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## Project Report (part)

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# Hearing Impairment Simulation

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*This report is accompanied by a MATLAB package that can be  
requested by mail.*

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## Abstract

Hearing loss is a widespread phenomenon, especially among elders. It is characterized by a poor hearing of quiet voices and poor understanding of speech. Hearing instruments intend to help the hearing impaired to overcome some of their disabilities. Most of the hearing impaired population suffer from hearing loss in the high-frequency range. It leads to a confusion between fricatives such as /s/ and /sh/ or between other phonemes.

This impairment cannot be addressed by current hearing instruments, not by the simple ones that amplify all sounds equally, neither by the more complicated digital instruments that can amplify different ranges of frequencies, since they treat the sounds globally without paying attention to the specific characteristics of each phoneme.

This stresses the need for a new application which should be designated especially for the improvement of the perception of phonemes which are characterized by high-frequency components. Such technology has not been implemented yet.

The first part of this project deals with classification of unvoiced fricatives /s/, /sh/, /f/ and /th/. The features of different phonemes are studied and extracted. Simple classification methods, KNN and LDA are used, providing an average error of 23.5%. Moreover, the relatively new method of diffusion maps is used for dimensionality reduction.

In the second part of this project we aim at developing an algorithm for improving discrimination of the unvoiced fricatives /s/ and /sh/ (aka sibilants), for a possible future implementation in digital hearing instruments.

A practical method to modify the spectral characteristics of the sibilants by manipulating the Linear Prediction Coefficients of these phonemes is introduced. Three methods are suggested: moving the spectral peak that characterizes the sibilants by shifting some of the poles of the LPC filter, expanding the spectral peak bandwidth, and a combination of the above.

For the evaluation of these methods, two tests were conducted on hearing impaired subjects. The results show an improvement in discrimination between manipulated phonemes over non-manipulated ones.

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# 1 Introduction & problem description

Hearing loss is a widespread phenomenon, especially among elders. Typical difficulties appear in hearing frequencies beyond 2.5KHz. This may cause special difficulty in the perception of various phonemes in conversational speech, specifically in discriminating between unvoiced fricatives, such as /s/, /sh/, /f/ and /th/.

Comprehensive statistics on the worldwide hearing impairment prevalence is given in [1]. According to this report, hearing impairment is the most frequent sensory deficit in human populations, affecting more than 250 million people in the world. Consequences of hearing impairment include inability to interpret speech sounds, often producing a reduced ability to communicate, delay in language acquisition, economic and educational disadvantages, social isolation and stigmatisation.



Figure 1: International Symbol for Deafness

Current hearing instruments can amplify different ranges of frequencies and can overcome background noise, but (usually) they do not deal with improving the perception and discrimination of individual phonemes.

Our main goal in this project is to develop an algorithm that may improve the hearing impaired ability to discriminate between unvoiced fricatives and may be used in future hearing instruments.



Figure 2: Hearing instrument

## 2 Hearing impairment simulation

We implemented two simulations of hearing impairment. The first one is spectral smearing, in which the effects of reduced frequency selectivity on the representation of speech stimuli in the auditory system were simulated by ‘smearing’ the spectra of the stimuli.

The second one simulates the effects of loudness recruitment, by using a set of filters with varying width, in order to create loudness sensations in a normal ear that would resemble those in an impaired ear with recruitment.

### 2.1 Spectral smearing

The smearing was designed to evoke excitation patterns in a normal ear that would resemble those evoked in an impaired ear using unsmearred stimuli. This kind of simulation simulates cochlear<sup>1</sup> hearing loss, which is generally associated with a variety of deficits in the ability to analyze sounds, including reduced frequency selectivity. An example of the smearing process is depicted in Figure 3.

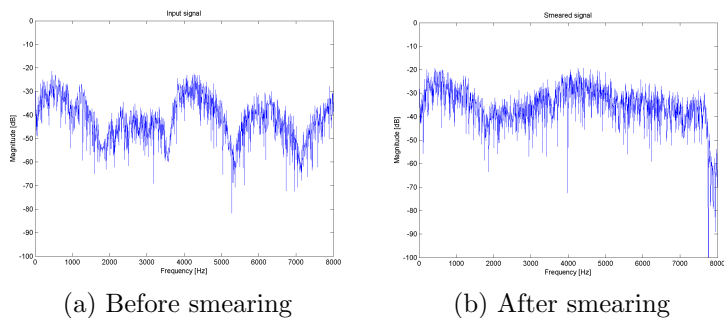


Figure 3: Example of smearing process

The steps for applying smearing to a signal can be found in [2]. Briefly, the model assumes a pattern of auditory filters for both normal and impaired ears of the form:

$$W(g) = (1 + pg)exp(-pg)$$

where  $p$  is some predefined constant, which can differ for lower and upper sides of the filter, and  $g$  is the deviation from the center frequency  $f_c$  of the filter, divided by  $f_c$ . The signal was transformed to the DFT domain after applying Hamming window to it, and its power spectrum was filtered

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<sup>1</sup>The cochlea is the auditory portion of the inner ear.

