Hands-Free Telecommunication for Elderly Persons Suffering from Hearing Deficiencies

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***Abstract*—Speech communication is the most natural form of human interaction. Communication by means of telephones, mobile phones or video-conference systems is common nowadays especially amongst younger persons. In the past years, also a growing amount of elderly people has started to extensively use communication systems since more and more people live apart from their relatives, friends or acquaintances. However, especially elderly people suffer from hearing loss, which often prevents them from using acoustic communication devices. While approximately every second European adult of age 65+ has a hearing loss that requires treatment, only the minority actually wears hearing aids for different reasons. To tackle this problem, this contribution deals with a personalized and adaptable communication system that enhances the acoustic signal and incorporates the individual hearing loss of a hearing-impaired person. By this, the typical elderly user is enabled to take part in natural communication again.**

I. Introduction

The demographic change leads to a continuous growth of the older population [1]–[3]. As a consequence, the number of elderly who live alone and are mentally healthy but have special needs and disabilities will increase. One of the most prevalent disabilities of the aging population is hearing impairment. It concerns more than 50% of the European population aged 65 years and older [4]–[7]. Because the sense of hearing is the basis for communication and social interaction, the risk of elderly people suffering from loneliness and lack of social integration is particularly high. This risk is especially increased if family members and acquaintances live and work at distant places. To counteract social exclusion, we propose a new kind of communication system which incorporates supporting technologies specifically for elderly persons suffering from hearing deficiencies. Although extensive research and development is recently carried out in the Ambient Assisted Living (AAL) domain with a focus on e.g. tele-care, telemonitoring, embedded systems architectures and integration and home-automation [8], technologies supporting the hearing-impaired elderly in their communication with relatives, friends and care-givers are not sufficiently respected and investigated in the respective scenario. With the aim to investigate new technologies supporting hearing-impaired elderly in future AAL-ICT scenarios and applications, and to lower the risk of social exclusion, this contribution presents a hands-free communication system applicable for people with hearing deficiencies that can be accessed from different places of the living area. For this purpose, strategies for reduction of ambient noise and acoustic echoes are applied and an individual hearing loss of the user can be compensated. It has been shown in the literature that acoustic signal quality can be improved by the individual signal processing stages [9]–[12] that are used in the proposed system and also in combined systems of noise reduction and echo cancellation [13]. It can be expected that addition of individual hearing loss compensation will further increase the communication quality by reducing the listening effort and increasing audibility of the relevant parts of the acoustic signals [14]. A thorough evaluation of the proposed system will be subject of future research.

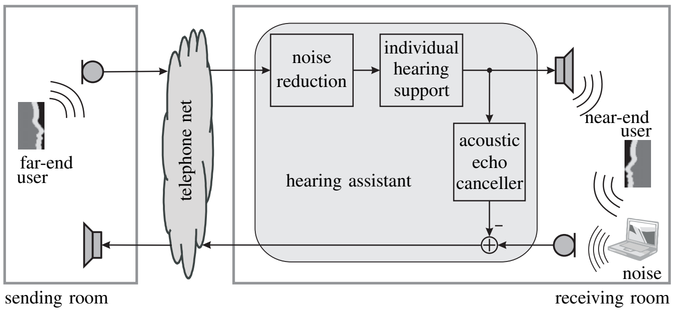

Fig. 1. Schematic of the proposed communication system for elderly.

Fig. 1 shows the general structure of the proposed hands-free system. The signal of the far-end user is transmitted to the near-end room (receiving room) and uttered by the loudspeaker. Here, it is picked up by the microphone again as an acoustic echo and would be transmitted back to the far-end user without further signal processing. The far-end user would have to listen to his or her own voice delayed by twice the transmission delay of the system which would be very annoying. An acoustic echo canceller estimates the echo part contained in the microphone signal and removes it from the microphone signal (cf. Section II-B). Apart from acoustic echoes, ambient noise is picked up by the microphone which has to be suppressed before the signal is presented to the communication partner. This is done by the noise-reduction subsystem depicted in Fig. 1 and described in Section II-A. If the individual hearing loss of the user is known to the system, a compensation that is normally done in hearing aids can be done by the system. By this, persons suffering from mild or moderate hearing losses who are not equipped

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with hearing aids or are unwilling to wear their hearing aids may nevertheless participate in natural and convenient communication.

