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A study on signal processing methods applied to hearing aids

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***Abstract* — This paper presents a short survey on current technology available in hearing aids with a focus on digital signal processing techniques used. First, factors influencing the hearing aid effectiveness are introduced. Then, examples of the present DSP methods and strategies are provided. Also, a description of current limitations of hearing aids and future trends of development are shown. Finally, the notion of computational auditory scene analysis is presented as a possible solution for improving quality of speech and music perception while using a hearing prosthesis.**

Keywords* — *digital hearing aids; signal processing; multichannel processing; noise and distortion reduction, feedback compensation

I. INTRODUCTION

Digital hearing aids became now the natural solution for hearing impaired persons. There are however a lot of choices that need to be done by the wearer, i.e. type of the hearing aid, signal processing algorithms used, possibility of programming, connectivity with external devices such as telephones or TV. Each of these factors have an influence on the usability of such an aid (as well as the obvious impact on the price). The more sophisticated solutions are introduced in a device, the more power it needs. Therefore, designers of hearing aids concentrate not only on the DSP methods used, but also on the optimum power consumption, which results very often in the necessary trade-offs [1].

It is said that with present technology a hearing aid should be more than just hearing. However, most hearing aids are designed for improved speech understanding, especially in background noise. But, as Littmann *et al*. [9] pointed out one factor that does not receive sufficient attention is the amount of listening effort required by the patient to realize this goal in everyday use. This problem is starting to be addressed in the context of seeking objective methods of evaluating listening effort [10].

In this paper a survey of current digital signal technology and processing strategies devoted to hearing aids is presented. Also, a description of current limitations of hearing aids and future trends of development are shown.

II. MULTICHANNEL PROCESSING

Hearing loss is usually different for different frequency ranges. For some frequency ranges sound must be amplified, for other attenuated. This is quite obvious if one considers that typical noise sounds (such as wind, fans, road noise) are often in low frequencies, much lower than the normal speech frequency. This might enable to design the filtering strategies that will attenuate or suppress noisy sounds.

Multichannel processing should also be performed due to different hearing characteristics of the device wearer. In some frequency bands sound dynamic range must be increased (using expander), for some it must be decreased (using compressor) to match the loss of hearing of a user of the hearing aid.

Hearing loss can be measured using the categorical loudness scaling. This can be demonstrated as in Figure 1. Figure 1 shows that as the frequency increases, the difference of loudness level between normal listener and a hearing-impaired person grows. There is also one easily observable phenomenon, the so-called loudness recruitment.

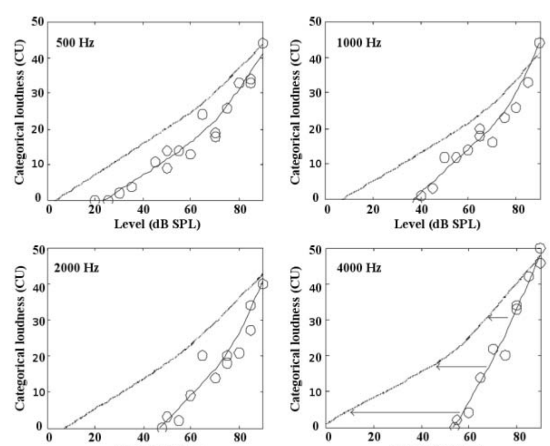

Fig. 1: Loudness as a function of level for a hearing-impaired listener (circles) and normal listeners (dashed line) [2]

Loudness recruitment is a phenomenon, often perceived by hearing impaired persons – while a person can't hear quiet and normal sounds, he/she can easily hear loud sounds or loud sounds are even uncomfortably noisy. This may be very annoying in various situations – e.g. after leaving a relatively quiet space to another space in which the background noise is much louder, ear struggles with the sudden sound level elevation and loud noise may be perceived as uncomfortable.

