.	Comparative analysis of wavelet transform filtering systems for noise reduction	PLoS ONE journal	2022	The selection of WT system settings significantly affects the efficiency of de-noising procedure.
4.	Denoising Speech Signals for Digital Hearing aids: A wavelet Based Approach	ResearchGate	2011	This paper describes the research developing a wavelet based, single microphone noise reduction algorithm for use in digital hearing aids.

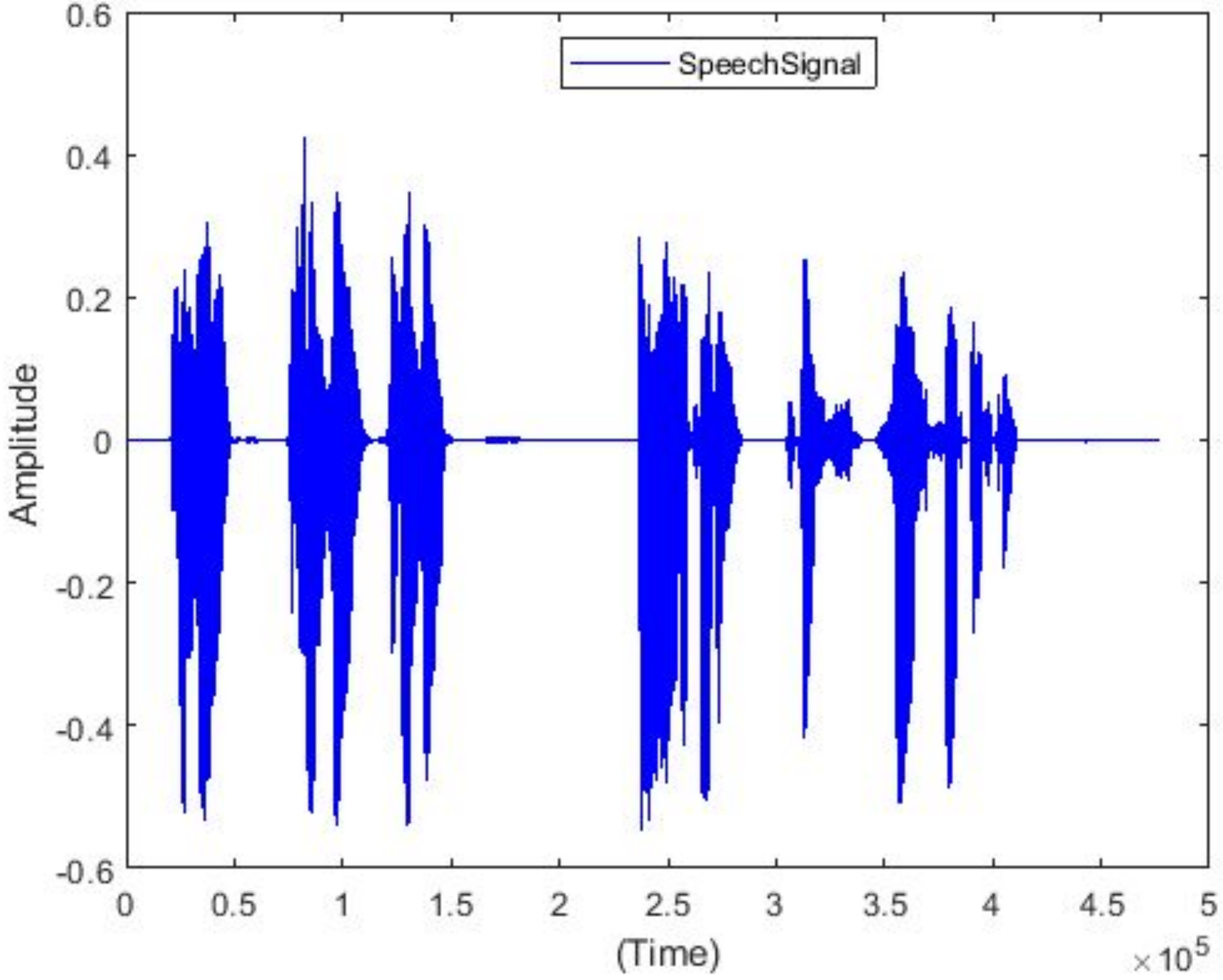
Study of Speech Signal and its Characteristics

The speech is an one -dimensional function of time. Speech signal is a non-stationary signal.

