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Pre-Amble

The HearCom SP4/WP9 contains four planned deliverables on the PHS, of which this is the last. The first deliverable, D-9-2 (2005) “Software development platform Master Hearing Aid (MHA) with a preliminary implementation of one preprocessing method” is a description of the Master Hearing Aid software with some modifications, written to make an implementation of algorithms in WP5 possible on this platform. The second deliverable, D-9-3 (2006) “Specification of a Personal Hearing System (PHS) and its Interfaces” specifies the PHS, both in a personal communication system environment and as a research platform. The third deliverable, D-9-4 (2007) “Report on the implementation of a Personal Hearing System (PHS) for use in WP7” is a description of the PHS for use in WP7 evaluations and of the PHS research platform. This last deliverable D-9-12 describes the PHS as a prototype for a hearing system in a personal communication system, and a portable field testing device.

1 Executive Summary

Within the HearCom project, the Personal Hearing System (PHS) based on the HoerTech Master Hearing Aid software has been specified and implemented. Developments include special interfaces for insertion of telecommunication audio streams, user interfaces for switching programs and volume, and professional interfaces for fitting and calibration. A dedicated audio interface has been implemented for field testing with hearing aid shells.

The implemented PHS is

- a laboratory hearing aid algorithm test platform for evaluation of WP5 algorithms within WP7.
- a portable field test research hearing system.
- a prototype for the PHS functionality within the Personal communication system.

In this deliverable, the focus is on the portable field test device and the prototype implementation.

This deliverable has been submitted to EURASIP Journal on Advances in Signal Processing (JASP), special issue devoted to ”Digital Signal Processing for Hearing Instruments” for publication.

Abstract

A novel hearing system software is described which may be integrated into consumer communication devices for supporting audio communication in difficult listening conditions, in particular for the elderly and for hearing-impaired persons. An analysis of possible hardware and software architectures shows that such a personal hearing system (PHS) can be implemented on future advanced cell phone-like personal communication systems (PCS) using standard hardware and software architectures provided that the floating-point performance of such devices is increased and wireless body area networks (WBAN) are improved in terms of data rate and signal delay. Specific possible applications are assessed using a prototype PCS implemented on a small notebook computer with a dedicated audio interface. It is shown that the prototype PCS can integrate acoustic communication, telephony, public announcement systems and a PHS for hearing support. Specific PHS algorithms are shown to improve hearing in difficult listening conditions. An exemplary binaural coherence-based speech enhancement scheme that represents a large class of possible processing schemes is shown to be compatible with the concept of a central processor conected to the hearing aids with a BAN and to require BAN data rates far below the audio data rate with relaxed delay constraints.

2 Introduction

Hearing disabilities are a major problem in modern societies: Whereas a large part of the society – 28.8% of US population – suffers from hearing problems in everyday life, less than 25% of those hearing impaired people use a hearing system to reduce the effects of hearing impairment [11]. Two major reasons for not using a hearing system are (i) stigmatization, and (ii) insufficient performance of common hearing systems in respect to improvement of communication abilities, in particular in adverse listening conditions with high levels of background noise and reverberation [12]. At the same time, most of the inhabitants of industrial nations use modern communication devices, mainly mobile phones, but also portable MP3 players and other multimedia devices: The mobile phone penetration in Europe is above 100% in most countries, and in some countries exceeds even 130% [19]. General motivation for this study is to investigate the technological perspectives of improving hearing support and its acceptance by integrating communication services and hearing support systems using scalable hard- and software.

Hearing aids are the standard solution to provide hearing support for hearing-impaired persons. In many adverse conditions, however, current hearing aids are insufficient to alleviate the limitations in personal communication and social activities of the hearing-impaired. Most challenging problems are howling due to acoustic feedback from the hearing aid receiver to the microphones [8], interference of cell phone radio frequency components with hearing aids [17], and low signal-to-noise ratios (SNR) in public locations caused by competing noise sources or reverberation [9]. A number of partial solutions addressing these problems are available in current hearings aids. Signal processing solutions comprise noise reduction algorithms like spectral subtraction and directional microphones [9]. Other assistive solutions comprise direct signal transmission by telecoils, infrared and radio systems [2, 13]. Recent technological progress opens up possibilities of improving these solutions. New bridging systems, currently intended mainly for connection to communication and home entertainment devices, are based on the digital BlueTooth protocol, e.g., the ELI system [20]. New scalable algorithms can be adopted to different listening situations and communication environments and are expected to be beneficial for the end user either in terms of improved speech intelligibility or by enhancing speech quality and reducing listening effort [3]. Adding a central processor based on standard hardware to the hearing aids gives the advantage of advanced processing performance and access to binaural audio information [1]. A combination of the new signal processing schemes and communication options has not been explored yet, and the final user benefit remains to be investigated. Dedicated prototype systems as investigated in this study might facilitate this type of research.

