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A SYSTEM FOR CONTROL OF HEARING INSTRUMENT SELECTION AND ADJUSTMENT BASED ON EVALUATION OF CORRECT TRANSMISSION OF SPEECH ELEMENTS AND FEATURES

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Abstract: Modern digital hearing aids provide unprecedented means of compensating for hearing impairments. However, this comes at the price of adjusting complex processing parameters. To achieve an individual optimum, all processing parameters have to be carefully pre- and re-adjusted. The required time and effort can often not be made available. To provide an independent means of optimizing speech intelligibility, we developed a dedicated solution using mostly off-the-shelf hardware and a dedicated software. With this, a pc-literate hearing impaired person can independently check his aided speech reception. The result can be a valid basis for improvement of the hearing instrument parameter adjustment.

Keywords: hearing impaired, individual adjustment, hearing instruments, speech reception

1. Motivation

When acquiring a new hearing aid, a hearing impaired person usually receives an initial fitting by an audiologist. After a habituation period of two to three weeks, a corrective fine-tuning is advised. There is usually little time for that task. In cases of severe sensory hearing deficits more time than is available is often required.

The audiologist uses a pure-tone audiogram as basis for the hearing aid adjustment. Sometimes, a supra-threshold scaling procedure is added; narrow-band noise is used to avoid the sinusoid that is not well representing natural speech. Most often Word lists are used to achieve a fine tuning of the parameter settings. Standardised word tests are administered to assess a measure for speech intelligibility and the degree of hearing loss. In Germany, typically the Freiburger, Kündig or Sotcheck word lists, and sentence test (e.g. Marburger Satztest) are utilized for historical reasons. In clinical research, further speech audiometric techniques have been investigated (cp. Kollmeier 1995).

We may identify a number of shortcomings in the standard process:
First, given the correct choice of the instrument, there is little time for the audiologist to achieve an individual optimum by utilizing all of the fine-tuning features of modern digital hearing aids.

And secondly, often no attempt is made to introduce an analysis that is able to identify weak points in the transmission of speech on the basis of single speech elements or speech features in order to resolve the deficiencies by readjustment or change of the configuration of the hearing instrumentation. In cases of severe sensory damage with typically narrow dynamic ranges, a precise adjustment of AGC and spectral compression characteristic is the most salient condition for achieving an individual optimum of speech intelligibility. (We assume that resonances caused by the mechanical coupling have been sufficiently flattend.) It is of utmost importance that any valuable feature within the usable spectral range is transformed onto a well audible level that yields best possible individual distinctness.

A third issue is that the user has usually no means of control over the quality of speech transmission as it is given to him by these more or less "prescribed" procedures. So he is often unable to judge whether a different instrument or a different adjustment might improve his or her speech reception.

Beyond the adjustment of the basic input-output characteristics, modern hearing aids offer a lot more: Especially noise suppression algorithms and external "add-on" microphones. Here also efficient test procedures are usually lacking or simply not offered by the standard audiological procedures. The user might want to find out if and when in situations with external noise surroundings, one and which one of the internal noise-reduction procedures may give him a sufficient signal-to-noise ratio....or if the use of external microphones and which of the various configurations is required. The user is here too in urgent needs of more and useable guidance.

In order to address the aforementioned various issues of quality control, a prototypical device consisting of a standard personal computer, dedicated software and inexpensive extra hardware was developed. The device can be used as calibrated sound source, emitting building blocks of speech to be listened to and evaluated by the user. In this regard, the approach can be compared with the Otometry introduced by J.A. Victoreen (1973). But instead of damped sinusoids, actual complex speech feature signals are used; signals representative for all salient speech element classes, not only vowels.

With this device, the user himself can check - in the privacy of his own home or a room in a nearby facility of an interest group etc. - the functionality of his hearing aid, identify shortcomings and find out what programmes to use and when to employ additional devices as FM microphones or to move the microphone nearer to the speaker.