The remainder of this contribution is organized as follows: the signal processing strategies for noise and echo suppression are described in Section II while Section III focuses on the compensation of individual hearing losses. Section IV concludes the paper.

Notation: The following notation is used throughout the paper: The discrete time-, frequency-, and block-indices are denoted by k, n, and ℓ, respectively. All frequency domain variables are printed in sans-serif letters (e.g. $\mathsf{x}[n, \ell]$). By this, time and frequency domain are distinguishable even if the dependence of the variable k or n is omitted as in $\mathsf{x}[\ell]$. The superscripts ^{T}, *, and ^{H} denote the transposition, the complex conjugation, and the Hermitian transposition, respectively. The operator $*$ denotes the convolution of two sequences, $\mathrm{E}\{\cdot\}$ is the expectation operator.

II. Speech Enhancement

Interfering noises or acoustic echoes disturb natural communication. While this already challenges normal-hearing users, the consequences for people suffering from hearing impairments can be serious and may even prevent them from using the communication systems at all.

Intelligent technological methods can improve the communication quality by removing undesired noise and acoustic echoes. A further pre-processing of the acoustic signals tuned to the needs of the individual user can partly compensate for the detrimental effects of a possible hearing loss (cf. Section III). The speech-enhancement stage of the proposed communication system comprises two techniques to achieve an enhanced speech signal preferably akin to the desired undisturbed speech: single channel noise reduction (cf. Section II-A) and acoustic echo cancellation (cf. Section II-B).

A. Noise Reduction

Single channel noise reduction is a powerful technique to enhance the perceived quality of a speech signal, particularly, when the disturbing signal tagged as "noise" has a different spectro-temporal statistic than the desired speech signal [15], [16].

Fig. 2 shows the general problem for noise reduction: The microphone signal $y[k]$ which is the sum of the desired signal $s[k]$ and an arbitrary noise signal $n[k]$ is processed by a noise reduction filter $\mathsf{g}[k]$ to produce an enhanced speech signal $\hat{s}[k]$. The goal of the filter is to minimize the difference between the output signal $\hat{s}[k]$ and the unknown clean speech signal $s[k]$. Spectrograms as a time-frequency representation of the signals at different stages of the processing chain are exemplarily shown in Fig. 2. Here, dark areas indicate high signal energy while light areas indicate low energy in dB. It is obvious from the spectrograms that the microphone signal $y[k]$ is a superposition of desired signal $s[k]$ and noise part $n[k]$ and that both signals overlap both in time and in frequency. The comparison of the spectrograms of microphone signal $y[k]$

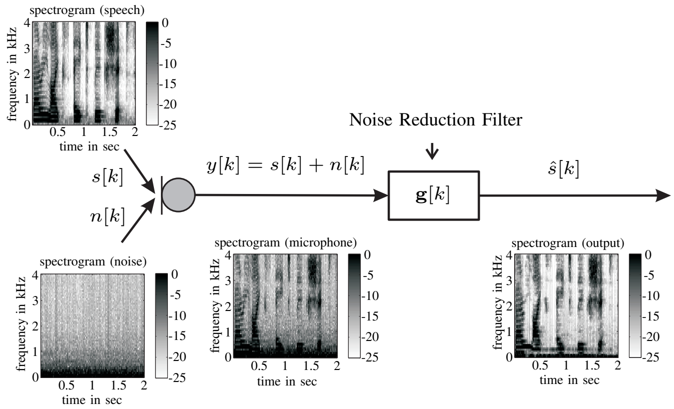

Fig. 2. Schematic of a single-channel noise reduction system and the corresponding signal spectrograms.

and enhanced signal $\hat{s}[k]$ shows that, despite the previously discussed problem, a signal enhancement is possible. This is achieved by exploiting the different signal statistics of desired speech part and disturbance.