Contrary to a hearing-impaired with moderate or strong hearing loss, a person with mild hearing loss, or more generally any person with light to moderate problems in hearing under adverse circumstances, will not wear a hearing aid nor other hearing support system. Hearing support systems which are add-ons to existing communication devices might be beneficial for those users and their acceptance is expected to be higher than that of conventional hearing aids. The SM100 by SoundID [6] could be regarded as such a 'beginner' hearing aid. This device is a BlueTooth headset with personalized signal processing. Beside of processing incoming digital audio streams, it can process environmental sounds picked up by a directional microphone on the device. The pricing of this product is in the upper range of BlueTooth headsets, far below conventional hearing aids. The advertisements of the SM100 seem to avoid the term 'hearing aid', which would also imply legal issues as for medical devices. Although first attempts have been made to bring new assistive listening devices for persons with mild hearing losses on the market (e.g. SM100 by SoundID; A200, SET810S, SET820S by Sennheiser), the perspectives for such hearing systems appear to be unclear. Further market-related research into possible applications and new services therefore seems indicated. General constraints of this path towards new products are discussed in this study.

Another factor that influences and might facilitate the further development of hearing support systems is the availability of standard hard- and software for mobile devices. The possibility of using high level programming languages both in algorithm development and for the final application allows scaling of algorithms with the continuously improving performance of new hardware. Based on such scalable systems, the integration of hearing support systems for slight-to-moderate hearing losses with communication applications seems to be feasible in principle, but is yet to be assessed in more detail. One step towards that direction are low-delay realtime signal processing systems based on standard hard- and software, such as the Master Hearing Aid (MHA) [5], a development framework for hearing aid algorithms. Another hardware-related factor is the development of Wireless Body Area Networks (WBAN), that can be seen as an enabling technology for mobile health care [14] and that could mediate the communication between a central processor and audio headsets attached to the ear like hearing aids.

In summary, recent developments open up the possibility of merging the functionality of traditional hearing aids and other hearing support systems for slight-to-moderate hearing losses on scalable standard hardware. This combination will be defined as a Personal Hearing System (PHS). Furthermore, the integration of this PHS with general and new communication applications of mobile phones and PDA's to define a Personal Communication System (PCS) may lead to new applications and improved hearing support. User inquiries regarding the acceptance of such a PCS have been carried out within the EU project HearCom [10] and its general acceptance was demonstrated provided that the device is not larger than a mobile phone and includes its functionality. Some specific solutions to this already exist, but audio applications with scalable listening support for different types of hearing losses and having a connection to personal communication and multimedia devices are not yet available. The aim of this study is therefore to establish a basis for further research and development along these lines. In section 3, a possible architecture of a PCS is outlined and possible applications are discussed in section 4. In particular, it is discussed to what extend an architecture with a central processor can be compatible with current advanced hearing aid algorithms and what are the resulting constraints on the wireless link between central processor and audio headsets. Section 5 describes the implementation of a prototype PCS which runs on a netbook computer and hosts four representative signal enhancement algorithms. A first evaluation of hardware requirements (e.g., processing power, wireless link requirements), software requirements (scalable signal processing) and of the expected benefit for the end users is performed using this PCS prototype.

3 Architecture

The PCS is a handheld concentrator of information to facilitate personal communication. Fig. 1 shows a block diagram of the projected PCS and its applications. The PCS is assumed to be a development based on new advanced mobile telephones and Personal Digital Assistants (PDA's). The reason for selecting a mobile phone as a PCS platform is the availability of audio and data networking channels, like GSM, UMTS, BlueTooth, and WiFi. The processing capabilities are continuously growing, as the offered services are growing – many current mobile phones offer Internet applications, word processors, and multimedia applications. Many devices also include a global positioning system, which can be utilized by public announcement services.