2. Level measurement and hearing loss

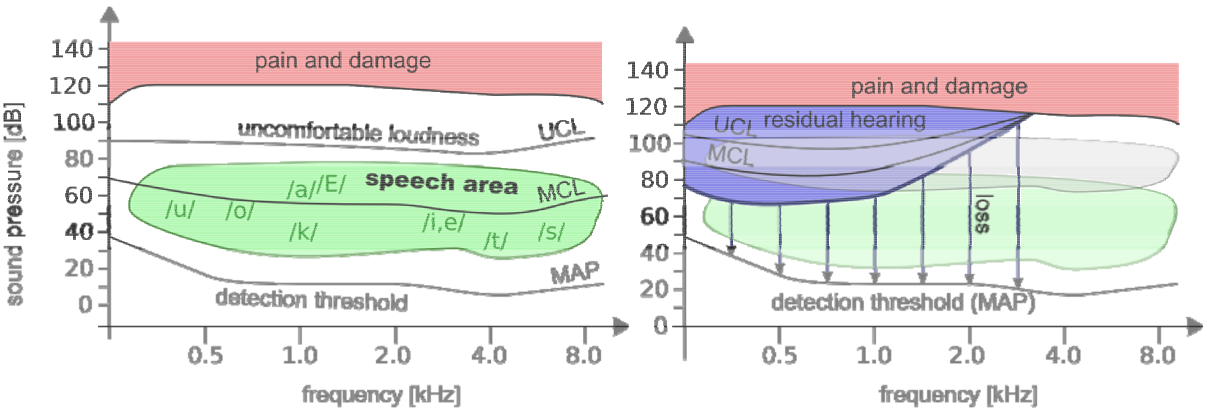

Figure 1: normal and impaired hearing thresholds

To characterize a person's hearing ability, basic frequency-dependent isophonic levels are measured. The threshold, as measured with sinusoids, is called minimum audible pressure (MAP). The upper end of useable sound pressures is marked by the levels above which pain and damage are inflicted on the ear. Below that, the level of uncomfortable loudness (UCL) is usually measured. In between lies the hearing area, here the most comfortable level (MCL) is measured. Salient spectral speech energies are marked by the phonetic symbol of the corresponding sound. It can be seen that these different energies span a considerable dynamic and frequency range. All speech sounds in the speech area (based on natural levels of a speaker in 1m distance) fit comfortably in the hearing area of a normal hearing person (figure 1, left).

When drawing the detection threshold of a person with severely impaired hearing, the residual hearing area is considerably smaller, and the loss to be compensated can be identified directly. Almost all people with cochlear hearing loss suffer from loudness recruitment, i.e. the fact that a sound level increased above the threshold grows in loudness much faster than for a person with normal hearing.

Modern digital hearing instruments can be adjusted to achieve a very good compensation for the frequency dependent recruitment if a significant number of frequency bands is adjusted separately.
As clear and instructive as pure tone measurement charts may appear - the hearing ability is far more complex than a two dimensional area plot can convene.

First of all, the perception of mixed sounds and varying levels underlies various psychoacoustical effects not portrayed here. When hearing a sound of a given frequency, weaker sounds may not be perceived when close by in frequency or time, they are "hidden" or masked by the original sound, which is called spatial and temporal masking, respectively. For persons with sensorineural hearing deficits, the number of independent critical bands is diminished drastically from 24 to e.g. 3. Therefore, these persons suffer from very strong masking (cp. Moore 1998), which results in the inability to separate speech from surrounding noise.

This problem is even more aggravated, if within the speech stream - despite best fitting - perceptive gaps are produced, since some speech elements cannot be made audible by pure amplification. This occurs often in cases of severe high tone losses; speech elements or features that reside in these regions of the basilar membrane are "lost" for cortical processing; even worse: they produce wrong segmentation in word entities. Such a grave disturbance of the natural segmentation can at a certain point no longer be compensated by context-analysis; speech intelligibility drops markedly and these hearing impaired individuals are forced to rely mainly on lip-reading.

In order to optimize speech intelligibility a hearing instrument or add-on device can apply a large variety of nonlinear measures (cp. Plinge & Bauer 2005). Such complex non-linear processing has to be fine tuned to each specific hearing loss individually, making the fitting procedure a complex process with a multitude of parameters to be adjusted. To maximize the benefit for the specific user, testing the users' ability to perceive speech using hearing instruments with add-on devices, and their respective parameter setting, is evaluated.

The pure tone audiogram gives a rough assessment of the hearing loss only at threshold. Word and sentence tests are done above threshold; they can be used to verify in what proportion the goal of restoring speech understanding was achieved. However, they do not show which speech elements are poorly or not at all perceived. A tool allowing sufficient analysis of transmission of speech segments is urgently needed. To fill in that quality-assessment gap and provide persons with impaired hearing with an independent and inexpensive tool to test the combined (over-all) speech understanding ability, the system presented here was designed.

3. Hardware and Calibration

The System can be assembled using a PC or laptop, a USB sound card, high quality speakers, some acoustic insulation, a linear microphone, and (initially) a calibrated sound pressure source (cp. figure 2).

