Hearing impairment simulation

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6 Hearing impairment simulation

In order to examine the method described in section 4 for exaggeration of /s/ and /sh/ on subjects with normal hearing, 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.

6.1 Spectral smearing

The smearing was designed to evoke excitation patterns in a normal ear that would resemble those evoked in an impaired ear using unsmeared stimuli. This kind of simulation simulates cochlear\(^2\) 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 29.

![Figure 29: Example of smearing process](image)

The steps for applying smearing to a signal can be found in [16]. 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 cochlea is the auditory portion of the inner ear.
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 (multiplied) by $W(g)$. Components corresponding to 6875Hz and up were set to zero. The whole process is depicted in Figure 30.

Figure 30: Smearing process

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

6.2 Loudness recruitment

When sensorineural hearing loss\(^3\) 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 [17].

The steps for simulating this phenomenon can be found in [18]. 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 [19]. The filters have varying width, according to the ERB scale [20]. An example can be seen in Figure 31.

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:

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

\(^3\)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.
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 32.

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