Audio is played to the user via a pair of audio headsets. These audio headsets are housing loudspeakers/receivers for audio playback. Each audio headset also has two or three microphones, which can be configured to form a directional microphone for picking up environmental sounds, and the own voice of the user for phone application. As an option, the audio headsets provide audio processing capabilities similar to hearing aids.

A short range wireless link (Wireless Body Area Network, WBAN) provides the connection between the PCS and the audio headsets, and optionally between the two audio headsets at the left and right ear. Mid range and wide range links are used to establish connections to telecommunication network providers and to local information services. All links are part of the wireless Personal Communication Link (PCL), which supplies information to the PCS and between the PCS and the audio headsets, as a successor for the inductive link (telecoil) of current hearing aids. The PCL might convey text- or audio-based information including alarm signals.

A key application on the PCS is the PHS: The audio communication channels of the PCS, e.g. telephony, public announcement and home entertainment, are processed in the PHS with personalized signal enhancement schemes and played back through the audio headsets. In addition to the PCS audio communication channels the PHS can process environmental sounds picked up by the headset microphones near the user's ears. Processing methods may differ depending on the input, i.e., acoustic input or input through the PCS communication channels. The functionality of the PHS covers that of a conventional hearing aid, and adds some additional features: (i) Increased connectivity: The PCS provides services, which can connect external sources with the PHS. (ii) Advanced audio signal processing schemes: The performance and battery size of the central processing device allows for algorithms which otherwise would not run on conventional hearing aids. (iii) Potential of production cost reduction: Usage of standard hardware may reduce

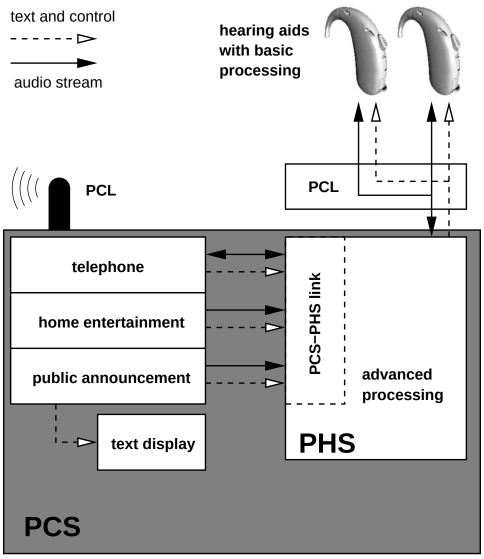

Figure 1: Architecture of a Personal Communication System (PCS) hosting the personal hearing system. The PCS (large shaded box) is hosted on an advanced mobile phone. The Personal Hearing System (PHS) is a software component for signal enhancement, processing audio output of the PCS communication channels and processing environmental signals. The Personal Communication Link (PCL) transfers environmental sounds to the PHS and the processed sounds or control information back to the audio headsets.

production, marketing, distribution and service costs, if consumer headsets with slight modifications, e.g., addition of microphones for processing of environmental sounds, can be used.