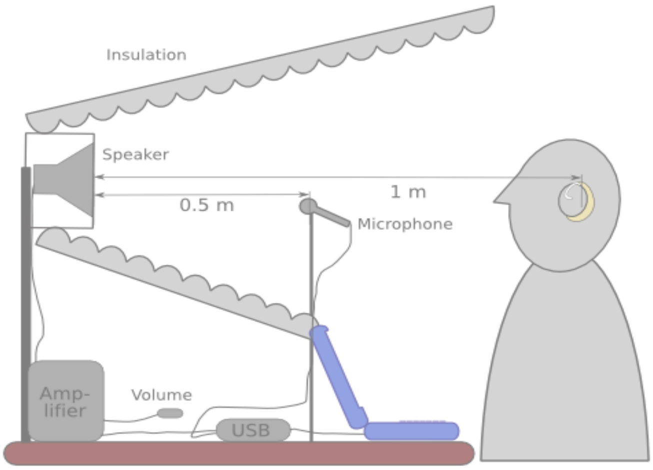

Figure 2, system setup

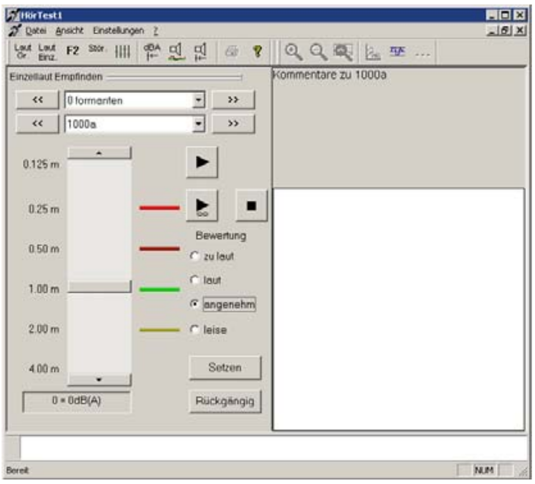

Figure 3, Software, judging a single sound

The use of a USB sound card avoids electrical distortions in the audio signals, which would not only falsify measurements but could also annoy or injure the hearing-aid user. The speaker has to be selected carefully. It should have a near linear frequency characteristic and yield enough sound pressure to produce speech enhanced by 30dB plus 12dB reserve for linearization. A few affordable consumer speakers were found to have sufficient quality. The acoustic insulation is used to reduce reflections (in the otherwise untreated living- or workroom) to a sufficient low level. The (high quality) microphone is used for sound level calibration and frequency linearization, so it should be stable in level and have a linear frequency response.) For this, a cheap alternative to expensive microphones (industry standard for calibrated sound measurements) was devised, consisting of a small battery powered circuit board with a quality electret microphone capsule.

The software contains tools for the general setup and calibration, such as frequency characteristic compensation, level measurement and distortion control. By recording white noise, the speakers' frequency characteristic can be measured (approximately). From this measurement, an FIR filter with linear phase and the inverse frequency characteristic is calculated and subsequently used to linearize the sound output between 500 and 5000 Hz. In order to produce defined sound pressure levels, the system is calibrated using a calibrated sound pressure source. A 1kHz sound of 96dB(A) is produced and recorded by the microphone. So the recorded sound level of the speaker can be deduced, and the speaker volume adjusted to an adequate level.

4. Software Tests

In order to test and evaluate the perception of individual speech elements, prerecorded sounds are played, individually or in groups, with a calibrated normal-speech level set to the equivalent of the speaker's level at a distance of 1m from the hearing aid. To achieve the effect of distances smaller or lager than 1m, the level can be adjusted in terms of "virtual distance", i.e. an increase of 6dB is portrayed as 0.5m speaker distance in the software (according to the square root law of sound pressure variation from a localized source). This is done to make the effects of varying speaker distances more transparent to the user and let him relate to a real world scenario. In this way he/she can experience the maximum speaker distance that guarantees a loudness level that yields good comfort and best distinctness of the sound(s) under control.

Additionally, the effect of adding various surround noises can be evaluated and the (hopefully) beneficial effect of the hearing instrument's noise reduction algorithms.

The virtual distance setting can be classified by the user as either "weak, but audible", "comfortable", "loud", or "too loud". These levels represent the levels shown defining the hearing area in figure 1. In order to convey speech naturally, the first three should be achievable with the hearing aid. If everything is optimally adjusted, "comfortable" should be located in the 1m distance (+/-0dB), "weak" be several meters away (-18dB) and loud near the ear (+12 to 18dB), just as a normal hearing person perceives without any supplemental instrumentation.

4.1 Vowels

Using band filtered vowels with calibrated natural levels, it is possible to test transmission in different frequency bands. The vowel components are played back consecutively in a group and the user is asked to adjust the virtual speaker distance in a way that percept of each is pleasant and distinct. Hereby, the basic compression characteristic of the hearing aid is tested. The hearing instrument should be able to amplify and compress all components of vowel-like sounds so that they are shifted onto a level of most comfortable hearing and most distinct classification. If, e. g. the second formant of /i/ is too weak or of /E/ is too loud, the local compression in that band has to be re-adjusted.

The optimum levels of the test vowels can be individually set, thereby providing a quantitative measurement in virtual distance (and equivalent dB's), so that the required re-adjustments are made transparent. The result can be commented upon and printed together with all levels, which should give any audiologist or hearing aid dispenser a clear picture of what is to be done.