In practice, most noise reduction schemes are applied in the block-frequency-domain and rely on the assumption that the power spectral density of the noise part $\Phi_{\mathsf{nn}}[n, \ell]$ is more stationary than the power spectral density of the speech part $\Phi_{\mathsf{ss}}[n, \ell]$. This condition is at least partly fulfilled for many practical situations like factory workplaces (machinery noise), cars (noise from engine and tires) or open-office areas (ventilation, typing, printer noise, etc.), which offers a large potential for noise-reduction schemes based on the so-called short-time spectral attenuation (STSA) approach.

Single-channel noise-reduction schemes estimate the current signal-to-noise ratio (SNR) in several frequency bands n within short time intervals of about 10 to 30 ms and calculate a suppression rule depending on that estimate. The suppression rule defines the amount of attenuation within each frequency band n of a given short time frame ℓ .

Noise attenuation is performed by the frequency-domain adaptive filter $\mathsf{g}[n, \ell]$ which is applied to each short-time spectral frame of the input signal $\mathsf{y}[n, \ell]$. By this, the filter is capable to track changes of the noise and the speech signal. Various different weighting rules exist in the literature that aim to reduce noise while leaving the desired speech signal untouched [9], [17]–[20]. However, as generally the true power spectral densities of speech and noise are unknown, all adaptive filter rules have to deal with estimates and, thus, have to aim at keeping the estimation error small.

One of the basic filters for noise reduction is the Wiener filter $\mathsf{g}^W[n]$. It is designed by the so-called minimum mean squared error (MMSE)-approach, namely by minimizing the mean square of the error signal $\mathrm{E}\{|\mathsf{e}[n]|^2\} = \mathrm{E}\{|\mathsf{s}[n] - \hat{\mathsf{s}}[n]|^2\}$ in every block ℓ and for every frequency bin n [20].

$$\mathsf{g}^W[n, \ell] = \frac{\Phi_{\mathsf{ys}}[n, \ell]}{\Phi_{\mathsf{yy}}[n, \ell]} \tag{1}$$

$$\mathsf{g}[n,\ell] = \frac{\Phi_{\mathsf{yy}}[n,\ell] - \Phi_{\mathsf{nn}}[n,\ell]}{\Phi_{\mathsf{yy}}[n,\ell]} \quad . \tag{2}$$

Other common spectral weighting rules for noise reduction that can be used in the proposed system are spectral subtraction [17] and the so-called MMSE-STSA/logSTSA estimators according to [18], [19]. Common to all of these noise-reduction filters is that they need an estimate of the unknown noise power spectral density $\Phi_{\mathsf{nn}}[n,\ell]$ as obvious from (2). This can either be estimated during speech pauses using a speech pause detector [21] or via continuous noise estimation [22], [23].

Since all single-channel short time spectral attenuation (STSA) noise-reduction algorithms can be expressed by a filter $\mathsf{g}[n,\ell]$ that suppresses parts of the microphone signal $\mathsf{y}[n]$, they will affect both, the noise component and the desired speech signal. Therefore, the aim of any noise-reduction algorithm has to be the optimum trade-off between noise reduction and distortion of the desired signal. Although the filter can be designed to mathematically perform the best trade-off in terms of noise reduction versus not affecting the desired signal, an unwanted side-effect of noise reduction is always a certain amount of cancellation of the desired signal component, which reduces the signal quality. Furthermore, state-of-the-art single-channel noise-reduction schemes still suffer from the so-called musical noise problem. Musical noise is caused by residual noise that is small in amplitude but clearly perceivable by a human listener since it sounds unnatural due to its non-stationary nature [15].

Noise-reduction schemes incorporating models of the human auditory system [9], [13] partly avoid the musical noise problem and, therefore, lead to perceptually better results. This is achieved on the one hand by exploiting the fact that noise parts that are below the hearing threshold are not perceived be the human listener and, thus, do not have to be suppressed. This provides more degrees of freedom to the noise suppression filter. On the other hand, distortions of the signals additionally can be hidden below the threshold of hearing which leads to better sounding signals [13].

Depending on the input signal, noise characteristic and individual hearing ability humans often show different preferences for noise reduction schemes. Therefore, in the proposed communication system different noise-reduction methods can be chosen by the user. The following methods are available: Wiener filter [20], spectral subtraction [17], Ephraim and Malah [18], [19], and psychoacoustically motivated noise reduction [9], [13]. Additionally, the methods can be combined with different noise estimation techniques [21]–[23].