For processing the PCS audio communication channels, an uni-directional link from the central processor to the headsets is sufficient and the link delay is not critical. Processing environmental sounds in the central processor, however, requires a bidirectional link which needs further consideration. In general, all processing blocks can be run either on the audio headsets or on the central processor. The decision on which block runs where depends on several issues: (i) The computational performance and battery capacity of the audio headsets is typically low and does not allow complex algorithms. (ii) The central processor or the PCL might not be available continuously because of wireless link breakdowns. Therefore, at least basic processing like amplification for hearing loss correction is required to run on the audio headsets. (iii) Depending on the properties of the PCL, the delay might exceed the tolerable delay for processing of environmental sounds [16], and will constraint the algorithms on the central processor. Link delays smaller than 10 ms would allow routing the signal through the central processor. In typical hearing aid applications, signal enhancement schemes precede the processing blocks for hearing loss correction (e.g., amplification and compression). To avoid the transmission of several signal streams, only one set of successive processing blocks can be run on the central processor. As to whether emerging WBAN technology might be powerful enough to achieve the delay limit seems unclear yet. If the total link delay is longer than about 10 ms, the signal path needs to remain completely on the audio headsets. Then, processing on the central processor is restricted to signal analysis schemes that control processing parameters of the signal path, e.g., classification of the acoustical environment, direction of arrival estimation and parameter extraction for blind source separation. In general it seems feasible that these complex signal analysis schemes and upcoming complex processing performance demanding algorithms for Auditory Scene Analysis[18] might not necessarily be part of the signal path. The projected architecture might therefore be suited for these algorithms, which could benefit from the high signal processing and battery power of the central processor. Other requirements for the link are bandwidth and low power consumption: To allow for multi-channel audio processing, several (typically two or three) microphone signals from each ear are required, asking for sufficient link bandwidth. Additionally, if signals are transmitted in compressed form, the link signal encoder should not modify the signal phase to avoid artifacts and performance decreases in multi-channel processing. To ensure long battery life, the link should use low power. To reduce the link power consumption, the PHS could provide only advanced processing on demand. Switching on advanced processing and the link might be either controlled manually or by an automatic audio analysis in the headsets.

4 Applications

This section describes possible audio applications of the PCS. Four audio communication channels are introduced: Acoustic communication and sensing, telephony, public announcements and entertainment. Solutions for each of these communication channels will be specified.

4.1 Acoustic Communication

The possibility of processing environmental sounds is of importance for all users who are wearing the headsets not only when using the mobile phone. The processing of environmental sounds gives hard constraints regarding the total delay between incoming microphone signal and receiver output as outlined above.

The architecture of the PHS with a central processor gives the ability to process binaural information in the central processor and unilateral information either in the central processor or in the audio headsets. Considering typical processing schemes in hearing aids, unilateral processing comprises dynamic compression, single channel noise reduction and feedback cancellation. Typical applications of the central processor are binaural and multi-microphone methods, e.g., binaural ambient noise reduction, beamformer and blind source separation [9]. If the link delay is not sufficiently small to route the signal path through the central processor, binaural processing can still be achieved assuming a signal analysis algorithm running on the central processor processes signals from left and right side and controls the signal path on both sides.

4.2 Telephony

Telephony is probably the most important service of a modern communication device. The phone device is connected to the telephone network provider via a wireless link (e.g., GSM, 3G) or a cable connection.

In current devices, support for the hearing impaired is given via a magnetic connection to hearing aids (telecoil), and by personal amplification of the receiver signal and ring tones. Some devices even provide frequency shaping, although the lack of common fitting procedures make this feature barely accessible. Some devices – mainly land-line devices – give support for the visually impaired by reducing the number and increasing the size of knobs.

In the PCS, the voice from the remote partner is transmitted to the PHS for personalized amplification and pre-processing. The voice of the local talker is picked up either by the audio headset microphone or by a microphone in the PCS, and transmitted to the partner.

4.3 Public Announcement

Public announcement services are used in public transport (e.g., train stations, underground, bus stations, and airports), as well as in education and entertainment (e.g., schools, museums, theater, churches, concerts, sport events).

Current technologies include amplification of the sound via loudspeakers, transmission to hearing aids or headphones via telecoil, infrared or FM radio, or personal audio guides. Visual presentation of information is also possible (e.g., theaters, cinema). New methods also include SMS broadcast.

In the PCS, public announcements are received via the PCL. Visual information can be displayed on the PCS display, and sounds are transmitted to the PHS for personalized processing. Ambient intelligence systems that provide content selected and formatted to the users needs, e.g., selection of only relevant announcements in airports, automatic presentation of announcements in the users native language, or presentation of information based on the users position become feasible with the PCS architecture.

4.4 Home entertainment

Home entertainment, e.g., TV, radio, CD players, MP3 players or Internet media streamers, provide an output signal which is optimal for the normal hearing listener. The use of headphones is usually not suitable for hearing aid users, because they can not be worn together with hearing aid devices. Using headphones without hearing aids would lack the personalized frequency shaping, amplification and dynamic compression. Assistive listening devices can be connected via the analog output (headphone output) with an adaptor to telecoil or BlueTooth, or in more recent devices directly via BlueTooth. Using assistive listening devices can alleviate the effects of weak transmission systems (e.g., loudspeakers), room acoustics and background noises. Research on the integration of assistive technologies into common home entertainment devices is currently done in the EU project Hearing at Home [15].