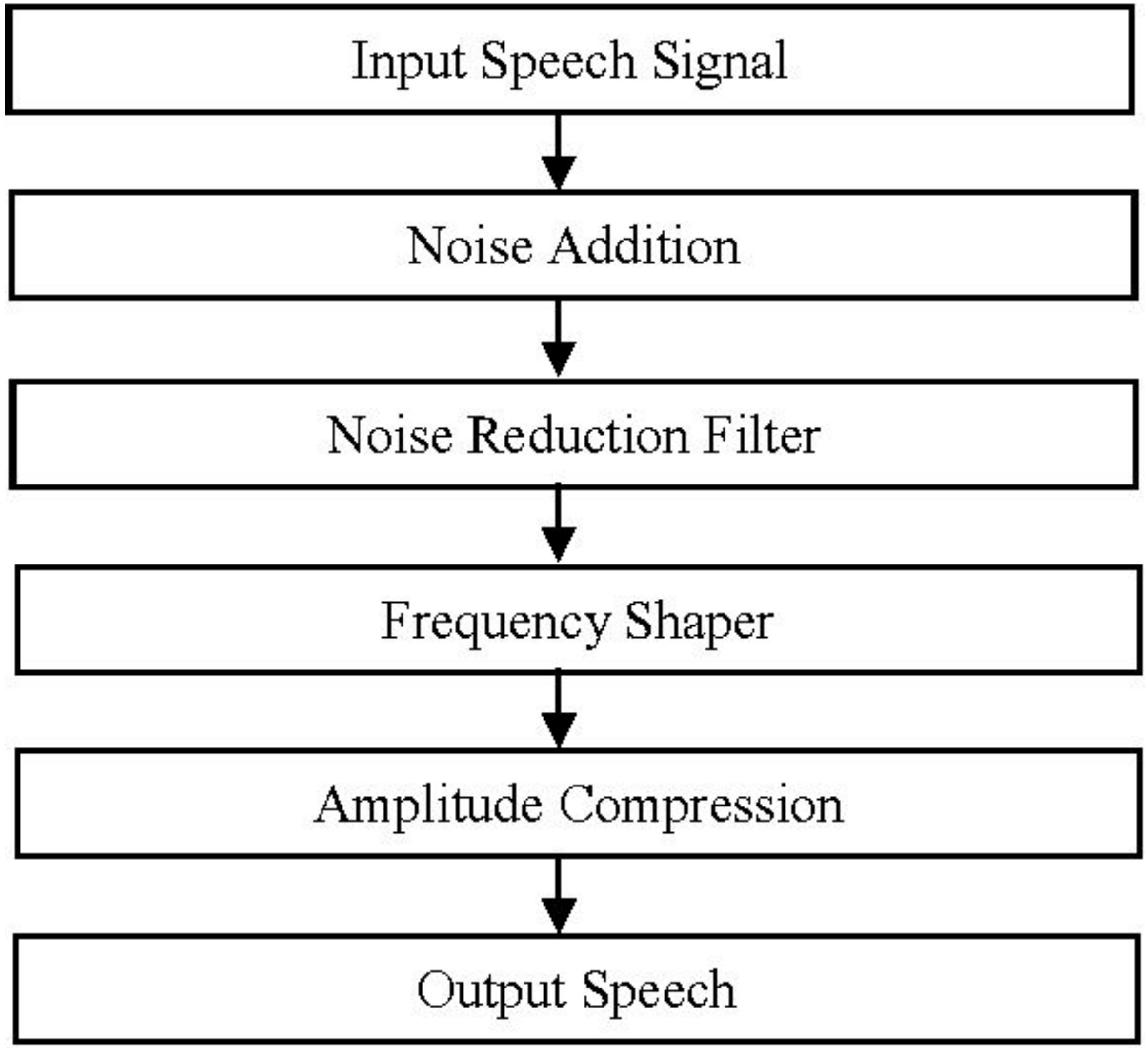
The generally accepted standard range of **Audible frequencies is 20 to 20,000 Hz**, although the range of frequencies individuals hear is greatly influenced by environmental factors. Frequencies below 20 Hz are generally felt rather than heard, assuming the amplitude of the vibration is great enough. Frequencies above 20,000 Hz can sometimes be sensed by young people. High frequencies are the first to be affected by hearing loss due to age and/or prolonged exposure to very loud noises.