B. Acoustic Echo Cancellation

Hands-free systems often have the drawback that the microphone does not only pick up the desired signal of the near-end user $s_n[k]$ but also ambient noise $n[k]$ and the signal played back by the loudspeaker as an acoustic echo $\psi[k]$. The loudspeaker signal basically contains the far-end speech $s_f[k]$: this is why the far-end user might hear an echo of his or her own speech signal when a hands-free system is used at the near-end side. The acoustic echo $\psi[k]$ is caused by the fact that the loudspeaker signal is reflected at the room boundaries (walls, floor and ceiling) and, thus, arrives at the microphone various times with slightly different delay. Mathematically, the room can be characterized by the so-called room impulse response (RIR), $\mathbf{h}[k]$ [24]. Thus, the signal picked up by the microphone is given by $y[k] = s_n[k] + \psi[k] + n[k]$, where $\psi[k]$ denotes the echo which is the far-end speech signal $s_f[k]$ convolved with the room impulse response $\mathbf{h}[k]$.

A schematic that can be used for cancellation of acoustic echoes and the corresponding signal spectrograms are depicted in Fig. 3. Here, two different filters $\mathbf{c}_{\text{AEC}}[k]$ and $\mathbf{p}_{\text{AEC}}[k]$ reduce the unwanted acoustic echo part $\psi[k]$ contained in the microphone signal $y[k]$.

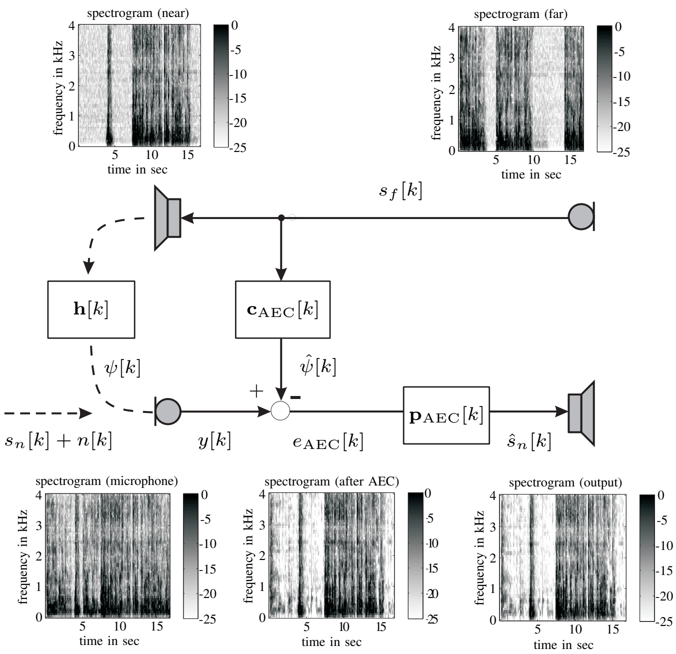

Fig. 3. Schematic of acoustic echo cancellation system and corresponding spectrograms

In summary, the goal of the AEC filter is to reduce the acoustic echo $\psi[k]$ while leaving the near-end speech signal $s_n[k]$ untouched. This can be done as follows: The acoustic echo canceller $\mathbf{c}_{\text{AEC}}[k]$ tries to identify the RIR and, by this, calculates an estimate of the acoustic echo. The output of the AEC filter $\hat{\psi}[k]$ is subtracted from the microphone signal and the resulting signal $e_{\text{AEC}}[k]$ contains a reduced echo part. This residual echo is caused by the fact that the RIR which has to be identified is of infinite length while the AEC filter is shorter in general [11], [24]. Furthermore, since gradient algorithms like the common normalized least mean squares (NLMS) algorithm or its various extensions are applied [20]

$$\mathbf{c}_{\text{AEC}}[k+1] = \mathbf{c}_{\text{AEC}}[k] + \frac{\mu}{\mathbf{s}_f^T[k]\mathbf{s}_f[k]}\mathbf{s}_f[k]e_{\text{AEC}}[k] \tag{3}$$

the echo estimate $\hat{\psi}[k]$ in general does not contain all echo parts. In (3), μ is the step-size of the algorithm which influences its tracking speed, meaning the algorithm's capability to track changes of the time-varying room impulse response. A post-filter $\mathbf{p}_{\mathrm{AEC}}[k]$ is, thus, applied to further reduce the residual echo [25] and support the filter $\mathbf{c}_{\mathrm{AEC}}[k]$. Again, exploiting knowledge about the human auditory system for designing the post-filter [9] leads to perceptually better results.

