FP6–004171 HEARCOM

Hearing in the Communication Society

INTEGRATED PROJECT

Information Society Technologies

Hearing in the Communication Society

EU Final Report for Publication

Date of Submission:	19 July 2010
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Version:	1.0
Total number of pages:	36

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PU	Public	PU
PP	Restricted to other programme participants (including the Commission Services)	
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Project co-funded by the European Commission within the Sixth Framework Programme (2002-2006)		
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Acknowledgement

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Pre-Amble

This report provides a public overview on the outcomes of the EU-funded HearCom project that was conducted from 1 September 2004 until 28 February 2009.

For scientific results the reader is referred to an overview article that will be published in Acta Acustica united with Acustica (Vlaming et al, 2010).

Abstract

Twenty-eight partners participated in the FP6 EU-funded project HearCom, with the goal to improve hearing in our communication society. One of the main achievements has been the provision of advanced hearing screening tests by telephone and Internet. For hearing diagnostics it was aimed at the harmonization of hearing diagnostic tests within Europe. For this the concept of the Auditory Profile has been developed with several tests for various languages. Hearing problems are also a result of adverse acoustical circumstances for which the effects have been studied, modelled and evaluated for hearing impaired. For hearing rehabilitation a large scale comparison study was performed on signal enhancement techniques (algorithms) for hearing devices. Modern technology may assist on hearing and communication by the use of wireless technology and automatic speech transcription. On this it is shown that improvements for auditory communication can be obtained, but that technology should develop further. An overview is given on the HearCom portal with sections for screening diagnostics, hearing information for the public and professionals, and a new service called HearCompanion that provides step-by-step support for the hearing rehabilitation process.

Introduction

Our society is strongly and increasingly communication-oriented. As much of this focuses on sound and speech, many people experience severe limitations in their activities, caused either by hearing loss or by poor environmental conditions. In most cases these environmental conditions only just meet the needs for adult, native-language and normal hearing people, but fall short for vulnerable groups like elderly, hard of hearing, young children, or second-language users.

A consortium under the name of HearCom (acronym for Hearing in the Communication Society) was established, and successfully applied for a European grant to work on new methods for reducing the limitations in auditory communication. HearCom consisted of 28 partners being universities, clinics, research institutes, user organizations and manufacturers in the field of speech and hearing, telecommunications, and Internet. The project addressed the mechanisms that influence communication problems as well as the development of methods for screening, rehabilitation, and evaluation. These methods were developed for major European language areas with the goal to create a common European approach. Internet services played a central role in disseminating the HearCom results to a broad audience. This article will give an overview on some of the main research outcomes of the Project, which covered the period 2004 - 2009.

The focus of the HEARCOM Project was on:

- The identification and characterization of auditory communication limitations
- The identification, modelling, and evaluation of ambient conditions that limit auditive communication in everyday situations
- The development of standardized testing and evaluation procedures for hearing impaired persons.
- The development of rehabilitation and signal enhancement techniques that compensate adverse ambient conditions and personal disabilities
- The development of innovative assistive personal communication technology based on wireless communication links and assistive applications integrated in mainstream technologies
- The development of Internet services that assist individuals and professionals in the improvement and compensation of communication problems.

The research, technical development and innovation activities are divided up into 13 workpackages, organized in 5 sub projects. These can be seen on the diagram below:

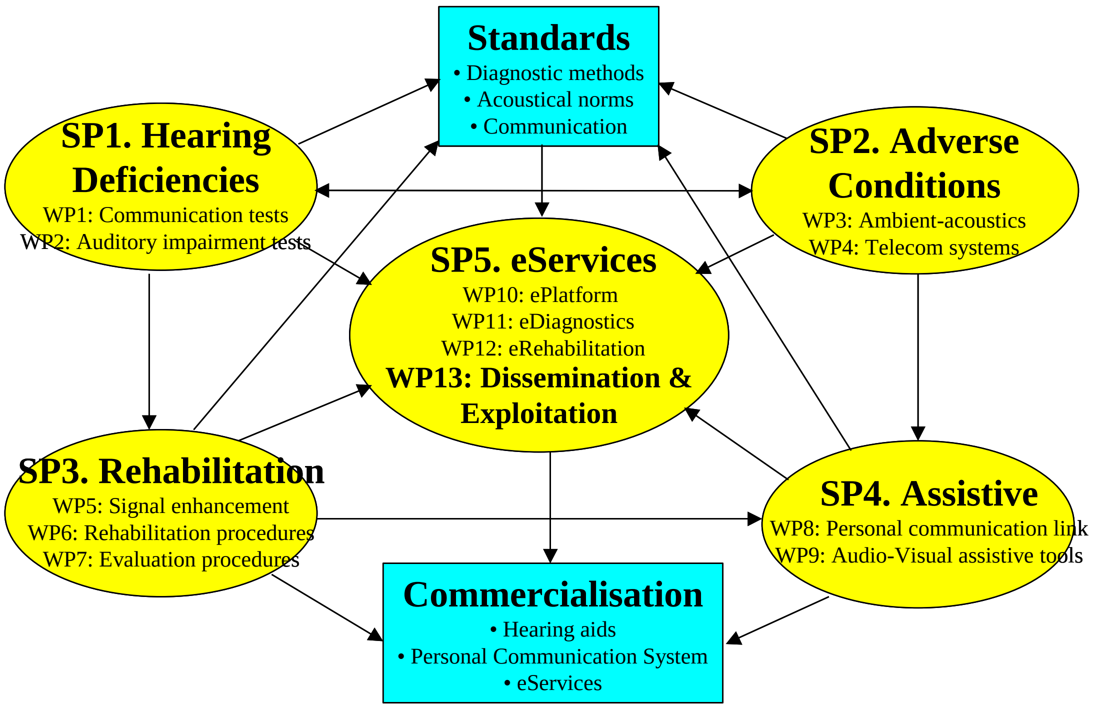

Figure 1: Structure of HearCom

The uniqueness of the HearCom program was that (1) it addressed a broad area of research topics, all related to possible problems prohibiting full participation in the modern communication society, and (2) it was targeted at a broad range of end-users, from individuals interested in hearing, professionals in clinical audiology and in hearing-aid design, and members of the engineering community (architects, acousticians, telecom-specialists) shaping our products and our environment.

In a nutshell, the over-all structure of the research within HearCom can be characterized as follows: Central theme is the advancement of our knowledge on hearing impairment underlying the problems in auditory communication. That knowledge is then used in the R&D aimed at improving the situation: to provide better room acoustics and telecom systems, better hearing devices, and dedicated assistive technology applications. This paper reflects that same structure and the HearCom research in these different fields will be reviewed briefly.

Hearing Screening and Diagnostics

In the field of hearing deficiencies, the pure-tone audiogram is considered a fundamental aspect of characterizing a hearing loss. The classical hearing aid aims at compensating that loss by appropriate sound amplification. However, it is generally recognized that in many cases this will not fully restore communication performance (Moore, 1995), especially in marginal situations with ambient noise, reverberation or other disturbing factors. It is understood that, when distinguishing between conductive losses (mal-functioning of the mechanical chain of sound transmission to the cochlea) and cochlear or retro-cochlear impairments, only the first type can be fully compensated by the sound amplification of the classical hearing aid. The second types of impairment are often associated with so-called supra-threshold deficits: even sounds well above the hearing threshold are still processed sub-optimally in comparison to the normal ear. Identifying the nature of the impairments underlying such supra-threshold deficits is a main issue in today’s auditory research.