Classification of Hearing loss	**Range**
Normal hearing	-10 dB - 26 dB
Mild hearing loss	30 dB - 40 dB
Moderate hearing loss	40 dB - 70 dB
Severe hearing loss	70 dB - 90 dB

Waveform of Speech Signal



System Block Diagram



Methodology

- The input speech signal takes the form of a human voice. For producing an adjusted output speech signal which can be audible to the hearing impaired person.
- The input speech signal will pass through several functions i.e., noise addition, noise reduction filter, frequency shaper and amplitude compression before producing an adjusted output speech signal which is audible to the hearing impaired person.

Detail Design

1. **Noise Addition -**

   In this system, To simulate a real situation, the Additive White Gaussian Noise (AWGN) and random noise are added to the input speech signal. The noise (AWGN) has a continuous and uniform frequency spectrum over a specified frequency band and has equal power per Hertz of this band. It consists of all frequencies at equal intensity and has a normal (Gaussian) probability density function.

   Equation for AWGN -

   Y= s + 10^(-Eb_N0_dB/20)*n;

   where 's' is the transmitted sequence, Eb_N0_dB is the SNR and 'n' is the Additive White Gaussian Noise.

2. Noise Reduction Filter -

A major anxiety for the people with hearing loss is the capability of hearing aids to differentiate intended speech signals in a noisy environment. Hence, to eliminate the noise, a reduction filter function is used in this design. To suppress the noise in the signal, the wavelet filter function is used.

Why Wavelet Filter?

- In order to reduce the noise of speech signal many techniques are available like digital filters (FIR or IIR), adaptive method and wavelet transform thresholding methods. However, digital filters and adaptive methods can be applied to signal whose statistical characteristics are stationary in many cases. Recently the wavelet transform has been proven to be useful tool for non-stationary signal analysis.

Alfaouri M, Daqrouq K. ECG Signal Denoising By Wavelet Transform Thresholding. Am J Appl Sci. 2008;5:276–281.

- Wavelet filter localize features in our data to different scales, we can preserve important signal while removing noise. The basic idea behind wavelet denoising, is that the wavelet transform leads to a sparse representation for many real-world signals. Means the wavelet transform concentrates signal in a few large-magnitude wavelet coefficients. Wavelet coefficients which are small in value are typically noise and you can "shrink" those coefficients or remove them without affecting the signal. After we threshold the coefficients, we reconstruct the data using the inverse wavelet transform

https://www.researchgate.net/profile/Khaled-Daqrouq/publication/256980860_denoising_ajassp/links/02e7e5242060d002bb000000/denoising-ajassp.pdf

Mathematical expression for Wavelet function is:

$$\psi_{a,b}(t) = \frac{1}{\sqrt{|a|}}\psi\left(\frac{t-b}{a}\right)$$

where a represents the scaling parameter for dilation and ‘b’ represents the moving parameter for translation for the entire signal location

The signal model of optimal filtering is shown below:

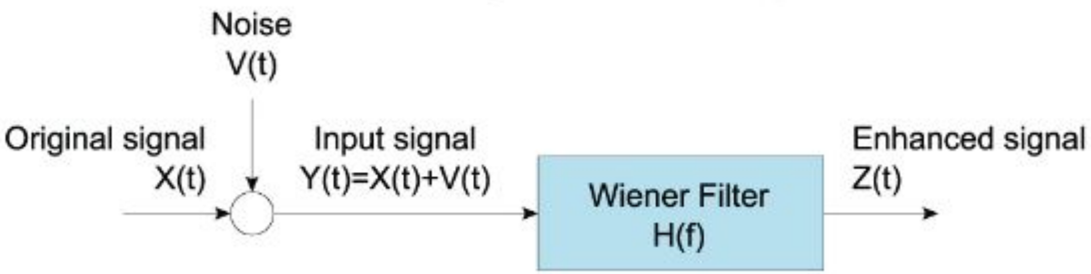


Figure 1. Optimal (Wiener) filtering signal flow

The signal of interest (X) is contaminated by an additive noise (V). In order to improve the signal integrity, the observed noisy signal (Y) is passed through a filter. The output of the filter (Z) is an estimate of the unknown signal (X). The estimation error, the difference of X and Z, has the lowest possible power if the frequency response of the filter (H) is given by:

$$H(f) = \frac{S_X(f)}{S_X(f) + S_V(f)}$$

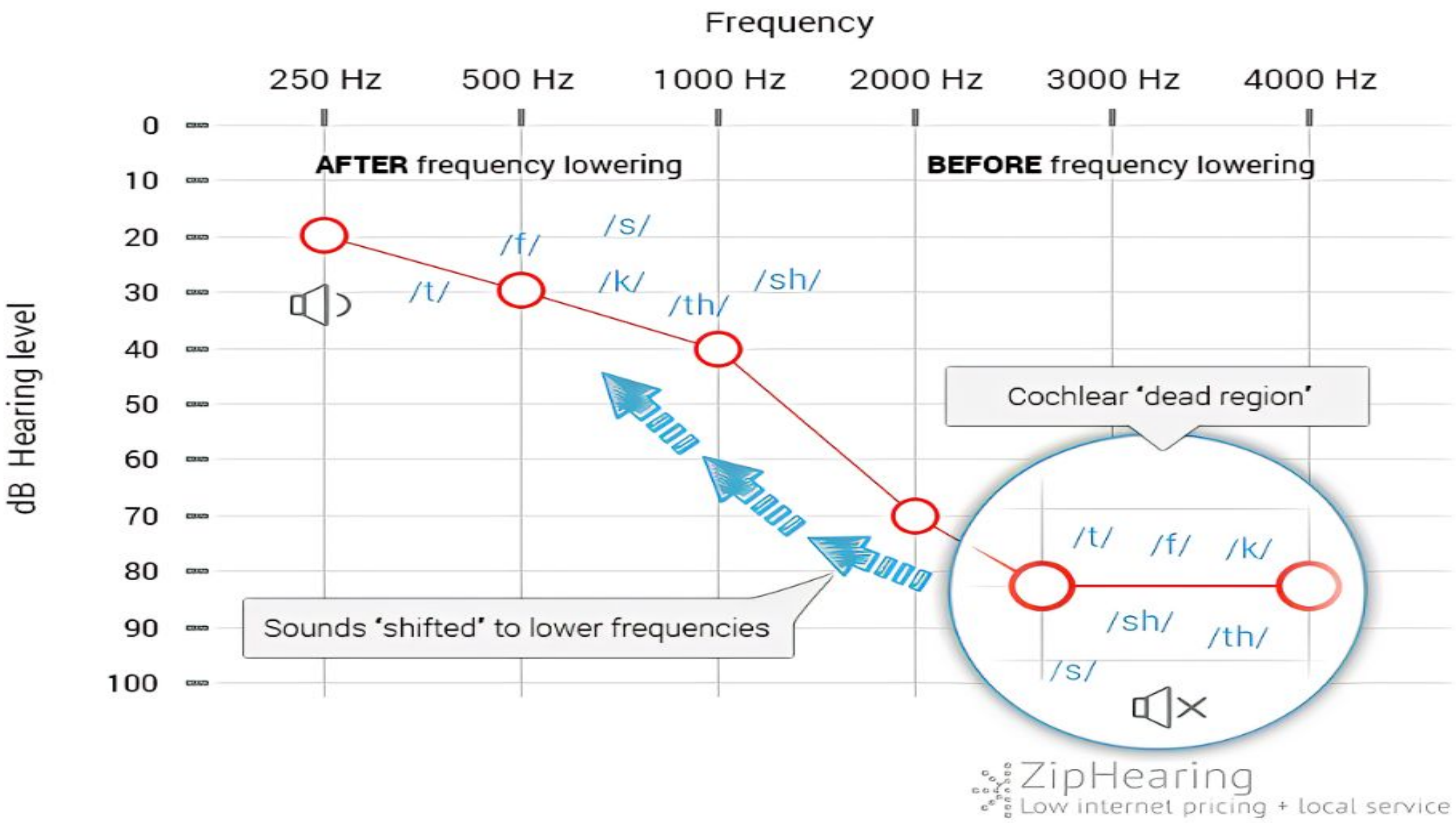
where S_X and S_V are the power spectral density functions (PSD) of X and V, respectively, which are, by assumption, statistically independent.

[Noise Reduction Filter (lectrosonics.com)](lectrosonics.com)

3. Frequency Shaper -

One major complaint of hearing aid users is that the hearing aid amplifies all signals rather than the significant signal that they desire to hear. Most hearing impaired people have difficulties hearing high frequency signals. Therefore, the frequency shaper is designed to correct for loss of hearing at certain frequencies. It applies high gain for higher frequencies and vice versa.