This results in the need of using an automatic gain control (AGC) for hearing aids as the level needs to be adjusted differently for different frequency bands and gain should depend on the input level. Currently the AGC systems commonly use the spectral analysis for dividing the input signal into separate bands (this has been depicted in Figure 2). Most often the constant bandwidth between 100 and 500 Hz is

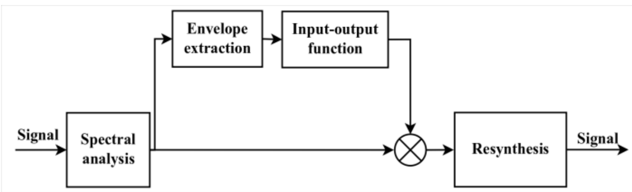

Fig. 2: Signal-flow for multiband AGC processing

used, and then approx. 1/3 octave filters for frequencies above 500 Hz are utilized [2].

AGC systems can work differently depending on the attack/release time. Systems with long time (several seconds) are usually called Automatic Volume Control (AVC) while the ones that have attack/release time of several milliseconds are called syllabic compression as they can adapt as fast as single syllables. There are also systems that combine both fast and slow time constants (dual compression) [2].

III. Environment adaptation

A. Noise and distortion reduction

There are some obvious factors influencing how the hearing impaired person perceives the hearing aid effectiveness:

- one of the biggest problems for hearing impaired persons is a loss of temporal and spectral resolution in auditory processing of the impaired ear; this results in a loss of signal-to-noise ratio especially in noisy environments - it is much lower than for the well-hearing person (approx. 4-10 dB [2]); this reduces dramatically the possibility of just raising the level of sound in hearing aid – this strategy would easily lead to uncomfortable hearing perception in a noisy environment,

- wind-like noise in the background must be attenuated in the hearing aid; however the all-spectrum amplification would easily lead to raising the noise level as well,

- in noisy environments it's usually more comfortable if only sounds coming from angle 0 are amplified, while the rest is attenuated – this also means that the wearer should be looking at the person with whom he/she is speaking; however there are a lot of situations where this strategy does not work – for example when a hearing impaired person is driving a car – then the sound from right/left side should not be attenuated,

- hearing aid is a small device, which results in placing microphone quite close to the receiver; that means that hearing aid will suffer from feedback problems,

- hearing aid should effectively attenuate sudden, strong sounds – as the overall characteristics of the device might amplify those loud sounds,

- hearing impaired persons are very often affected by the damage of the outer hair cells which results in reduction of precise frequency perception; this means that there is an effect of stronger signal masking in the frequency domain (e.g. sounds of a frequency 500 Hz may mask the 1 kHz tones) – which might effectively reduce speech intelligibility.

The above points lead to some strategies that are used in the modern hearing aids.

1) Feedback reduction

Hearing aids are constructed in the way that forces to mount microphone (or microphones) quite close to the receiver. As mentioned before, this increases the possibility of feedback occurring (sound from the receiver goes back to the microphone). Currently advanced feedback cancellation adaptive filters are used in modern devices. However the most effective method of reducing feedback is using the correctly fitted receivers in the ear canal. This reduces the sound leakage from speaker to microphone.

Sound leakage occurs through the vent, i.e. the space between ear and hearing aid. Apparently devices (especially the small ones – in-the-ear) cannot be strictly fitted into the canal. This causes the effect of occlusion, which is responsible for the own voice distraction. However the vent increases the possibility of sound leakage and of the unwanted feedback. The feedback path can be depicted as in Figure 3.

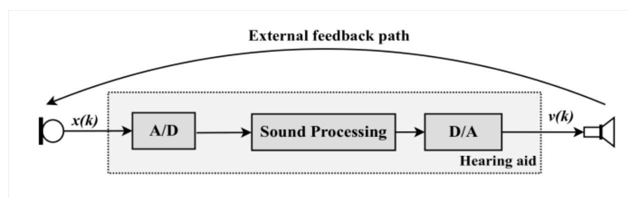

Fig. 3: Feedback path

Feedback occurring in the hearing aid depends on the following factors [2]:

- vent size,
- type of hearing aid (ITE, BTE etc.),
- obstacles close to the hearing aid device (hat, hands, etc.),
- the physical fit of the aid in the ear canal due to external reasons, e.g. jaw movement.