Hearing screening by telephone

It is well known that about 15% of European population have a hearing problem. From this group only a relatively small proportion of about 30% will seek treatment. When treatment is sought, problems may already persist for several or many years. Untreated hearing loss may harm society and personal life economically and psychosocially.

Obviously there is a large reluctance to seek help. The main reason is a lack of awareness on hearing problems, but also stigma, costs, usability and the availability of hearing aids and hearing care do play a large role. Moreover hearing problems tend to build up rather slowly, by which people will not recognize first signs and will postpone or forget to take action. A major step forward for the timely discovery of hearing problems can be the provision of hearing screening by telephone and Internet. Such screening will be easy accessible, has low costs, can be done at any moment and is repeatable, and - importantly – is done in private without family or other people, which makes it an objective and convincing method.

Screening can be made either subjectively by using questionnaires or objectively by means of a functional test. Questionnaires have been used for screening widely in the past but have low attractiveness for people to complete, and may lead to underestimation as people may try to ignore their hearing problem. A functional test will be more objective, but in general will require a calibrated presentation of sound samples and a human supervisor to record the scores. The problem of calibration can be avoided by applying speech in noise tests provided that the overall audibility of presentation is good. Also the test can be performed without supervisor when the answers are recognised automatically when the user selects from a closed set of response alternatives (e.g. pictures or digits).

Based on this basic principle Smits (*Smits* et al., 2004) developed a new hearing screening test, based on the presentation of digit triplets by telephone. This test consists of 23 triplets spoken at variable levels amidst a constant background noise level. The levels of the triplets are varied up-down depending on giving an incorrect or correct answer. The average speech to noise level of he last 5 to 23 triplets is used to calculate the speech reception threshold for triplets (SRT in dB SNR), corresponding to the 50% threshold of recognition. Reference values for normal hearing have been obtained. As a test result three ratings are possible: good, intermediate and poor. When poor the user is advised to contact the GP and possibly to obtain a full diagnosis by an audiologist, ENT-doctor or hearing aid dispenser. In case the result is intermediate, the user is assured that this non optimal result is quite common but that he may ask for more information at his GP. The effect of the new test was evaluated by asking a group of callers to complete a questionnaire as sent afterwards by mail. This evaluation showed that from the responding callers those with a poor or intermediate result about 50% indeed have contacted their GP and/or hearing professional (*Smits* et al., 2006).

This telephone screening test is language dependent and therefore new tests were developed within the EU HearCom Project for use in several European languages: English (2005), German (2008), Polish (2008), French (2009) and Greek (2009). Also special versions have been made by other parties in cooperation with HearCom for Welsh (2009), Swiss-German (2008) and US-English (2010). Figure 1 shows the introduction dates of the telephone hearing screening tests in Europe. Depending on publicity in newspapers, radio and television, these tests have found large usage and the total amount of callers to date is 1 million or more.

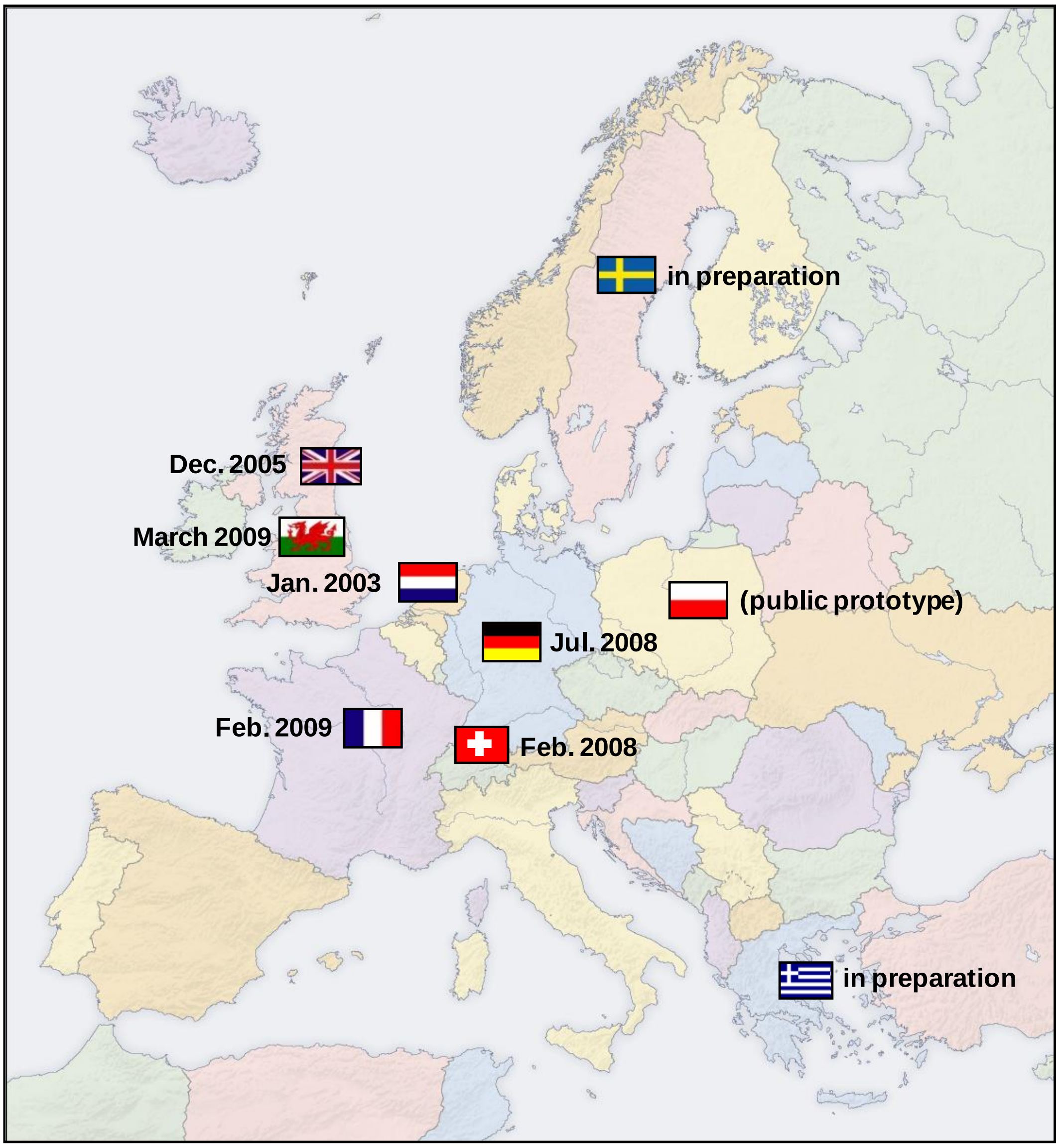

Figure 2: Introductions of hearing screening test by telephone in Europe

Hearing screening by Internet

In a next step the triplet screening test was reworked for Internet presentation, which has numerous advantages compared to telephone screening: better audio quality compared to the 300-3400 Hz bandwidth for telephone, binaural presentation by loudspeakers or headphones. In addition there are more possibilities to add explanatory texts, graphics and follow-up tests and also the test is free of (telephone-cost) charge. These Internet versions required new balancing and calibration of the broadband digit triplets (100-16000 Hz) and subsequent validation tests for normative groups of normal and hearing impaired test persons (see HearCom D11-6). These Internet digit triplet tests have been launched through the HearCom Internet portal (www.HearCom.eu) as well by follow up projects. Figure 3 shows the keypad shown on the computer screen for entering the heard 3 digits of the triplet, presented in background noise which varies in level according to the up-down procedure for determining the SRT for 50% correct understanding of the triplets.