Hearing Aid Frequency Lowering

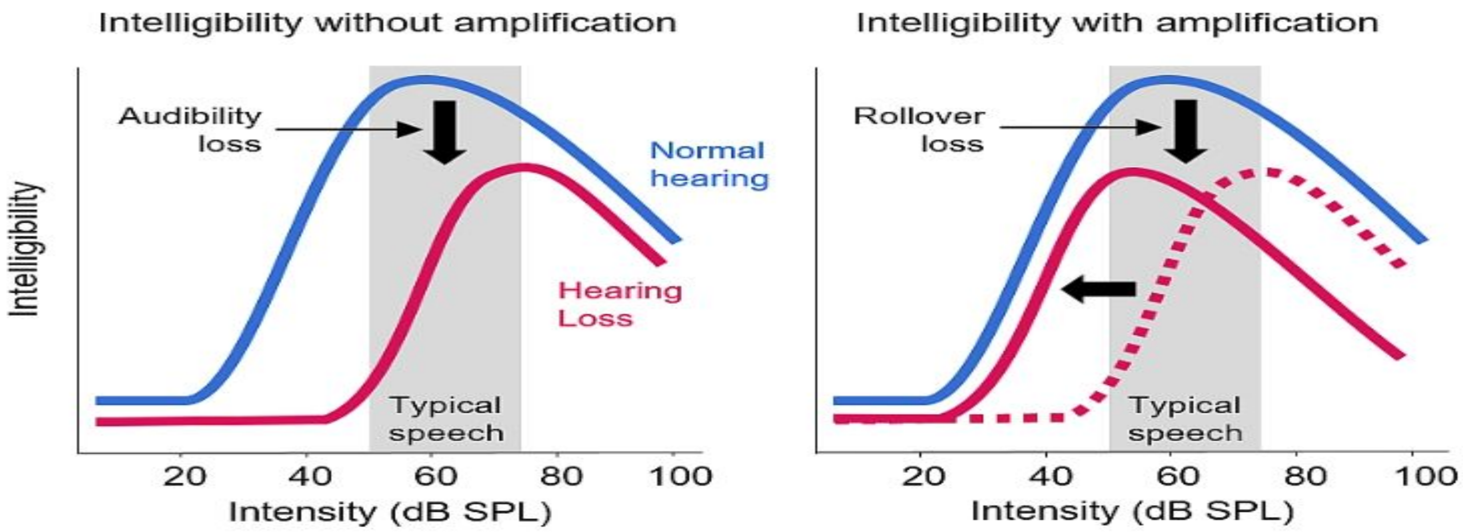


4. Amplitude Compression -

Amplitude compression function is the task of controlling the overall gain of a speech amplification system. Amplitude compression will ensure that the amplified signal will not exceed saturation power. Saturation power is where the sound signal begins to become uncomfortable.

When we compared the neural code for speech with normal hearing and hearing loss at the same high sound levels produced by the hearing aid, we found that the activity patterns elicited by different speech syllables were equally distinguishable with or without hearing loss. We finally concluded that rather than suffering from supra-threshold effects of hearing loss, listeners with hearing aids are simply experiencing the same rollover effect as those with normal hearing**. And when we looked back through the literature on previous studies of speech perception with normal hearing and hearing loss at high sound levels, we found plenty of evidence to support this conclusion.

A neuro-perceptual perspective on hearing



The hearing aid dilemma: With normal hearing (blue), the intelligibility of speech is high at typical conversational sound levels but decreases at higher levels, an effect known as rollover. With hearing loss (red, left), intelligibility is decreased at conversational levels because of decreased audibility. With amplification (red, right), the audibility of speech is restored but the effects of rollover keep intelligibility below normal.

[The hearing aid dilemma: amplification, compression, and distortion of the neural code | Nature Portfolio Bioengineering Community]

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[3] Navdeep Kaur and Hardeep Singh Ryait, "Study of Digital Hearing Aid Using Frequency Shaping Function", International Journal of Engineering Research and Technology (IJERT), vol. 2, no. 5, May 2013.

[4] John G. Proakis and G.D. Manolakis, "Digital Signal Processing" in , New Jersey:Prentice Hall, 2010.