The first two factors are static and thus can be modelled. The two latter must be adaptively taken into consideration. As the feedback path has a usual filter representation, this means that it can also be depicted by the frequency response. Any type of the above factors influences the frequency response.

For the static factors the usual method is to measure the feedback path of the already fitted hearing aid and to limit the gain, so that the closed-loop gain is smaller than 1 for all frequency components. But even if the above has been implemented, feedback usually cannot be avoided as any obstacles might provoke it. Therefore the adaptive dynamic methods must be utilized to ensure that the risk of feedback is minimized.

Two methods are currently commonly used [2]:

- selective attenuation of frequency components for which feedback occurs,

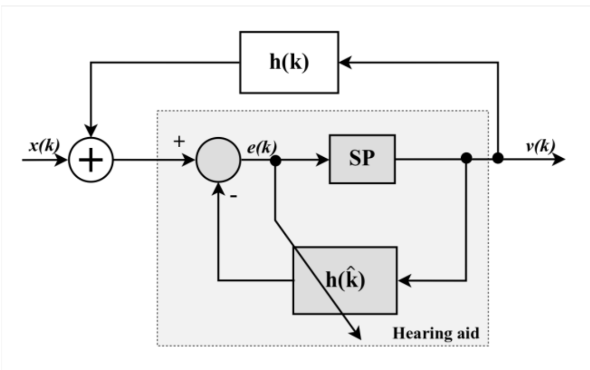

Fig. 4: Feedback compensation. h(k) represents the external feedback path impulse response. h(k̂) represents the adaptive filter

- feedback compensation – this requires modeling the feedback path as a parallel filter and subtracting the result of filtering from the original signal; feedback compensation is realized through utilizing an adaptive filter like in Figure 4.

2) Directional microphones

Directional microphones (physically directional, e.g. using single diaphragm) are not a good solution, as they do not easily adapt to the situation. As was previously mentioned, there are some circumstances in which a hearing device needs to be directional, while in others - omni-directional. This is usually achieved by using the first-order beamforming. The schema of this solution is presented in Figure 5.

Usually two microphones are used for this purpose. Signal gathered by one microphone is then delayed by time T_0. The directional characteristics of this microphone set can then be modified using the a parameter. Examples of such characteristics using the given parameters are presented in Figure 6.

Two microphones compose a first order differential microphone. However there are solutions where second order microphones are used. The schema of such an approach is depicted in Figure 7.

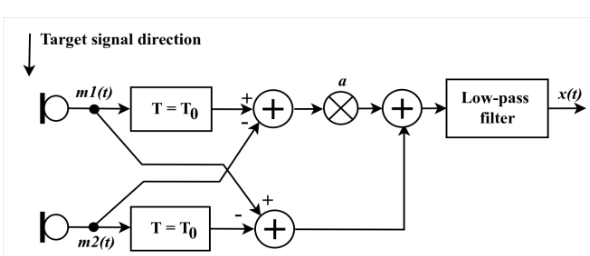

Fig. 5: Differential microphone setting

The second order microphone might be used for frequencies above 1 kHz (due to high sensitivity of the microphones) but these frequencies are very important for speech intelligibility. The directivity index may be higher of 2 dB compared with the normal first-order microphone which can be significant for speech intelligibility.

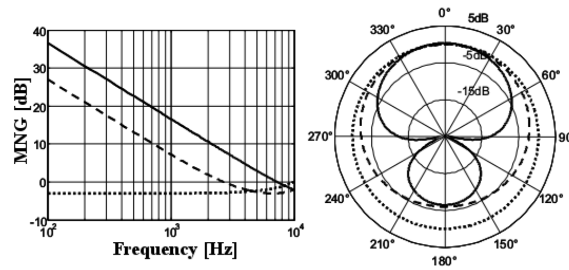

Fig. 6: Different directional characteristics for different parameters setup for the microphone noise gain function (MNG) [1]. Microphone noise gain (left) and directional pattern for $a = 0.5$ (full), $a = -0.5$