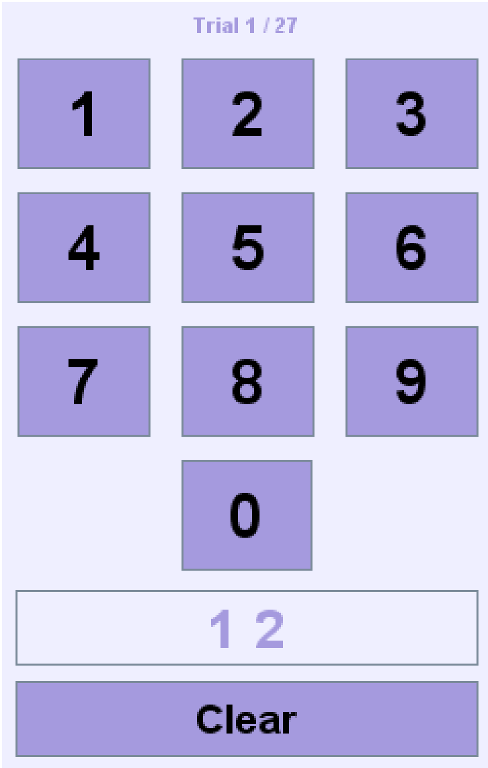

Figure 3: Internet keypad for entering the 3 digits

In supplement to the digit triplet tests, the Internet makes it possible to make additional screening tests. Three additional screening tests were developed:

1) Sound localization in the horizontal plane: as long as the stimuli are above hearing threshold the localization performance is to a large extent independent of overall level (Vliegen & Van Opstal, 2004), thus can be performed also for Internet. Hence two variants of localization tests were developed for the HearCom Internet portal:

   a) The minimum audible angles (MAA). This test evaluates the minimum angle between two successive sounds that can be detected by hearing. For this a fixed loudspeaker setup is used, for which by an advanced cross-talk cancelation technique the angle between two (virtual) sound sources can be varied to less than 1 degree. For normal listeners the minimum angle is between 1 and 4 degrees, but for people having hearing problems at one ear or people having auditory processing problems the angle will be larger.

   b) Absolute localization test in which the listener must indicate the direction of the sound source. Based on an acoustical virtual environment (AVE) a localization screening test was designed for the Internet (Borß et al, 2008 and 2010).

2) Questionnaires can be used without the need for the presentation of sounds, thus is relatively easy to be implemented for Internet. Drawback is that questionnaires are self reporting and may underestimate hearing problems that are not noticed or are ignored. In combination with an objective test like the triplet screening test a reliable estimate of hearing disability and handicap can be obtained. Within HearCom a hearing handicap questionnaire was developed of 18 questions for Dutch, English and German. This questionnaire was evaluated with a test group of 129 testpersons. The HearCom questionnaire has a sensitivity of 0.85 and specificity of 0.81. Details on the development and validation of the new HearCom hearing handicap questionnaire are found in D-11-5.

While all screening tests listed here provide a speedy assessment of hearing abilities, it should be noted that even a combination of several screening tests examine only a subgroup of all the abilities necessary for normal hearing. For instance the hearing threshold and the uncomfortable level are not tested. Therefore, even if someone obtains normal test results in the screening tests, it is important to give the advice to have the hearing checked professionally if the person feels to have a problem with hearing.

All above screening tests have been made available to the general public at www.HearCom.eu.

Harmonised hearing diagnostics

Hearing diagnostic tests largely differ in Europe due to language and cultural differences. HearCom has made an important step in harmonizing the various clinical hearing tests by defining the Auditory Profile which characterizes the individual's auditory impairment profile in a comparable way across Europe. By this it is the goal to extend and to unify hearing care in Europe. A broad application of the Auditory Profile will reveal epidemiologic data about the incidence and prevalence of hearing deficits.

In HearCom, the approach was as follows: First, we agreed upon a concise set of tests to define an individual's communication performance. For this, we did build upon the broad experience in the Consortium with speech tests e.g., the Speech Reception Test with short meaningful sentences (Plomp, 1986), or the Oldenburger sentence intelligibility test (Kollmeier & Wesselkamp, 1997, Wagener et al, 1999), sound localization tests and representative types of ambient conditions (ICRA-noises). Second, we defined a relatively large set of auditory impairment tests, including tests on relevant cognitive functions. These tests were administered to listeners with normal hearing (reference data), and to large groups of listeners with various degrees of hearing impairment. The analysis of this database showed the relations between type and degree of impairments and reduced communication performance.

The components of the Auditory Profile should be relevant for auditory communication performance. With this in mind six fields were selected as indicated in Table 1. In the right column the selected tests have been defined. These tests were then implemented on a common test platform for clinical use in at least four languages.

Diagnostic tests of the Auditory Profile	
Field	**Test**
1) Loudness perception	• Acalos (Brand & Hohmann, 2002)
2) Frequency resolution and temporal acuity	• Combined F and T-test (Larsby & Arlinger, 1998)
3) Speech perception	• Sentence test: SRT in quiet, stationary, fluctuating noise (Plomp & Mimpen, 1979) • Matrix-OLSA Test (Kollmeier & Wesselkamp, 1997)
4) Binaural processing	• Minimum Audible Angle (MAA) (Grantham et al, 2003) • Interaural Level Difference (ILD) and Binaural Intelligibility Level Difference (BILD) (Johansson & Arlinger, 2002).
5) Subjective judgments	• Gothenburg Profile (Ringdahl et al, 1998) • Effort scaling for speech in noise (Larsby et al, 2008)
6) Cognitive abilities	• Lexical decision making test (Larsby et al, 2005)

Table 1: List of tests included in the auditory profile

The tests of the Auditory Profile have been evaluated in two multinational multi-center studies. Variations between languages and clinic have been identified and assessed. The multi-center studies yielded a large number of outcome measures per ear and per subject (30 normal-hearing and 72 hearing-impaired participants, D-11-6). A factor analysis explained 73% of the variance. The factor structure revealed dimensions that could be interpreted as related to the processing in the high frequencies (dimension 1), audibility (dimension 2), low-frequency processing (dimension 3), and recruitment (dimension 4). For the hearing-impaired group speech perception in stationary noise is mainly determined by the hearing loss at 3 kHz and by the frequency resolution at 500 Hz. In fluctuating noise also the temporal resolution at 3 kHz becomes relevant. These data show that speech perception is not only determined by audibility, but supra-threshold processing is also very important.