5 Implementation of a Prototype System

To assess the PCS architecture and applications experimentally, a prototype PCS has been implemented on a small notebook computer ('netbook') in combination with a SmartPhone. To demonstrate PHS applications, several signal enhancement algorithms have been realized on the PCS prototype using the MHA algorithm development environment, see section 5.2.1. A dedicated audio interface used to connect audio headsets to the netbook is outlined in section 5.2.2. A phone service as a prototype application of the PCS, implemented on a smartphone, is described in section 5.3.2. One signal enhancement algorithm (coherence-based de-reverberation [3]) was taken as an example and has been tested on its conformity with the concept of the PHS.

5.1 Architecture

See Fig. 2 for a schematic signal flow of the prototype system. The PHS is implemented using a separate notebook computer. Notebook computers deliver sufficient performance for outstanding signal processing. Selecting signal processing algorithms carefully and using performance optimization techniques, allows for stripping down the PC platform. However, a floating point processor is required for prototype algorithms and prevents from using fixed-point processor based PDAs or Smartphones. Using floating point algorithms enables fast prototyping and very early field testing. In a later step it is possible to recode positively evaluated algorithms to a fixed-point representation and install these on PDAs or Smartphones. PCS services are implemented on a SmartPhone with networking capabilities. The PCS-PHS link is realized as a WiFi network connection. The audio headsets are hearing aid shells with microphones and receiver, without signal processing capabilities. The audio headsets are connected to the PHS via cables and a dedicated audio interface. The audio headset signal processing capabilities are simulated on the central processor. Fig. 3 shows a netbook based PHS prototype.

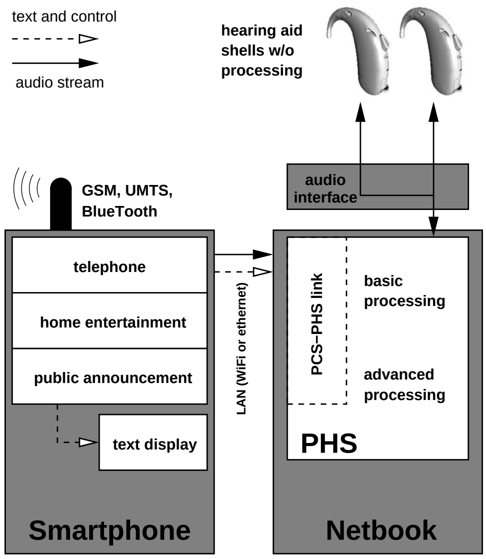

Figure 2: Prototype implementation of the PCS. The PCS services are hosted in a Smartphone, the PHS (mainly signal processing) is hosted in a portable PC. The PC connects to the hearing aid shells via a dedicated audio interface. architecture.

5.2 Hardware components

In the following sections the hardware components of the prototype implementation are described.

5.2.1 Netbook: Asus Eee PC

For the prototype system, a miniature notebook has been used as a hardware accelerated floating point processor for the PHS: The Asus Eee PC is a small and lightweight notebook PC, it's size is about 15*22 cm, weighting 990 grams. It provides an Intel Celeron processor M, running at a clock rate of 630 MHz. To gain low delay signal processing in a standard operating system environment, a Linux operating system (UbuntuStudio 8.04) with a low-delay real-time patched kernel (2.6.24-22-rt) has been installed. For comparison, the system was also installed on an Acer Aspire netbook PC and a standard desktop PC.

Figure 3: PHS prototype based on the Asus Eee PC, with a dimension of 22×15 cm and a weight of 1.2 kg, including the sound card and audio headsets.