III. Hearing Loss Compensation

The previously described signal enhancement techniques lead to a benefit for both, normal-hearing and hearing-impaired persons. However, the achieved signal enhancement does not tackle the specific problems of persons suffering from hearing losses that are a wide-spread deficiency amongst the elderly [4]–[6].

A. Multi-band Dynamic Compression

A hearing loss has several detrimental effects on sound perception. The most obvious effect is an elevated hearing threshold, i.e. signals have to be presented louder than normal to be perceivable for the hearing-impaired person. In principle, this could be achieved by a simple amplification of the sound. However, restoring audibility is not enough, since also the perception of sounds above the hearing-threshold can be considerably different due to hearing impairment. The most important effect related to the modified sound perception is called loudness recruitment [26]. This effect results from the reduced dynamic range observed in hearing-impaired listeners. It can be often observed that the sound level, at which sounds become uncomfortably loud, is similar for normal-hearing and hearing-impaired listeners. This is illustrated in Fig. 4, which shows equal loudness contours for a normal-hearing (solid lines) and a hearing-impaired person (dashed lines), i.e. the level, expressed in dB hearing level (dB HL), which is required to produce the same loudness as a function of frequency. While the curves indicating the perception of 'very loud' sounds are relatively flat for both persons, 'very soft' sounds require a larger level due to the hearing impairment. In the example shown in Fig.4, the hearing impairment is strongly frequency-dependent: while at low frequencies the person has normal hearing, the hearing impairment amounts to more than 50 dB HL at higher frequencies. Such a high-frequency hearing loss is typically observed in elderly persons. In general, therefore, hearing-impaired people have a frequency-dependent limited range in which sounds are audible [27]. As a consequence, a smaller level increase is needed to make just audible sounds uncomfortably loud.

For acoustic communication, both an elevated threshold and a modified loudness perception have significant detrimental impact [27]. Even under good acoustic conditions without background noise or strong reverberation, communication can be difficult and tiring for hearing-impaired listeners. Therefore, technical support compensating part of the hearing loss is highly desirable. However, since the perception of loudness not only depends on level, but can also vary considerably with frequency as shown in Fig. 4, the compensation for individual hearing losses is a non-trivial problem.

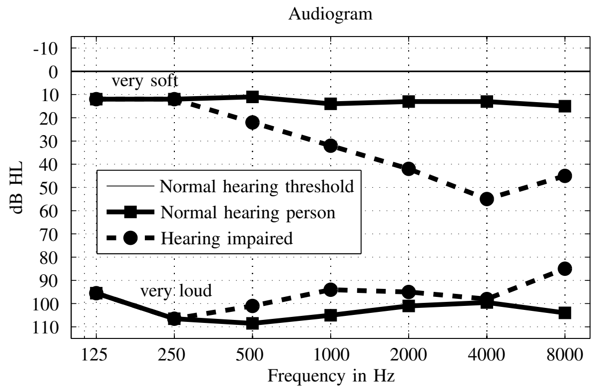

Fig. 4. Equal-loudness contours of a normal-hearing person (solid lines) and a hearing-impaired person (dashed lines).

Since the complex human auditory system is not fully understood yet, one major goal of a hearing-aid algorithm is to at least compensate for the altered loudness impression in hearing-impaired listeners and/or to provide the optimum presentation level of the input signal. To achieve this goal a linear frequency shape is needed in order to equalize the frequency response independently on the input level on the one hand. On the other hand, a nonlinear compression scheme is required in order to compress the large dynamic range of input signals to the reduced or limited dynamic range of the impaired ears. An input/output (IO) characteristic to map loudness perception of a normal-hearing (NH) person to that of a hearing-impaired (HI) person on a log-log scale is depicted in Fig. 5. While the dashed line would not lead to any compression it is obvious from the solid line in Fig. 5 that very soft sounds have a much higher level for hearing-impaired persons (output) than for normal-hearing persons (input). However, the level for very loud sounds can be even slightly lower for hearing-impaired persons than it is for normal-hearing persons.