All test procedures that have been included in the Auditory Profile (except for pure tone audiometry) are available as plug-ins for the clinical software package Oldenburg Measurement Applications (OMA). OMA software was developed with the aim to offer audiologists an instrument to conveniently conduct new methods in hearing diagnostics using a flexible and modular system, independent of whether their workplace is a clinic, a research facility or in the hearing acoustics branch. All test procedures are available as modules of a single software package, providing the same look and feel for all tests and enabling the storage and analysis of test results within one single database. Demo versions of the tests of the auditory profile can be found at http://hearcom.eu/prof/DiagnosingHearingLoss.html.

Hearing Acoustics

Room acoustics

With respect to room acoustics, the STI-procedure (Steeneken & Houtgast, 1980) is a common approach for translating the physical measures of a room in terms of intelligibility. The STI (Speech Transmission Index) is an index between 0 and 1.0, and its relation with speech intelligibility is well established for normal hearing persons. In actual situations, the STI can also be measured directly with dedicated equipment. The STI-approach, defined in the IEC standard IEC 60268-13 (2003), is closely related to the SII (Speech Intelligibility Index, ANSI S3.5, 1997), which is a calculation scheme concerned primarily with the effect of interfering noise on speech intelligibility. Using these tools, designers aim at acceptable acoustical conditions in classrooms, meeting rooms, theatres, public-address systems covering large areas, or in public transport. However, the design criteria do not take into account the special needs of more vulnerable groups, like elderly, hearing impaired, children in classrooms, or non-native listeners. Within HearCom the STI and the SII models have been extended to encompass the information losses arising from hearing impairment (Festen & Plomp, 2002) or from being a non-native listener (van Wijngaarden et al, 2002). The relation between an individual’s hearing deficit (the auditory profile) and the influence of room acoustics on speech perception has been investigated systematically, aiming at guidelines and tools for designers to meet criteria which also reflect the needs of the more vulnerable groups.

Standardized versions of the STI and SII are monaural models, based on single-channel estimates. SII and STI were designed to predict intelligibility in diotic listening conditions (i.e., same signal at left and right ear), based on measurements with a single microphone. This means that any binaural intelligibility benefits are disregarded. The benefit of listening to speech with two ears instead of one in conditions with background babble is known as the cocktail party effect. A significant body of scientific research on this topic, spanning half a century, provides ample resources to draw from for devising binaural intelligibility models. Models that cover aspects of binaural hearing will extend the scope to other applications, yielding more accurate prediction results. Within the HearCom project binaural extensions of the SII and of the STI have now been developed.

Binaural Speech Intelligibility Model (BSIM) based on the SII

The binaural extension of the SII (Beutelmann, 2006) evaluates binaural speech and noise signals and predicts the speech intelligibility benefit for spatially separated speech and noise sources in anechoic conditions as well as in realistic rooms. Predictions can be made for both normally-hearing and hearing-impaired subjects, based on the audiogram.

The binaural extension of the SII does, in principle, not change the SII method, but acts as a front-end which determines the additional signal-to-noise ratio (SNR) improvement due to better-ear listening and binaural interaction. It was developed on the basis of the work by vom Hövel (1984). The binaural speech and noise signals are divided into ERB-wide frequency bands with the help of an auditory (gammatone) filter bank (Hohmann, 2002). In each of the frequency bands, the maximally achievable SNR is computed using the Equalization-Cancellation (EC) principle (Durlach, 1963). The EC process aims at eliminating the noise signal due to destructive interference by subtracting one of the channels from the other, after equalizing a potential interaural time delay and level difference.

In order to match the model performance to human data, the process contains artificial inaccuracies of the equalization operations. The audiogram is incorporated in form of a hypothetical internal noise, which sets an upper limit for the SNR in each frequency band. The SNRs in each frequency band are passed to the SII, from which the speech intelligibility or a speech reception threshold (SRT, the speech level or overall SNR at which 50% intelligibility is reached) can be calculated.

Binaural STI

The STI method assumes that the intelligibility of a transmitted speech signal is related to the preservation of the original spectrotemporal differences (modulations) between the speech sounds. These differences may among other things be reduced by bandpass limiting, masking noise, and reverberation. In the STI model the reduction of differences is quantified by looking at the modulation transfer in a number of frequency (octave) bands. For developing a binaural version of the STI, it was aimed at improving prediction in cases where sources of speech and interference (noise, reverberation) are separated spatially.

A binaural version of the STI (Bin-STI) was developed based on interaural cross correlograms (Jeffress, 1948), where signals in left and right ear are measured and divided into octave bands. The interaural cross correlation in each band is calculated, yielding a number of internal (time-delayed) spectral representations. These representations are processed as if corresponding to a single-channel STI measurement. By selecting the representation with the maximum modulation transfer per octave band, an overall binaural STI can be computed.

The new model was validated for a range of dichotic listening conditions (i.e., different signals at left and right ear), featuring anechoic, classroom, listening room, and strongly echoic environments (cathedral). Comparison of subjective speech intelligibility measurement (CVC word scores) with predicted scores showed good correspondence for these binaural conditions, much better than with the standard STI. The outcome for monaural conditions is identical to the standard STI. See D-3-5 for more details.

Telecom acoustics

Like in room acoustics, modern telecom systems are planned and implemented in order to satisfy the needs of normal-hearing users. The planning process is facilitated by models, which aim at predicting an index related to speech communication quality on the basis of the physical channel characteristics. The ITU-T recommends the so-called E-model for this purpose (ITU-T G.107) (International Telecommunication Union, 2002). Within a working group of ITU-T (Q.8/12) headed by one of the partners, this model has recently been extended in order to provide planning guidelines for interconnected PSTN/ISDN/VoIP networks. Still, planning is limited to "standard" normal-hearing users, and to traditional handset telephone interfaces. An extension of the planning predictions towards the needs of vulnerable users is highly desirable, in order to not exclude this group from the benefits of modern telecommunication services.

Both instrumental models E-Model and PESQ (Rehmann et al, 2002) achieve good to very good correlations with subjective data as long as normal-hearing subject groups are concerned. The results of the E-Model deviate from subjective data with increasing hearing loss of the subjects, whereas PESQ also achieves good predictions of ratings from mildly-to-moderately hearing-impaired subjects if the loudness of the speech is set to a comfortable level. However, neither model can predict subjective quality ratings from hearing-impaired subjects when the speech presentation level is varied around the optimal setting.

As a consequence both models have been extended for use with hearing impaired listeners. These models have been used to perform a series of experiments, with different groups of listeners, on the interaction between hearing impairment (the auditory profile) and channel degradation with respect to speech intelligibility. The new PESQ for Hearing impairment (PESQ-HI) shows good correlation between objective and subjective speech quality measurements from listeners with mild and moderate hearing loss. The correlation of the extended E-model for hearing impairment is promising but more data would be needed. More details are found in D-4-3 and D-4-5.

The results are used to formulate recommendations for possible signal adaptations at the receiver’s side and will be introduced in the ITU standardization activities (ITU-T Study Group 12).

Signal Enhancement for Hearing

Modern digital hearing aids go far beyond the classical simple sound-amplification devices (Dillon, 2001). They may include extensive signal-processing algorithms adapted to the needs of the individual hearing impaired, and to the prevailing ambient acoustical conditions. Processing capacity and speed are still improving, leading to a great variety of possible algorithms and parameter settings (Levitt, 2001).