5.2.2 Dedicated Audio Interface

A detailed market survey showed that commercially available audio interfaces cannot satisfy all requirements for the mobile PHS prototype. High quality devices as used in recording studios offer the required signal quality and low latency but are not portable because of size, weight and external power supply. Portable consumer products do not offer quality, low latency and required number of capture and playback channels. Therefore a dedicated USB audio interface has been developed which fulfills the requirements of the PHS prototype. The audio interface has been developed in two variants: A device with four inputs and two outputs to drive two audio headsets with two microphones in each headset ('USBSC4/2'), and a device with six inputs and two outputs, for two audio headsets with three microphones each ('USBSC6/2'). The basis for both devices is a printed circuit board (PCB). The USBSC4/2 contains one PCB as shown as PCB1 in Fig. 4. Assembled are two stereo ADs (four channels) and one stereo DA (two channels). A micro controller (μC) implements the USB2.0 interface to PC hardware. A complex programmable logic device (CPLD) serves as glue-logic between μC and AD/DA. The advantage of CPLDs is the possibility to reconfigure their interconnecting structure in system. This feature is used to connect two PCBs and build one device with more channels (USBSC6/2). A simple reconfiguration of the CPLDs and a software exchange on the μC (exchange of firmware) enables the configuration of other devices in shortest time, as for example a device with four inputs and outputs, or eight inputs and no outputs. The hardware is generally applicable. With minor modifications it can be used as a mobile recording device or as consumer sound card for multimedia PCs.

For future usage of the developed hardware, device variations in number and type of channels depending on user requirements are quickly retrievable. The architecture is extendable by a hardware signal processing unit for user defined audio pre-processing by exchanging the CPLD with more complex components like field programmable logic devices (FPGA). This could unload the host PC platform for maximum performance. This again minimizes the requirements for computing power and enables the assembly of small, portable embedded PC boxes. It has to be stated though, that the implementation of algorithms in FPGAs using a fix point hardware description language (HDL) like VHDL

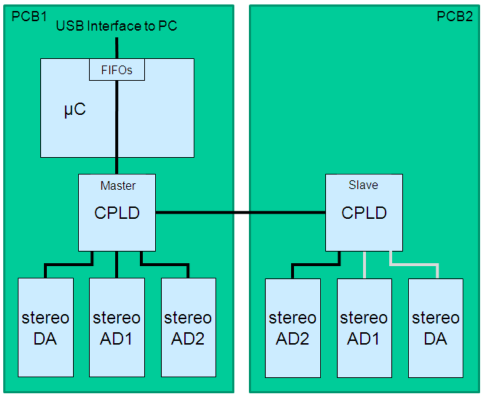

Figure 4: Architecture of the dedicated audio interface, with four inputs and two outputs (PCB1 only), or six inputs and two outputs (PCB1 and PCB2).

or Verilog is even more elaborate than transferring floating-point SW to fix-point. Thus, this proceeding is only adequate for well evaluated and often used algorithms like for example the FFT due to high non-recurring engineering costs.

The developed audio interface is a generic USB2.0 audio device that does not require dedicated software drivers for PCs/Notebooks running under the Linux operating system. The device utilizes the USB2.0 isochronous data transfer connection for low latency, and therefore does not work with USB1.

The audio interface is equipped with connectors to directly connect two hearing aid shells housing a receiver and up to three microphones. The device provides a microphone power supply. The USB audio interface is powered via the USB connection. RC-filters and ferrite beads are used to suppress noise introduced by the USB power supply. One RC-filter is placed directly at the supply input. Furthermore, at each AD- and DA-converter one filter is placed close to the analog and the digital supply respectively. Additionally, noise is suppressed by the use of ferrite beads in each supply line of each converter.

5.3 Software Components

In the next sections, the major software components used in the PCS prototype implementation are described.

5.3.1 PHS Algorithms

In the PHS prototype four representative signal enhancement schemes have been implemented: Single-channel noise suppression based on perceptually optimized spectral subtraction (SC1), Wiener-filter-based single-channel noise suppression (SC2), spatially preprocessed speech-distortion-weighted multi-channel Wiener filtering (MWF) and binaural coherence de-reverberation filter (COH) [3]. Individually fitted dynamic compression and frequency-dependent amplification was placed after the signal enhancement algorithm, to compensate for the user's hearing loss. Hearing loss compensation without any specific signal enhancement algorithm is labelled REF.

The prototype algorithms are processed at a sampling rate of 16 kHz in blocks of 32 samples, i.e., 2 ms. Audio samples are processed with 32 Bit floating point values, i.e., four bytes per sample.