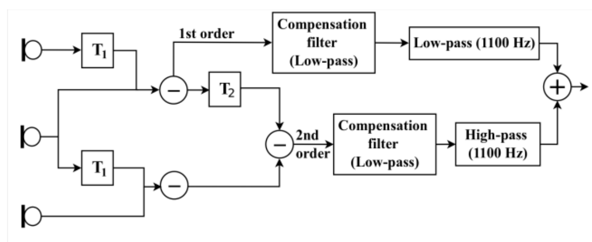

Fig. 7: Second order differential microphone

3) Noise reduction

A basic method of noise reduction is a long-term, modulation frequency-based reduction. The idea is to attenuate frequency components with very low SNR. The difficulty is how to detect the subbands that contain signal with low SNR. This might be performed using the modulation frequency analysis. Speech and music exhibit the modulation frequency above 4 Hz in comparison to more/less stationary noise. This can be used to detect subbands with low SNR and to attenuate signal in those bands. This method works well when noise and useful signal are located in different frequency bands. If not (or these ranges overlap), it may be perceived as not the most convenient method of reducing noise.

A more sophisticated method is based on the Wiener filter. This is based on the assumption that we can calculate the filter coefficients when we have the power spectral density of a noisy speech and noise itself. However in hearing aids the noise signal cannot easily be separated. Thus, various methods are employed, for example a hearing aid may estimate the noise by calculating its power spectral density in pauses (this requires the pause detection mechanism) or it may estimate it using sophisticated statistical methods. Both methods however can provide only long-term smoothed noise power estimates. By changing it to short-term, the algorithms cause musical noise. To recall, musical noise is a perceptual phenomenon that occurs when isolated peaks are left in a spectrum after processing with a spectral subtraction type algorithm. It may be avoided by using various methods such as overestimating the noise power spectral density PSD estimates or limiting the Wiener-filter values to a minimum, the so-called spectral floor [2].

The first method (overestimating the noise PSD estimates) ensures that short-term noise fluctuations do not result in changing the Wiener filter coefficients (the result of musical noise), however it may lead to reducing the overall audio quality of the signal – low-power components might be strongly attenuated. Using the second method (limiting the noise reduction to the spectral floor) generally removes this problem but actually reduces the noise reduction effectiveness.

Some other methods such as for example Ephraim-Malah-based, short-term smoothed noise reduction have been proposed, however they need to be implemented efficiently to be used in commercial solutions.

B. Hearing environment recognition

As a hearing aid is adapting to the new environment it is used in, constructors of modern devices can use it to store the typical environments settings. That's why high-end hearing aids use smart classification systems and training algorithms to help the hearing aid to adapt to different room/space characteristics.

One of the important features of the modern hearing aids is the possibility to auto-adapt to different environments. Generally speaking, the adaptation must be performed using different factors:

- microphone directionality,
- noise reduction algorithms,
- compression factors in given frequency bands.

The problem is when the hearing aid wearer enters into different environments, for example changes a quiet room to noisy street or gets out of the relatively quiet car to the music club. Some preconfigured settings can work well in certain circumstances, but might have a negative influence in another. Therefore there is a need to implement auto-adaptation in digital hearing aids. Most often this adaptation is performed in a general schema (Figure 8).

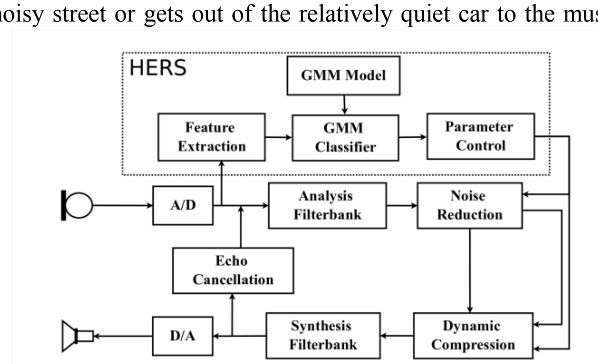

Fig. 8: Processing stages in digital hearing aids with a specialized hearing environment recognition system (HERS)

Typically, the recognition system extracts some feature vectors of the input audio signal and then matches it with the Gaussian mixture models (GMM) to set up available parameters of other algorithms (e.g. dynamic compression thresholds).