Leading questions in this field are 1) exactly what types of algorithm are most effective for the individual hearing impaired, 2) how to estimate the effectiveness of such algorithm, and 3) what rehabilitation procedure ensures the best solution for an individual.

Signal enhancement techniques

During the past decades large numbers of signal processing schemes have been proposed to re-establish intelligibility in adverse listening conditions, some of them with great, other with limited success. In the HearCom project five different, representative classes of signal enhancement techniques were considered:

- Adaptive feedback cancellation. These are techniques to combat feedback using a neutralizing adaptive electronic feedback path incorporated in the device. By using feedback cancellation it is possible to provide more amplification without the risk of annoying whistling feedback sounds.
- Adaptive beamforming. By combining signals from two or more microphones in the hearing aid or cochlear implant device, noise from a specific direction is suppressed, even if the direction of the source is changing.
- Single-channel noise suppression. These algorithms require only one microphone, and attempt to separate interesting sounds from unwanted noise based on their statistical characteristics.
- Blind source separation. This is a technique to separate sound sources, without knowing where they are. It is applied using microphone signals from two hearing aid devices at both sides of the user's head.
- Coherence-based dereverberation. This kind of approach suppresses diffuse noise and sound reflections from the room walls by combining signals from microphones from two hearing aid devices at both sides of the user's head.

Several algorithm variants belonging to the above mentioned classes have been implemented and evaluated. These variants differ in their theoretical approach and in the precise settings of signal-processing parameters that control their behaviour.

Sound demo materials have been prepared for each of the considered algorithms to show performance for different listening situations. This material has been made available at the website www.HearCom.eu, where also details on each algorithm can be found for information of professionals.

Evaluation of signal enhancement techniques

Reference implementations have been made for several algorithms. These implementations have first been evaluated by model simulations in relation to different categories of the auditory profile. Based on these model evaluations the algorithms were optimized and fitting rules for hearing aid use were made.

From a selection of the implemented algorithms, real time processing schemes have been implemented on the Personal Hearing System (PHS) as developed in HearCom. This PHS is based on a small portable computer that was developed for research evaluations of hearing aid signal processing and fitting (Grimm et al, 2006 and 2009).

Listening evaluations of 4 types of algorithms were made for 109 persons (normal and impaired hearing) at 4 four different labs. Tests were obtained in an office-like room for listening scenarios of target speech for a 3-talker respectively 1-talker babble noise source set-up. Results show that only the multi-channel adaptive beamforming algorithm (MWF) provided a benefit in measured SRT of more than 6 dB, see Figure 4. On the other hand the Blind Source Suppression (BSS) algorithm shows no improvements in the 3 noise source situation, but when only one noise source is present the Blind Source Suppression (BSS) algorithm provides 6 dB improvement which is a slightly better than the 5,5 dB improvement of the MWF algorithm in that situation.

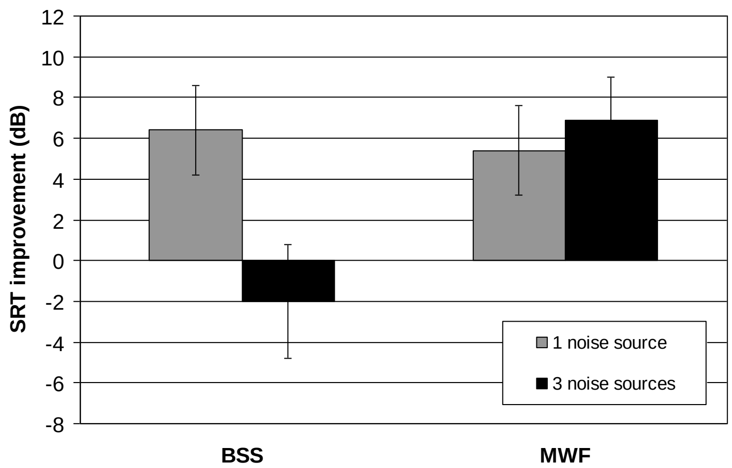

Figure 4: Comparison of SRT improvement in two noise scenarios (1 noise source and 3 noise sources) for 2 best algorithms: Blind Source Separation (BSS) and Multichannel Wiener Filtering adaptive Beamforming (MWF). Shown is the mean SRT improvement relative to the unprocessed condition.

The benefit required to fully compensate loss of speech understanding is dependent on the degree of hearing loss. For most people that use hearing aids, the loss of speech understanding in noise varies between 3 to 8 dB SRT. Obtaining an improvement of 6 dB thus may promise a significant reduction of hearing disability in speech understanding.

From the other 2 algorithms the single-channel noise suppression and the de-reverberation algorithm do not show objective improvements in terms of improved speech understanding. However these algorithms still have benefit in providing greater ease of listening effort, which significantly contributes to the subjective appraisal of hearing aid usage. (Luts et al, 2009 and Eneman et al, 2008b).

In practice hearing aids should be capable to switch between different algorithms based on the noise situation and personal preferences of the user. This should be done automatically for which additional algorithms will be needed to recognise typical listening situations.

Assistive Hearing

In many adverse listening conditions, assistive-technology applications are not sufficient to alleviate the limitations of a hearing and communication handicap. A number of solutions are available, like the use of hearing aid telecoil and infrared systems. Also remote microphone systems are available with a direct coupling to a hearing aid. The use of modern technology as available in GSM/GPRS/UTMS telephones, smartphones (personal digital assistants, PDA), netbooks and BlueTooth devices, makes it possible to adapt mainstream communication systems to the specific group of hearing aid users. For this a Personal Communication System (PCS) has been conceptualized that is based on PDA or smartphone technology. For this PCS a number of new assistive services have been exploited:

- A personal link to connect left and right hearing aids with each other and the PCS. The link is used to connect with mobile phones, remote microphones and other audio equipment. At present this technology in different variants has been introduced by several hearing aid manufacturers. In HearCom a optimized audio codec has been developed for this link (Li et al 2009)
- A handheld Personal Hearing System to supplement hearing aid signal processing (see Grimm et al 2009)
- Public Announcement service
- Bimodal speech and text using Automatic Speech Recognition.

The latter 2 of these services will be discussed here.

Public Announcement service

In many public places such as airports, train stations, event-halls, and churches, auditive announcements are made that are relevant for a restricted geographical area. Understanding these messages is not always easy, in particular when having a hearing disability or when being disturbed by background sounds. In supplement to existing public announcement (PA) systems, a wireless PA system is feasible that delivers the announcements both in auditive and textual format. These announcements can be given directly to the ear by headphone, or hearing aid, without acoustical interference; textually information is given visually at the handheld display of a personal communication system (smartphone, PDA). These modes can also be combined. The messages also will be stored and can be repeated and searched at any moment which will increase efficiency in time and effect. Also alternative local information such as time-tables can be offered as well. Such a wireless public announcement system can be deployed relatively easily based on standard local communication technologies such as Wi-Fi, Bluetooth. The client application will be relatively small and is downloadable to PDA or Smartphone.