If the user is willing and able, he can also try to establish “weak” and “loud” setting for all vowel-components, thereby testing the ability of the hearing aid to reflect modulation of each frequency band. If a “too loud” setting is achieved too easily, the safety mechanisms of the hearing aid should be checked and the amplification may have to be reduced.

4.2 Fricatives

With the same interfaces, sets of fricatives can be used. While a good adjustment may be achievable for /f/ and /S/, weakly articulated sounds, as e.g. the German /C/, may be only audible when drastically amplified, and the /s/ may even then remain inaudible due to severe high tone losses (here special compensatory replacement measures can be applied, cp. Plinge & Bauer 2005). The hearing impaired user can discover his personal limits with any available hearing aid. If, for example, he can perceive the /C/ only at virtual distances of less than 25cm, it becomes obvious that this may not be practical in everyday situations.

Voiced fricatives can be tested in a similar fashion. It is to be expected that they are audible, but they may not be distinguishable from each other or voiced sounds if the hearing instrument fails to transmit the fricative component.

4.3 Plosives

Since the plosives are vital to segmentation in German, as in all Indo-European languages, it is important that they are well beyond threshold and, favourably, also discriminable. Using plosives in minimal context (/ph/ etc.) the basic perception can be evaluated. Playing all unvoiced plosives in a loop, both their audibility and discriminability can be easily established.

Plosives in vowel context are used to test the temporal characteristics, specifically the required plosive-enhancement property of the hearing instrument. Since severely hearing impaired persons may require excessive amplification for plosive features, an optimal setting may not be technical feasible with state-of-the-art hearing instrumentation. As vowel-plosive combinations are provided for all voiced and voiceless plosives, critical cases can be identified.

5. Hearing in Noise

One very important issue for hearing impaired persons is hearing in noisy surroundings and groups of speakers. Masking and reduced dynamic range are efficiently disabling the person to make out weak speech elements in noise and to separate simultaneous sound streams, as a normal hearing person easily does (cp. Bregman 1990). The inability to locate the sound source or to separate several sources sadly forces him or her to avoid situations which bring her or him in contact with more than one person at a time, which has serious psychological and social implications. Digital hearing aids offer a variety of noise reduction mechanisms (cp. Edwards 2000). These are limited due to the small microphone distances enforced by behind-the-ear technology; efficient reduction of low frequency surround energy without disturbing the wanted speakers' energy remains a problem to be solved by different configurations and future work (cp. Bauer & Plinge 2004).

5.1 Single channel noise reduction

By mixing pre-recorded noise to the choice of phonemes given, the hearing aid's (single channel) noise suppression ability can be evaluated. Since the virtual distances can be widely adjusted, the hearing impaired user can directly compare even unrealistic scenarios, like having the speaker talk in 25cm distance to the hearing aid in a crowded office. By comparing the listening experience with different noise suppression algorithms and different settings, he can learn when to use and what to expect of each, preparing for the real situation.

5.2 Directional microphones

Using a second speaker (or another concentrated noise source) outside the normal frontal range of direction–of-arrival, the function of directional microphones can also be tested. After setting up a noise source in form of the speaker at a given angle of e.g. 80 to 110°, the hearing impaired user can experience up to what level directional side noise is sufficiently cancelled and up to what level he can still have an undisturbed perception of the frontal speaker.

5.3 FM microphones

Ideally, the use of an FM Microphone handed to the speaker should yield an extremely good signal to noise ratio and maximum levels, especially if the microphone is placed about less than 6cm beside the mouth in head-set fashion. It should enable the person with impaired hearing to hear everything that he or she could hear in down to 12cm virtual distance in the previous tests. This can be more or less the case, depending on the quality and construction of the microphone, as well as the quality of the radio transmission and input processing in the hearing aid.

In order to test any given equipment, the user can repeat all tests with the FM microphone mounted to the speaker (6cm beside the membrane). For testing the microphones ability to suppress noise, the second speaker is placed directly opposite the one with the microphone. When running the noise tests, the noise signal is played back atop of the speech signal, and the user can evaluate the quality by varying the noises virtual distance. He can relate these results to real situations like a second interfering speaker nearby, sitting in a car, train or crowded office.

6. Conclusions and Outlook

Using such a system, an individual optimum fitting of all parameters can be evaluated in important hearing scenarios. This should enable hearing impaired users to request first, an individually optimal fitting of all parameters of all programs and second, if required, an alternative choice of their hearing instrument. So they can finally draw best possible benefit from existing technology in hearing instrumentation and furthermore indicate future developments, such as replacement of sounds or features, that are inaudible today, by audible substitutes. As a next step, we will introduce the system into practice in private and clinical applications.

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