Signal Processing for Hearing Aids

A pair of properly functioning ears is of great importance for various common daily tasks. While the perception of speech is perhaps one of the most obvious examples, being notified by certain sounds like (fire-) alarms or traffic sounds can even be lifesaving. Therefore it is not surprising that hearing impairment may have a strong (social) impact. Typically, hearing problems will occur mostly with elderly people. However, due to the upcoming popularity of portable music-players, like the iPod, permanent hearing loss is currently also a problem for younger people.

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Several solutions exist for the auditory-challenged person amongst which a hearing-aid; a small electronic device typically worn in and/or behind the ear which acts as an amplifier of incoming sounds. While originally these hearing-aids were large heavy analogue devices, current state-of-the-art hearing aids are extremely small (see Figure 1) and are based on sophisticated digital signal processing algorithms. Besides amplification, hearing-aids also support other features like synchronization with all kinds of equipment, e.g., mobile phones and television.

At the 11th floor of the faculty EWI, the Signal and Information Processing lab (SIPlab) is performing research in the field of audio- and speech-related signal processing, led by Dr.ir. Richard Heusdens. One important field of research for SIPlab is hearing-aid related signal processing topics like noise-reduction, intelligibility improvement and multi-channel beam-forming. This research is done in collaboration with hearing-aid companies Oticon A/S (Copenhagen) and Exsilent (Amsterdam).

What is a hearing impairment?

Hearing impairment occurs when there is a problem with or damage to one or more parts of the ear. The main impact is an increase of the hearing threshold, which is the sound level below which a person's ear is unable to detect any sound. The difference between this increased threshold and the threshold of normal-hearing people is the hearing loss. Among the different types of hearing loss, sensorineural hearing loss is the most common (90% of the cases), and it occurs when the cochlea in the inner ear is affected.

Generally, a hearing loss comes together with a decreased functionality of several hearing abilities:

- Decreased audibility: due to the increase in hearing threshold, certain sounds (phonemes) become inaudible for hearing impaired people. Also, for a given sound, some of its frequencies may not be heard. For this, an amplification strategy is needed which amplifies sounds as a function of frequency.
- Decreased dynamic range: despite the increase in hearing threshold, it happens that the loudness level at which sounds become uncomfortable or even painful, stays at the same level. Together with the aforementioned increase of the hearing threshold this results in a decrease of the dynamic range of the ear. This means that compared to normal hearing people, hearing impaired people need to represent all sound levels within a smaller dynamic range. To make this happen, loud sounds should be amplified to a less extent than soft sounds. This is called compression.
- Decreased frequency resolution: for a normal hearing person, sounds at different frequencies activate different regions within the cochlea. The precision at which this happens is called the frequency resolution and allows the brain to separate signals at different frequencies from each other. The decreased frequency resolution in an impaired cochlea makes sounds at different frequencies being interpreted by the brain as a single sound, being unable to separate for example speech from noise.

Figure 1: The hearing aid device

The combination of the above mentioned hearing problems causes hearing impaired people to have a hard time to hear sounds in general and understand speech in particular. Depending on the degree of hearing loss, the impact on speech intelligibility varies. A mild hearing impairment will make it difficult to hear distant speech, even in quiet. A moderate loss will make speech understandable at close distances and loud levels, while severe losses will make conversational speech inaudible for the hearing impaired. The problem of understanding speech becomes even more severe when speech is presented in noise, e.g., in a train, a bar or in a noisy meeting room.

Hearing aid

The task of a hearing aid is to compensate for the above mentioned decreased functionalities of the human ear. To do so, a hearing aid consists of several basic building blocks that, altogether, lead to a highly complex and advanced instrument where a lot of signal processing takes place.

Figure 2 shows a very simplistic block-scheme of a hearing aid. The most important part of the hearing aid is the gain and compression block. Here, the input signal is amplified with a frequency dependent gain such that the incoming sound becomes audible. Subsequently, compression is applied to guarantee that the output signal falls within the decreased dynamic range of the listener.

In addition, prior to compensation of the decreased audibility, most modern hearing aids perform noise reduction. This is necessary since most hearing impaired people have problems to understand speech in a noisy environment. One reason for this is the decreased frequency resolution. Another important reason is that by amplifying the incoming signal in order to compensate for the hearing loss, noise that was not audible before amplification, might suddenly become audible after amplification. Depending on the number of microphones this is called single- or multi-microphone noise reduction.

The third important component in a hearing aid is the so-called feedback management algorithm. Applying feedback management is necessary to compensate for an unwanted side effect that is created by the hearing aid itself: sound that is produced by the loudspeaker inside the ear can leak out through the ear-canal and gets captured again by the microphone outside » the ear. This creates a feedback loop and can lead to a howling of the hearing aid.

To compensate for this, anti-feedback is applied. This is in particular important, because in many situations the hearing aid user does not notice the howling sound. However, the environment does: an embarrassing situation.