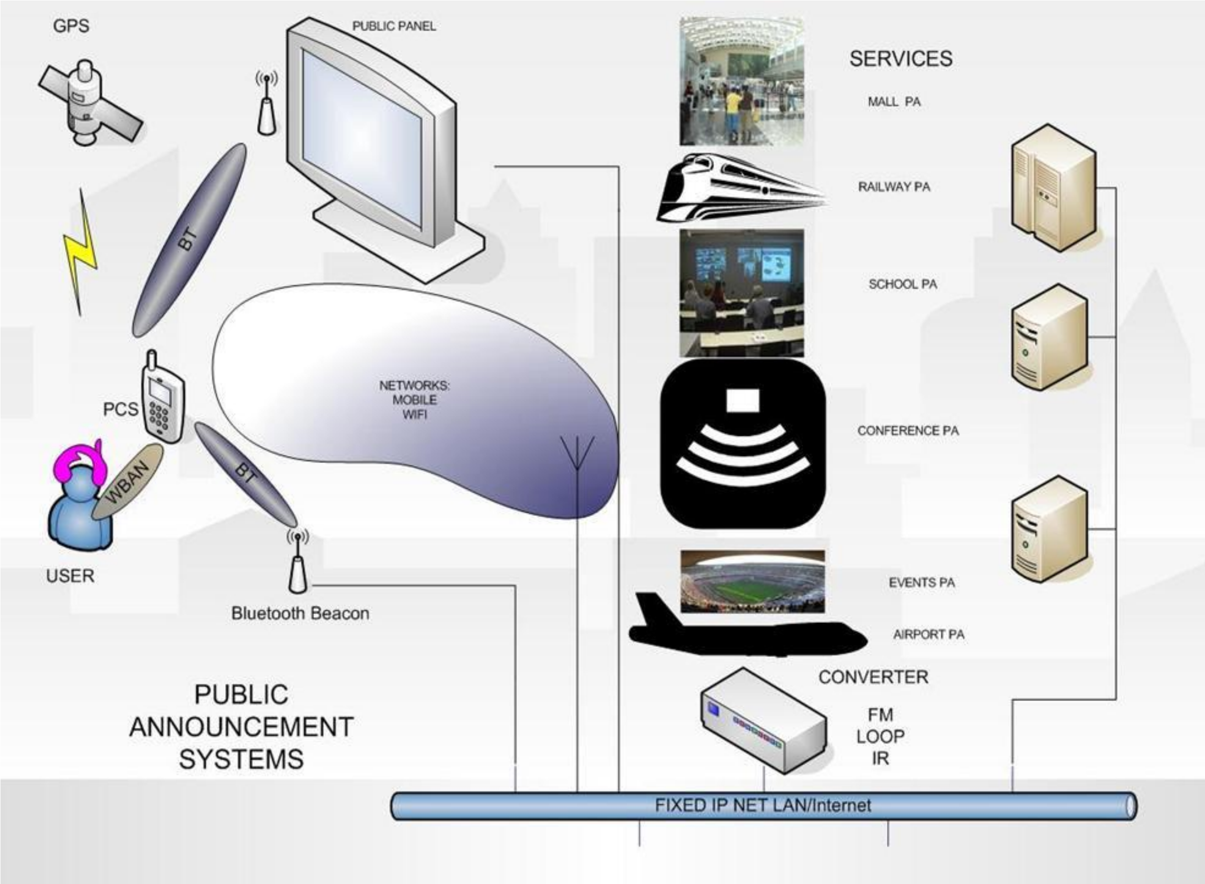

Figure 5: Wireless Public Announcement Service for people with hearing disability

A prototype wireless announcement system has been developed and demonstrated with end-users. This demonstration system is based on the use of a PDA having Wi-Fi connectivity. The announcement service center essentially is extended with Wi-Fi transponders and a server entry station for auditive and textual announcement messages. At the client side the messages are presented in audio through headsets or directly on hearing aids (using hearing aid telecoil neck-loop, or alternatively by hearing aid FM or Bluetooth add-ons). Textual information was presented on the users' display. All messages were stored in the handheld device and could be retrieved multiple times or searched for specific information. The wireless communication protocol is based on CAP (Common Alerting Protocol). The location of the user is determined by the identifiers of the Wi-Fi base-station or alternatively also by GPS technology. A roll-out of the service is hampered by availability of public Wi-Fi hotspots that can be adapted for this service and be offered to the public. For a more general solution also mobile telephone Internet connectivity should be researched further.

This wireless announcement service can also be adapted for use for vision impaired persons which have problems reading public display panels. For that a more short range (display panel area) version of the service is required.

Bimodal presentation of speech and text

Individuals that experience problems in speech understanding may be supported by the presentation of speech related textual information. These speech understanding difficulties can be caused by background noise and/or a hearing loss. The extra text can be generated by automatic speech recognition (ASR). For that purpose use is made of state-of-the-art speech recognition systems. However today's ASR systems still have problems in correctly converting speech into text, due to large variations in pronunciation and due to vocabulary size. At present ASR systems may reach a best practical accuracy in the order of 80% correct for well spoken language and medium sized vocabulary. By combining imperfect ASR textual output and partly intelligible speech, these two modes may complement each other (bimodally) such that a normal conversation will become possible.

Implemented in an assistive listening device based on a PDA (Personal Digital Assistant) or an intelligent mobile phone, the ASR system (by a remote Internet service) will recognize speech and display its output (text) on the screen of mobile telephone or PDA. The extra displayed text will so support the understanding of speech for the listener (see Figure 6).

Figure 6: example of presenting transcripted speech on a PDA-display.

However it is not proven that visually displayed text from an ASR system, having several recognition errors, really will assist in speech understanding. In particular, the presence of background noise and unclear pronunciation of the speech will increase the ASR system error rate. Additionally, the speech recognizer needs some time for processing, which leads to a delay in the text presentation. Therefore a study has been made that investigated to what degree visual information that is incomplete and presented with some delay may contribute to speech comprehension. In this study it was focused on the influence of the amount of recognition errors by the ASR system and the delay in the displayed text on the benefit obtained from the text. In addition the subjective listening effort is measured to examine the effort required by the listeners to process additional and partly incorrect visual information.