Different features might be employed to correctly recognize an environment. For example, one might calculate the envelope of the signal and then use the bandpass filter from 1-4 Hz. Usually clean speech without noise has the envelope oscillations within this range. But if noise is added or if music is considered these oscillations are not common. So this can be one of the features that may enable to decide that a hearing aid is dealing with speech without noise. To make further selection (speech in noise or music) the decision system must take another set of features into consideration.

These features can then be used to classify the environment employing for example the standard Bayes classifier or neural network. It is necessary that these algorithms learn a priori knowledge about the relationship between feature vector values and situation classes in appropriate training procedures, which have to be based on large and representative databases of everyday life signals [2].

Selecting a given set of algorithms is usually performed using the typical on/off logic. However the decision mechanism must not be very sensitive as it will lead to discomfort when constantly switching. Thus the fading mechanisms must be employed that realize smooth passing between different settings.

It is obvious that smart adaptation algorithms are not perfect and the decisions will not always be appropriate but the alternative is to let the user manually select the environmental settings, which in many cases is not really an alternative, e.g. for elderly hearing aid wearers.

IV. ADDITIONAL HEARING AID FEATURES

A. Programmability

Modern hearing aids usually have the possibility to be programmed, either from specialized devices, remote controls or even from mobile phone applications.

B. Zoom

Zoom allows to focus on a conversation in front of the wearer. It usually utilizes the power of directional microphones. „Zoom" is in fact the marketing name for the (usually) first-order differential microphones.

C. Echo blocking

Echo blocking function enables to reduce the echoing sounds in spatial rooms, like big halls, churches. It improves strongly listening experience in such places. This type of processing detects the reverberant „tail" in the input signal. It does not affect speech intelligibility in small reverberant places or in signals where speech is the primary signal, but can definitely help in large reverberant places.

D. Impulse noise suppression

Impulse noise suppression algorithm captures a sudden sound shot, and softens it. This is especially important in the frequency ranges where a hearing device normally amplifies sounds. It should not affect speech clarity and readability.

E. Wind blocking

Listening experience might be easily disturbed by wind, introducing constant but changing noise. Wind blocking can eliminate wind-like sounds. These sounds are created by air movement or vibration on the microphone diaphragm and thus recorded – however they are not real sounds. There exist several strategies for eliminating wind noise [18]:

- dual microphone approach – wind noise recorded by two microphones is usually uncorrelated and is placed in the low frequencies; as uncorrelated low-frequency noise is detected, the hearing aid tries to suppress it. Suppression may be realized e.g. by changing the microphone characteristics to omnidirectional or suppressing certain frequency bands;

- using binaural systems transferring sound data between two hearing aids; as the wind noise comes usually from a single direction, one of the hearing aids is then more affected by the noise; then the system may stream the low-frequency sound from the less affected device to the second one – thus “replacing” the noisy signal with the clear one.

F. Tinnitus eliminating

Tinnitus eliminating function enables to generate soft noise that helps to reduce annoying tinnitus sounds. Eliminating tinnitus is also a part of a broad sound therapy available when using hearing aid. This therapy includes amplifying background sounds making the tinnitus less audible, improving communications with others (thus reducing stress levels) and helping to compensate for the wearer's hearing loss [17].

G. Noise blocking

Noise blocking function blocks all the background noise, like car engines, fans, etc. This is realized by the basic noise reduction mechanisms described earlier.