(multiplied) by  $W(g)$ . Components corresponding to 6875Hz and up were set to zero. The whole process is depicted in Figure 4.

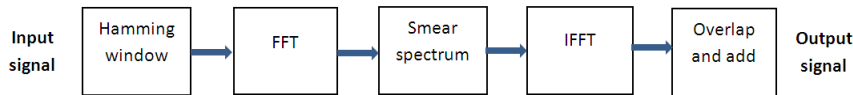


Figure 4: Smearing process

Full implementation can be found in the file `smear.m` which is attached to this report.

## 2.2 Loudness recruitment

When sensorineural hearing loss<sup>2</sup> is present, the perception of loudness is altered. Sounds at low levels (often perceived by those without hearing loss as relatively quiet) are no longer audible to the hearing impaired, but interestingly, sounds at high levels often are perceived as having the same loudness as they would for an unimpaired listener.

This phenomenon can be explained by the *loudness recruitment* theory: loudness grows more rapidly for these listeners than normal listeners with changes in level. This theory has been accepted as the classical explanation [3].

The steps for simulating this phenomenon can be found in [4]. The algorithm is as follows.

First, a set of 13 filters is built. Each filter is made up of four first-order ‘gammatone’ filters in series. The filters were created using Auditory Toolbox (version 2) from [5]. The filters have varying width, according to the *ERB scale* [6]. An example can be seen in Figure 5.

The Equivalent Rectangular Bandwidth or ERB is a measure used in psychoacoustics, which gives an approximation to the bandwidths of the filters in human hearing, using the unrealistic but convenient simplification of modeling the filters as rectangular band-pass filters.

The connection between the bandwidth and the central frequency of each filter is:

$$v = 11.17 \cdot \log \left( 1 + \frac{46.06 \cdot f}{f + 14678.49} \right)$$

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<sup>2</sup>Sensorineural hearing loss is a type of hearing loss in which the root cause lies in one of the carnial nerves, the inner ear, or central processing centers of the brain.

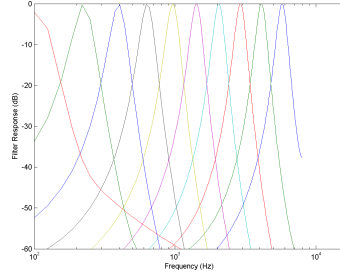


Figure 5: ERB filters

After filtering the signal, each channel result  $f(t)$  is transformed using Hilbert transform obtaining  $F_H(t)$ . The *envelope* is then extracted by:

$$E = |f(t) + iF_H(t)|$$

Moreover, a smoothed version of the envelope,  $E_{sm}$ , is obtained by calculating a running average, using a 10-ms rectangular averaging window. The *fine structure* within the envelope is calculated as

$$fine\_structure = f(t) / E$$

The processed envelope is obtained by:  $E_p = E \cdot E_{sm}^{N-1}$ , where  $N = 1.5$  between 0-900Hz, changes linearly from 1.5 to 3 in the frequency band 900 – 4500Hz, and equals to 3 in the frequencies which are higher than 4500Hz.

Finally, the processed output for each filter is defined as:

$$fine\_structure \cdot E_p$$

and the whole processed signal is achieved by summing all the processed output of the filters. This signal simulates loudness recruitment. The whole process is depicted in Figure 6.

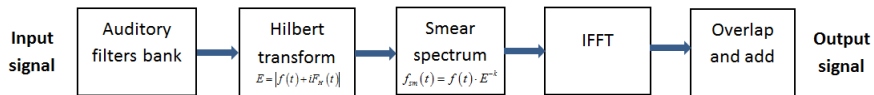


Figure 6: Loudness recruitment simulation

Full implementation can be found in the file `freq_impairment.m` which is attached to this report.

## References

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