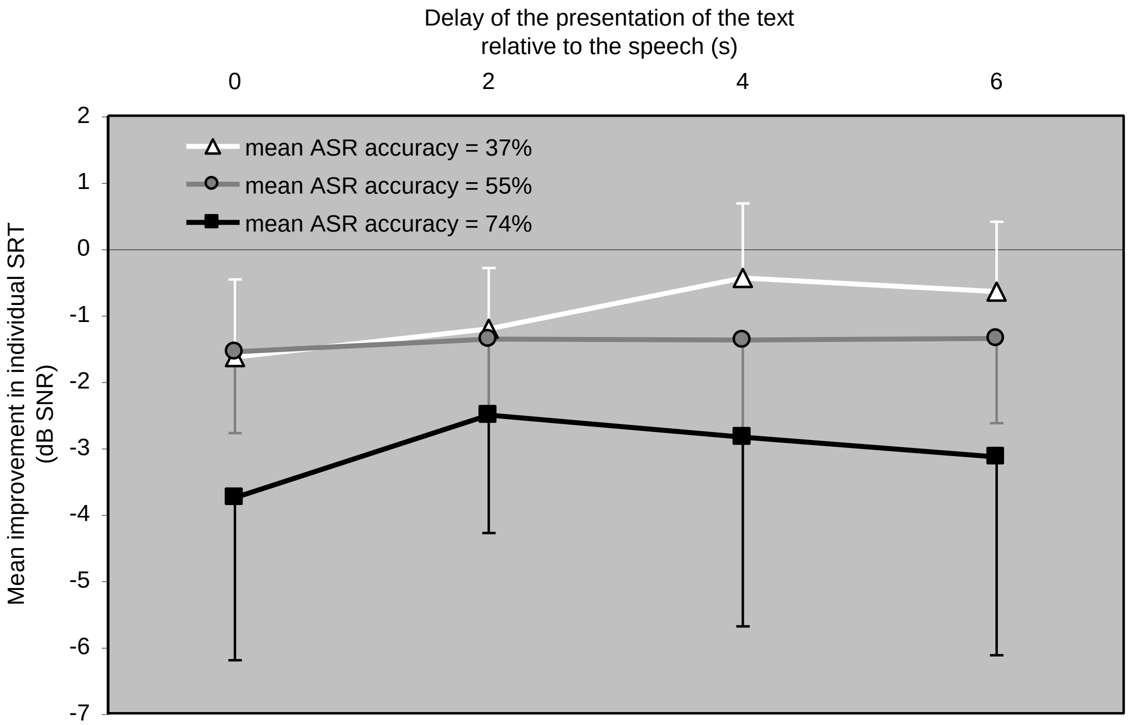

Figure 7: Mean Improvement in using speech recognition at 3 levels of accuracy and at 4 values of processing delay. A negative value in SRT improvement corresponds to a lower SNR needed for 50% speech understanding.

Figure 7 shows the benefit of speech understanding as lowering of the SNR (negative is better) for 3 levels of speech recognition performance. As speech recognition takes processing time this improvement is presented as a function of recognition delay. The improvement in speech understanding is expressed as the extra SNR of background noise that is tolerated. Hearing impaired persons have a deficit of about 4 to 8 dB SNR compared to normal hearing, which means that an improvement of 3 dB roughly reduces handicap by 40 to 70 %. (Zekveld et al, 2008, 2009)

From this figure it can be concluded that a benefit of about 3 dB can be obtained when speech recognition (ASR) accuracy is around 74%, with little influence of delay. However when accuracy is 55% or 37% the benefit reduces to 0.5 to 1.5 dB SNR and delay negatively influences performance.

From these results it is concluded that the simultaneous presentation of noise disturbed speech and partially incorrect text will benefit speech understanding. However to be practically useful (more than 2.5 dB improvement required) the accuracy rate of the speech recognizer should be 70% or better and preferably with minimal delay. At present this will not be achievable for speech recognition in daily conversations. However for specialist applications (i.e. with limited vocabulary for recognition) this technique may improve speech communications.

Hearing Internet Portal

A HearCom portal was developed to provide hearing services and to disseminate relevant information on hearing. The HearCom portal is divided into three main sections:

1. User section for the general public. This section contains the various hearing screening services, informational pages on hearing and the eRehabilitation service.

2. Professional section: contains background information on screening and clinical tests, demonstrations and other relevant information.

3. About the project: information on the Project and dissemination of reports.

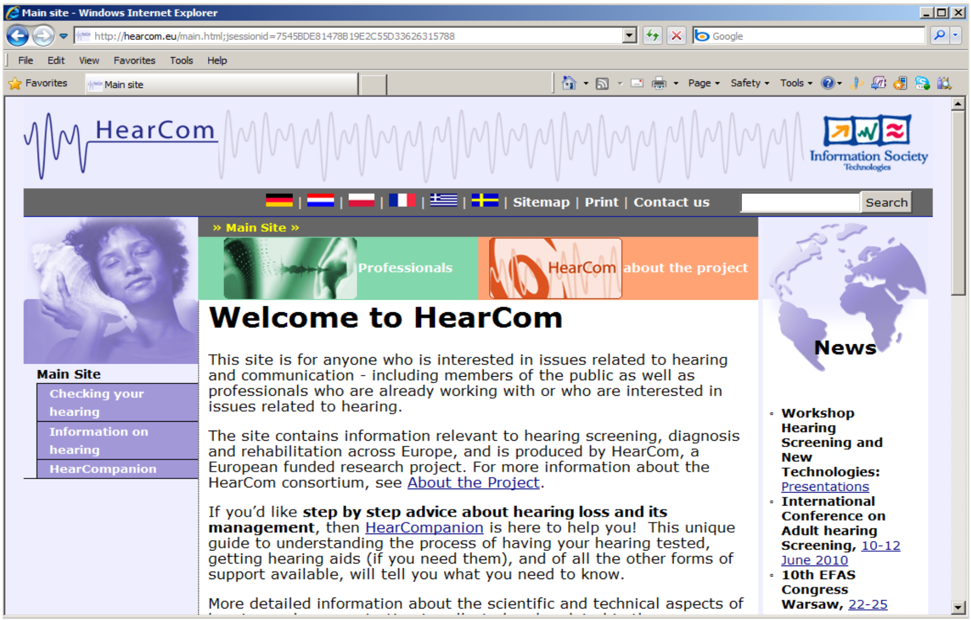

Figure 8: HearCom internet portal

The HearCom portal (figure 8) is available in several languages (English, Dutch, German, Swedish, Polish, Greek and French); the total number of web pages is more than 2000 publicly available web pages. The amount of pages in a specific language differs, so there are approximately 1000 web pages in English, 300 – 500 web pages in every language of Dutch, German and Polish, approx. 100 web pages in Swedish. For French and Greek only the triplet hearing screening test is offered (about 10 pages). The leading language is English, in which the majority of content is available.

User section

The user section is intended for individuals that experience problems with hearing and for those who desire information on hearing and hearing problems. The user contains 3 main subsections:

1. **Checking your hearing.** This section provides several hearing screening tests and advice for follow up. The screening tests comprise of:

    - Digit triplet test. Test is available in 7 languages.
    - Localization test based on detecting the Minimum Audible Angle (MAA) of 2 sound sources. The test is available in 5 languages.
    - Localization test based on detecting the location where a sound is produced. For this a virtual acoustical environment is made that simulates a living room. The simulation also includes sound reflections such to evaluate localization performance as in real live. The test is available in 4 languages.
    - HearCom Questionnaire of 18 questions. This is a questionnaire based on existing clinical questionnaires optimized for use as a screening test. The questionnaire is available in 3 languages.

For each of the above screening the result is presented in an advice that distinguishes in 3 categories (good, moderate, and not good). The 3rd category advises to contact a hearing professional who should make additional tests to confirm the hearing status.

2. **Information on Hearing.** This section provides information on hearing structured in the following themes:

    - How we hear.
    - What is Hearing loss
    - Types of hearing loss.
    - Preventing hearing loss.
    - Communication tips.
    - Tips for family members.

Each theme is explained by simple, but comprehensive background information.

3. **HearCompanion.** This section is a new user service for support on hearing rehabilitation. The service is to be used for people:

   - concerned on their hearing
   - just being a new user of hearing aids
   - using hearing aids for longer term but likes information for their optimal use.