V. Conclusions and Future Trends

A. Directional microphones

Using two microphones in a single hearing aid has disadvantages - the microphones are close to each other (due to overall size of a monaural hearing aid). The best option would be to have the binaural solution – so two hearing aids would communicate with each other (i.e. communication in terms of internal device settings [3]). Every device could contain single microphone and the directionality will be detected using those two microphones. But this solution impacts the power consumption, because these devices need to communicate wirelessly. This is a big problem due to the hearing aid size and an average battery capacity. However recent studies demonstrated [11] [12] that wireless ear-to-ear audio data transmission resulted in much more effective directivity of a hearing aids, especially when tested in natural cocktail-party like situations. This also allows for employing beamforming algorithms for completely-in-the-canal hearing aids.

One of the hearing aids producers developed a slightly different approach (called StereoZoom): both devices contain two microphones and they form together the 3rd order directional microphone which is used in specific situations when the extreme directionality towards the front is required [16].

B. Multichannel processing

More sophisticated AGC algorithms might also be employed. For example, controlling the parameters of AGC (release/attack times or input-output function) may be performed by smart classifiers. The data link between both hearing aids may also be beneficial to avoid localization problems – it may enable synchronizing the detected parameters.

C. Feedback reduction

Feedback reduction mechanisms may also take into consideration that hearing-impaired persons may use the hearing aids on both ears. If the data link is provided between both devices, then a hearing aid will be able to detect the feedback more accurately. The basic idea is that oscillations detected by one hearing aid can only be caused by feedback if the hearing aid on the other side did not detect oscillations of exactly the same frequency [2]. The concept is depicted in Figure 9.

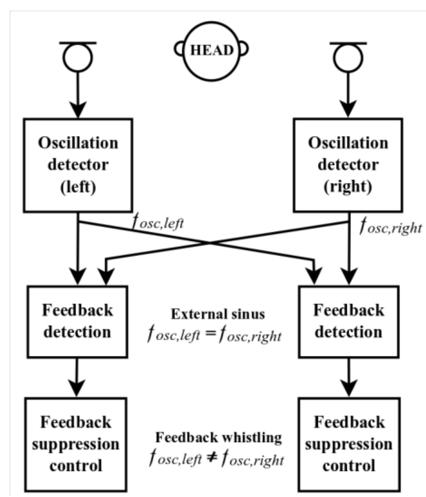

Fig. 9: Feedback reduction using data link between 2 hearing aids

D. Smart adaptation

The future of smart adaptation lies in using multimicrophone signals, with the option to employ binaural processing (thus requiring the data link between both hearing aids). The problem is that the single signal processing may fail in some common situations for example when talking in a restaurant with background music. The algorithms may base on specific detection of number, type and direction of the sound sources. These algorithms are known as computational auditory scene analysis (CASA)[2].

The purpose of CASA is to produce separate auditory streams from the sound mixture that is perceived by the hearing aid wearer. The number of streams must be limited (maximum 4 streams at a time [14]). While separating the streams, some other factors should be taken into account: sound masking in the same frequency band, difficulty in separating sounds (such as pink and white noise), and self-audibility of a given stream – which means that it must exceed a certain sound level to be considered as stream.

There are a lot of approaches to CASA, one of them is the ideal binary mask, described in [14]. The idea is to retain the time-frequency regions of a target sound that are stronger than the interference, and discard the regions that are weaker than the interference.

E. *Music processing*

One of the most difficult areas for the audiologists and hearing aids producers is music processing. For musicians digital hearing aids are usually less acceptable than the analog ones [13]. This happens among others due to the A/D converters that usually can't approach the theoretical limit of 96 dB dynamic range (this limit is due to the 16-bit architecture used). This means that if the signal level exceeds the level of 105 dB (SPL), the hearing aid may produce distortions.

One of the main difference between speech and music is the intensity level. Typically, normal conversation has the level of 65-70 dB SPL with some peaks around 80 dB SPL [15]. However even quiet music can have intensity level exceeding 90 dB SPL. Loud music may have peaks exceeding 120 dB SPL.

However recent studies demonstrated a new approach to A/D converters (TrueInput technology from Widex). It extended the A/D conversion limit to 113 dB SPL while keeping the noise floor low. As a result, the full range of modern hearing aids microphones may be utilized (up to 115 dB SPL without distortion). The results of the study proofed that this technology might be useful for musicians [13].

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