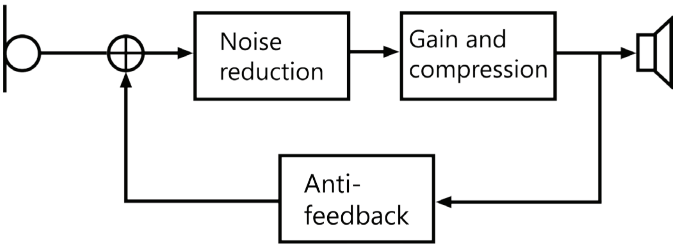

Figure 2: The anti-feedback incorporated into the circuit.

Hearing Aid Research in SIPlab

When the incoming sound of a hearing aid is degraded by environmental noise sources, the quality, i.e., listening comfort, but also the intelligibility of the hearing aid gets reduced. To overcome this, most hearing aids are equipped with a single- or multi-microphone noise reduction algorithm. Single-microphone noise reduction algorithms offer the possibility to reduce noise at certain frequencies, while trying not to harm the target sound, e.g., speech. With multi-microphone noise reduction it is in addition also possible to perform spatial filtering and cancel noise sources from certain directions. In general, noise reduction in hearing aids is based on a combination between a single and multi-microphone algorithm.

Non-stationary noise

Before it is possible to reduce noise, it is necessary to have some information on the noise source itself. Usually, this is the so-called noise power spectral density (PSD). For stationary noise sources, e.g., a fan or air-conditioning system, it is relatively easy to estimate the noise PSD. This can be done during time intervals when there is no speech presence. However, for non-stationary noise sources, e.g., a train or car that passes by, this is much more difficult. One of the challenges the SIPlab worked on during the last few years is estimation of the noise PSD for such non-stationary noise sources.

Intelligibility

One of the major challenges for single-microphone noise reduction is to increase the intelligibility of noisy speech. Although there has already been done quite some research on noise reduction, there is hardly any single-microphone noise reduction algorithm that is able to increase the intelligibility. Before being able to increase intelligibility, it is important to understand how the presence of noise and modifications applied on noisy speech affect the intelligibility. Therefore, researchers from the SIPlab recently developed a distortion measure that reflects the intelligibility of (processed) noisy speech. This distortion measure is very helpful, as it can be used to replace costly and time-consuming listening tests by human observers. In addition, this distortion measure can be used to derive a noise reduction algorithm that can increase the intelligibility.

Spatial cues

To localize sound, e.g., whether a bus approaches from behind, the human ear makes use of so-called spatial cues. Spatial cues are little differences in phase and intensity between the signals that are received in the left and right ear. Noise reduction in a hearing aid does destroy these spatial cues. As a result it might be difficult for a hearing aid wearer to localize from which direction a certain sound comes. For example, a bus that approaches in reality from a certain azimuth angle might sound for the hearing aid user as coming from the front. Besides annoying, this can be dangerous in traffic and working situations. Within the SIPlab, it is being investigated how to optimally perform noise reduction without distorting the spatial cues.

The final result

Future challenges for hearing aid noise reduction

Over the last years, more and more hearing aids are being equipped with wireless technology. Due to the miniaturization and the existence of very low-power circuitry it has been made possible to integrate a wireless transmitter/receiver into a hearing aid. The ability of the hearing aids to communicate wirelessly has an important impact.

First of all, in a binaural hearing-aid configuration, the left and right hearing aid can communicate with each other. This means that the signals captures by all microphones can be combined to introduce directivity in the hearing-aid system. To a certain extent, this is already done within a multi-channel hearing-aid to obtain better noise reduction performance. However, due to the limitation of physical distance (ranging from a few millimeters to 2-3 centimeters) between the microphones within a single hearing-aid, the spatial selectivity is limited. Including the microphone signals of the second hearing aid, located at a distance of about 17 cm, significantly improves the selectivity of the system (beam-forming). In addition, the introduction of wireless technology in hearing aids allows for communication with other electronic devices, like a cell phone, laptop, audio player, etc.

One step beyond the introduction of wireless technology will be the introduction of wireless sensor network technology in hearing aids. Recent advances in hardware technology have led to the emergence of low-cost, small, low-power microphones with onboard sensing, processing and wireless communication capabilities. These devices typically include a radio-frequency circuit, a low-power digital signal processor, a sensing unit, and a battery. Due to their low cost and low computational complexity design requirement, individual sensors may only be able to perform simple local computations and communicate over a short range at low data rates.

When deployed in large numbers across a spatial domain, these relatively primitive sensors can form a large-scale intelligent network that conveys and processes data with high precision and reliability. One possible (future) application could be high-accuracy (adaptive) beam-forming to enhance the quality and intelligibility of speech by placing many (hundreds or maybe thousands) microphones in certain environments, like home environments, churches, cafeterias, etc.

An alternative approach for improving the speech quality and intelligibility is to modify the source signal itself rather than the received signal, as it is currently done in most noise-reduction algorithms. So, instead of doing the processing at the receiver side (hearing aid, cell phone, audio-visual equipment), we can modify the source signal itself in certain applications. Examples of those scenarios are public address systems used in railway stations, airports, etc. The main difference between doing the processing at the transmitter side rather than at the receiver side is that in the latter case the noisy speech (clean speech contaminated by noise) is observed, while in the former the clean speech is available but we do not have access to the noise that will be added later on at the receiver side. How to modify the source signal such that e.g. intelligibility is improved is currently an open research area.

Artist impression of hearing aid.