The visitor of the HearCompanion is guided by means of personal questions along various aspects of hearing, communication and hearing aid usage. More information is found in Vlaming et al (2010).

Professional section

The professional section is intended for professionals who work in the area of hearing and communication. These can be professionals who assist persons having hearing and communication problems, but also professionals that work with requirements for optimal hearing and communication in various situations. Examples are: audiologists, ENT-doctors, family doctors (GP), speech and hearing therapists, architects, acousticians, engineers, etc.

The professional section contains information on subjects like:

- Background rationales of the hearing screening procedures as offered in the user section
- Backgrounds on the tests of the Auditory Profile
- Demonstration downloads for tests of the Auditory Profile
- Rehabilitation Procedures: fitting and optimalisation of hearing aids and cochlear implants
- Algorithm Development: details of several popular and novel signal enhancement algorithms with demonstrations.
- Information on designing website according to the principles of Inclusive Website Design
- Information on Assistive Technologies as researched by HearCom
- Information on Ambient Acoustics with demonstrations of the effect of adverse acoustics on speech understanding

Website design

The HearCom Portal is one of the key elements of the HearCom project, because through it, all the results of the project are made available to the different target audiences. In the development of the Hearing portal the guidelines have been followed of eEurope: Quality Criteria for Health related Websites which can be found at http://ec.europa.eu/information_society/eeurope/ehealth/quality/draft_guidelines/index_en.htm.

The main technical frameworks used by the HearCom portal are:

- An Apache 2.0 Web Server, that proxies and caches all requests to increase efficiency of the system.
- Several Apache Tomcat servlet engines (version 5.5.x) that host the Content Management System.
- Mysql RDBMS[1] database able to support BLOB[2] objects.
- Apache Axis 2.x as the HearCom Web Service integration framework.

The web pages in the entire portal were designed to be valid according to the W3C (X) HTML specification. The Contents Management System (CMS) has been adapted to the requirements of the HearCom project and partners. The content has been fully separated from its presentation as recommended by the W3C Web Content Accessibility Guidelines (WCAG 1.0).

Iterative accessibility evaluations were conducted during the entire lifecycle of the project including automatic and expert checks. Hearing impaired users were involved in the accessibility tests of the website. A user study of 32 test participants (normal hearing and hard of hearing) has been conducted to evaluate the usability issues.

[1] Relational Database Management System
[2] Binary Large Object

HearCom Partners

- VU University Medical Center, ENT Communication, Amsterdam, NL
- Cochlear Technology Center Europe, Mechelen, BE
- University of Leuven, Lab.Exp.ORL, Leuven, BE
- University Hospital Zürich, Lab of Experimental Audiology, Zurich, CH
- Fraunhofer Institute for Applied Information Technology, Sankt Augustin, DE
- Kompetenzzentrum HörTech, Oldenburg, DE
- Hörzentrum Oldenburg, Oldenburg, DE
- Kuratorium OFFIS e.V., Oldenburg, DE
- Ruhr-University Bochum, Institute of Communication Acoustics, Bochum, DE
- Siemens Audiologische Technik, Erlangen, DE
- Universität Erlangen, Erlangen, DE
- Universität Oldenburg, Oldenburg, DE
- Technical University of Denmark Ørsted-DTU, Lyngby, DK
- GN Resound A/S, Copenhagen, DK
- EFAS, European Federation of Audiology Societies
- Moviquity, Madrid, ES
- THALES Communications, Paris, FR
- Institute for Language and Speech Processing, Athens, GR
- Academic Medical Center University Amsterdam, Experimental Audiology, Amsterdam, NL
- Erasmus Medical Center, Experimental Audiology, Rotterdam, NL
- TNO Netherlands Organisation for Applied Scientific Research, Soesterberg, NL
- Kungl Tekniska Hogskolan, Stockholm, SE
- University Hospital Linkoping, Linkoping, SE
- University of Southampton, Southampton, UK
- RNID (The Royal National Institute for Deaf People), London, UK
- University College London, London, UK
- Dresden Technical University, Institute of Acoustics and Speech, Dresden, DE
- NXP Semiconductors (before Philips Semiconductors), Leuven, BE.

HearCom Public Reports

The HearCom project has produced more than 110 reports. Part of these reports have been made public available at the HearCom website. See http://hearcom.eu/about/DisseminationandExploitation

D-1-2: Report on the proposed set of communication performance tests

D-1-3: Protocol for implementation of communication tests in different languages

D-1-4a: First version on Internet screening tests in three languages

D-1-4b: First version of Internet screening tests in three languages (Final report and demonstrator)

D-1-5: First version of Internet screening tests for localization discrimination (Demonstrator)

D-1-6b: Development of a digit-triplet test in modern Greek

D-1-7: Report on normalization data and crosslanguage comparison for sentence tests

D-1-9: Report on an optimized inventory of Speech-based auditory screening & impairment tests for six languages

D-1-11: Telephone implementation of Polish Digit Triplet Test

D-2-1: Implementation of a preliminary test set for auditory impairments

D-2-1b: Demo version of the preliminary test set for auditory impairments in German

D-2-2: Procedures for the tests included in the auditory profile in four languages

D-2-4: Proposal for dissemination of the Auditory Profile via reference implementation and via Internet

D-2-5: Optimized, final set of impairment tests included in the auditory profile

D-2-6: Report about the results of the multicentre evaluation of the Auditory Profile

D-2-7: Internet demonstrator of auditory profile tests

D-2-8: Evaluation of the Auditory Profile in relation to auditory rehabilitation

D-3-6: First steps towards normative criteria for non-normal hearing listeners in adverse acoustic conditions

D-3-7: Improved Internet-based AVE for Demonstration and Self-Screening Purposes

D-3-10: Adverse condition demos and model predictions on the HearCom portal

D-4-3: Report on Experiments on the Performance of Normal and Non-normal-hearing Listeners for a Range of (Simulated) Transmission Conditions with Combined Technical Disturbances

D-4-5: Conclusions on the impact of hearing impairments on the quality measures w.r.t. telecom degradations

D-5-1: A sub-set of signal enhancement techniques operational on a PC-based system

D-6-1: Report on the analysis and evaluation of current fitting procedures used throughout Europe

D-6-2: Report on the feasibility of proposed fitting and rehabilitation pathway

D-6-3: Simulation studies of bimodal CI and hearing aid fittings

D-6-4: Report on outcomes of research on automated fitting for compression hearing aids

D-7-1: Speech recognition tests for different languages

D-7-1b: Speech recognition tests for two additional languages: Polish and French

D-7-2: Specification on 'standard speech-in-noise test' and 'standard environmental conditions' for evaluation of aided performance

D-7-3: Specification spatial hearing test

D-8-1: Requirements specification of personal link and related application services

D-9-1: Requirement specification of user needs' for assistive applications on a common platform

D-9-5: Definition of PCS platform and assistive communication services

D-9-11: Wireless Public Announcement Systems White Paper

D-9-12: The Personal Hearing System – a Software Hearing Aid for a Personal Communication System

D-10-1: Specification of profiles for the internet sound system and methods to control the internet sound system

D-10-2: User requirement specifications for hearing and communication Internet services

D-11-5: Development of a screening questionnaire

D-11-6: Validation of the screening localisation test and an inter comparison of all HearCom screening tests

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