A DIGITAL HEARING AID

Ankur Agarwal(Y09UC216)
Hansraj Gujar(Y09UC233)
Neeraj Gupta(Y09UC254)
Shrey Upadhyay(Y09UC282)
Sumit Gautam(Y09UC285)

INTRODUCTION

- In this project, we developed noise reduction, frequency, and amplitude filters for a configurable digital hearing aid (DHA).

Why digital?

- Analog hearing aids yield significant limitations due to
- Inadequate spectral shaping
- Narrow operating bandwidth
- Partial noise-reduction capability
- This leads to sub-optimal clarity and audibility restoration and sub-optimal speech perception in noisy environments.

Working

- The analog sound signal is converted into digital domain.
- The digital signal processor(filters) manipulates the signal without causing any distortion, so sounds come through more clearly and speech is easier to hear and understand.

Assumptions

- The highest frequency that most humans can hear is approximately 20 kHz. Therefore, before the signal enters the A/D converter, it will be lowpass-filtered to 20 kHz, which is also our sampling frequency. This will avoid aliasing during sampling.
- It does not operate in real-time, since we take in the entire speech signal and then manipulate it.

Design

1.

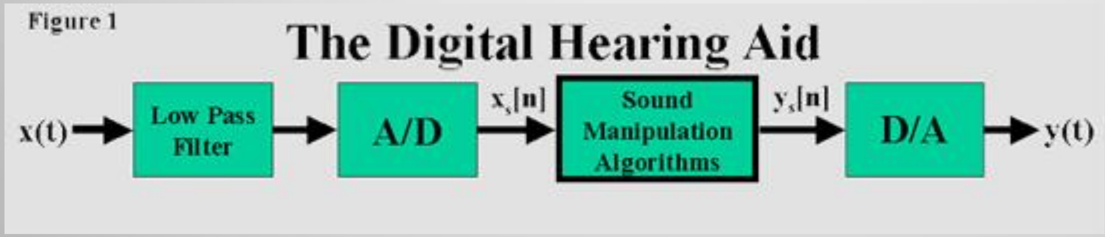

Figure 1
The Digital Hearing Aid
x(t)
Low Pass Filter
A/D
$x_s[n]$
Sound Manipulation Algorithms
$y_s[n]$
D/A
y(t)

2.

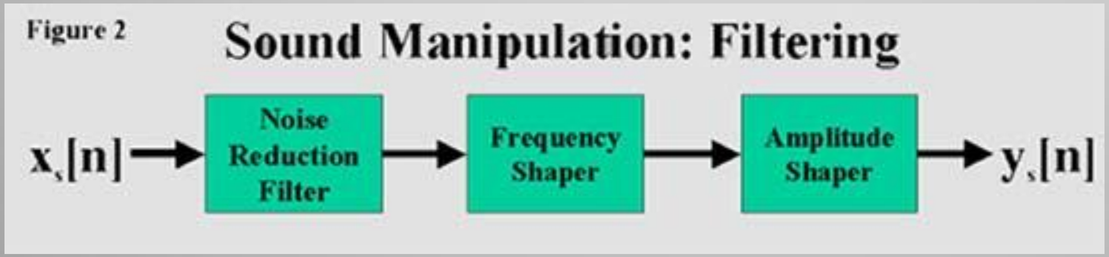

Figure 2
Sound Manipulation: Filtering
$x_s[n]$
Noise Reduction Filter
Frequency Shaper
Amplitude Shaper
$y_s[n]$

Noise reduction filter

- Assumptions
- External signals or Noise can be modelled using white gaussian noise.
- We used the command wdencmp, which performs noise reduction/compression using wavelets. It returns a de-noised version of the input signal using wavelet coefficients thresholding.

Frequency shaper

- The filter applies a gain greater than one to the frequencies that the user has difficulty hearing.
- As one of its parameters, the filter takes in a vector of frequencies, that define the user's hearing characteristics.
- For each range, the frequency shaper applies a certain gain based on the user's specific hearing characteristics . Thus, our frequency shaper is completely configurable to any user.

Amplitude shaper

- Once the signal has been passed through the Noise Reduction Filter and the Frequency Shaper, it will be passed through our Amplitude Shaper .
- We assume that the Frequency Shaper raises the frequencies that the user has difficulty hearing to sound pressure levels within his dynamic range of hearing.
- Therefore, all that our Amplitude Shaper has to do is check, bit by bit, that output power does not exceed a given saturation level, Psat.

- Since noise is concentrated in the low power levels as well, the filter also removes a significant amount of noise. Output power is equal to zero for levels below Psat.

Conclusion

- We successfully modeled a digital hearing aid in MATLAB by implementing a noise reduction filter, frequency shaper and an amplitude shaper for a given speech signal.