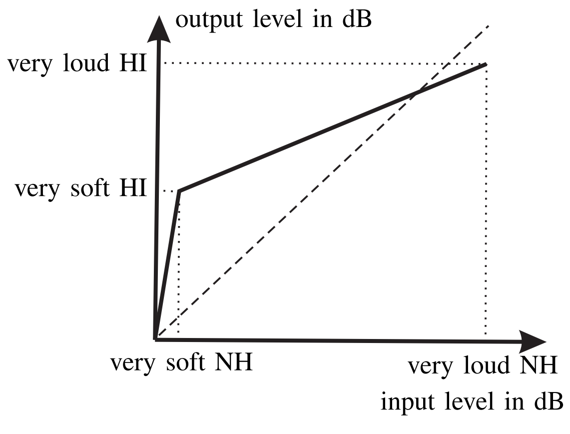

Fig. 5. Mapping between loudness of a normal-hearing (NH, input) and a hearing-impaired (HI, output) listener.

Modern digital hearing aids have different approaches to compensate for the deteriorated loudness perception using dynamic compression. A classic solution is a broadband (i.e. frequency-independent) automatic volume control (AVC). Apart from that, a variety of advanced multi-band dynamic compression methods have been suggested and are commonly used in hearing aids [28]–[31]. The communication system proposed in this contribution includes such a multi-band compression scheme [31]. This means that the system can cover the functionality of a hearing aid and therefore people can benefit from the improved communication quality without having to wear a hearing aid themselves.

B. Algorithm Implementation

Since in general the hearing loss depends on the frequency as shown in Fig. 4, multi-band dynamic compression algorithm working in several bands should be applied. However, although several multi-band dynamic compression schemes have been proposed which usually perform a dynamic compression independently in every frequency band, the sound quality (in most cases also the performance in terms of restoring speech intelligibility in quiet and noisy environments [12]) deteriorates if too many frequency channels with corresponding time constants are used [31]. Hence, a three-channel dynamic hearing aid algorithm is implemented based on the AVC compression scheme [31].

Fig. 6 shows the schematic of the three-band dynamic hearing aid compression algorithm used for the proposed communication system.

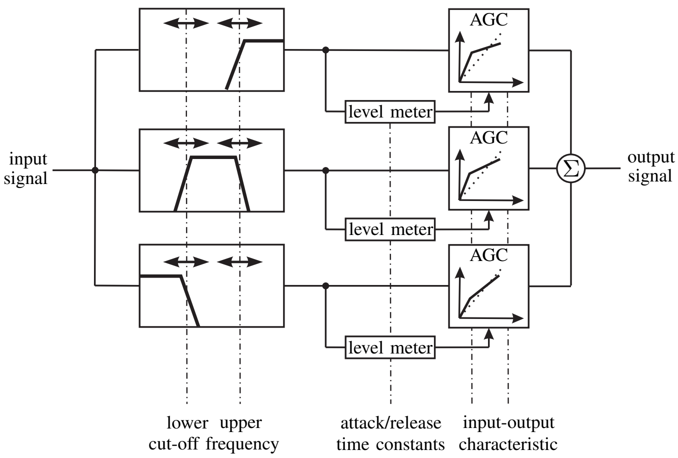

Fig. 6. Schematic of a three-channel dynamic hearing aid compression algorithm.

A filter bank with variable cut-off frequencies f_c is applied as shown in Fig. 6 to obtain band-limited signals $s_i[k]$ for the channels $i = 0$ (low-pass), $i = 1$ (band-pass) and $i = 2$ (high-pass).

$$s_i[k] = \hat{s}_f[k] * w_{\mathrm{FB},i}[k], \qquad i = \{0, 1, 2\} \tag{4}$$

In (4), $\hat{s}_f[k]$, that according to Fig. 1 is the output of the noise reduction sub-system, is the input signal for the dynamic compression algorithm. The filter-bank coefficients are denoted as $w_{\mathrm{FB},i}[k]$. Fourth order filters implemented in cascaded direct form II are applied. The cut-off frequencies depend on the individual hearing loss. For example, for the hearing loss shown in Fig. 4, cut-off frequencies of $f_{c,1} =$ 700 Hz and $f_{c,2} = 2000$ Hz are appropriate (see Fig. 7).

As a second processing step the signal level is calculated in each filter channel according to

$$L_i[k] = \alpha_i \cdot L_i[k-1] + (1 - \alpha_i) \cdot |s_i[k]| \tag{5}$$

with α_i being time constants of a first-order recursive smoothing filter.

Within each frequency channel, an I/O characteristic is defined, which prescribes the desired output level as a function of the estimated signal level on a log-log scale. The current gain in each band is then calculated from the respective input level (5) using the I/O characteristics and applied to the band signals. The output signal is formed by summing up the modified band signals. In this way, the frequency channels are compressed independently from each other.

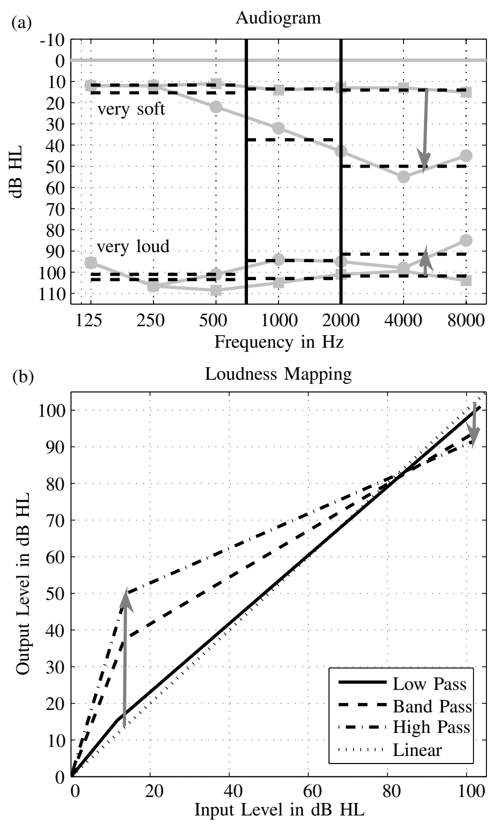

Fig. 7. Example of an I/O characteristic for the three-channel dynamic hearing aid algorithm using an AVC compression scheme.

The calculation of the I/O characteristic for the hearing-loss depicted in Fig. 4 is shown in Fig. 7. The band limits at $f_{c,1} = 700$ Hz and $f_{c,2} = 2000$ Hz are shown in Fig. 7 (a) and a mean loudness per band is calculated by averaging within the specific bands (dashed lines in Fig. 7 (a)). By this, the desired gain can be translated to the I/O characteristic depicted in Fig. 7 (b). Gray arrows exemplarily indicate the gain for the high-pass band in Fig. 7 (a) and (b).

IV. Conclusion and Outlook

A communication system that is explicitly designed for elderly persons suffering from hearing deficiencies was presented in this contribution. Since solutions for ambient assisted living address elderly persons and about 50% of the persons of age 65+ suffer from hearing problems, an individual hearing loss has to be considered by a communication system for elderly. The presented system combines several techniques to improve the communication quality by removing unwanted signal components like noise and echoes, and by compensating for individual hearing deficiencies.

The presented fitting of the algorithm's parameters by means of the audiogram is only a first step towards robust self-fitting strategies not involving professionals. Fitting of hearing aids is normally done by audiologists that have much experience with internal parameters of the algorithms. A thorough evaluation of acceptance and profit of the proposed system using different self-fitting strategies [32]–[34] is subject to future work. State-of-the-art hearing aids apply more sophisticated signal processing strategies and, thus, using hearing aids will be inevitable for persons suffering from severe hearing losses. However, the proposed system can compensate for mild and moderate hearing losses and partly prevent from loosing the ability to perceive certain sounds due to a lack of training of the human auditory system.

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