As an example, we look at the coherence based de-reverberation filter in more detail:

At both ears the algorithm splits the microphone signal into nine frequency bands. In each frequency band, the average phase φ across FFT bins belonging to the frequency band and across time (block length) is calculated. Comparing the phase with the phase of the contra-lateral side results in the inter-aural phase difference (IPD) within a frequency band. The phase difference $\varphi_l - \varphi_r$ is represented as a complex number on the unit circle, $z = e^{\varphi_l - \varphi_r}$. The estimated coherence is the gliding vector strength c of z, $c = |\langle z \rangle_\tau|$, with the averaging time constant τ. The estimated coherence is directly transformed into a gain by applying an exponent α, $G = c^\alpha$. A detailed description of the algorithm and its relation to a cross-correlation based de-reverberation algorithm can be found in [3].

5.3.2 Phone Service and PCS-PHS Link

To provide ways to connect PCS audio communication streams to the PHS, a specific network audio interface has been implemented in the PHS. Together with a sender application this interface forms the PCS-PHS link, which is using the Internet Protocol Suite (TCP/IP). The link can be established on demand, and contains a protocol to select appropriate mixing strategies for different signal sources: The level of the source signal can be matched with the environmental sound level, and environmental sounds can be suppressed for better speech intelligibility or alarm signal recognition. The mixing configuration is followed by the Pulse-code modulation (PCM) encoded audio stream. Whenever a phone connection is established, the sender application in the PCS is connecting to the PCS-PHS link and is recoding the phone's receiver output for transmission to the PHS. To avoid drop-outs in the audio stream, the signal from the phone has to be buffered, introducing a delay between input and output. To reduce the delay caused by the WiFi connection the packet size was reduced to a minimum. The total delay varies between 360 and 500 ms. The long delay is specific to the prototype implemenation; the final application will not include the WiFi link between PCS phone service and PHS, since both services are then hosted on the same machine.

5.4 Calibration Methods

To ensure that the prescribed gain in hearing aids is applied to the microphone signal, a hearing aid has to be calibrated. For level-dependent algorithms, such as dynamic compression, it is of major importance that the sound level of external sounds like speech or noise is estimated correctly.

The calibration method in the PHS consists of two stages: (i) The microphone signal is scaled to represent equivalent free field levels. The microphones are assumed to have a flat free-field response in a reasonably large frequency range. (ii) At the output, a frequency dependent gain corrects for the receiver response and the modified ear canal amplification with inserted ear moulds. The average normal hearing threshold for free field representation is defined in ISO 226 [7]. An ideal system (and any audiometer) must be able to reproduce this threshold. Thus, a system which can reproduce the normal hearing threshold can be assumed to have a flat system response. This defines a way to system correction: The hearing threshold of a large number of normal hearing subjects is recorded with the hearing aid receiver without equalization. The difference between the recorded average normal hearing threshold and the ISO normal hearing threshold defines the frequency dependent gain to be applied to the receiver. To allow a fast broadband re-calibration, the frequency dependent gain is normalized at 1 kHz and a broadband gain

	Asus Eee PC 4G Celeron M 630 MHz		Acer Aspire one Atom N270, 1.6 GHz		Desktop PC Pentium 4, 3GHz
Algorithm	% CPU	batt.	% CPU	batt.	% CPU
SC1	23.3%	3h14’	20.5%	2h28’	8.5%
SC2	12.0%	3h19’	11.0%	2h27’	4.0%
MWF	16.7%	3h16’	17.0%	2h09’	4.5%
COH	3.5%	3h22’	4.5%	2h27’	1.0%
MHA	33.1%		30.5%		12.0%
jackd	6.0%		4.8%		2.2%
IRQ	4.5%		4.2%		1.0%

Table 1: Performance of the PHS prototype system. In addition to the algorithm CPU time, the CPU time used for signal routing, resampling, overlap-add, spectral analysis and hearing loss correction was measured (labelled MHA), and also the CPU time used by the jackd sound server and the sound card interrupt handler.

is added. A correction for the binaural advantage of 2 dB has been taken into account in the broadband calibration value, since each ear is measured individually.

5.5 Evaluation Results

5.5.1 Computational complexity and power consumption

The computational complexity of the PHS prototype system is estimated by measuring the CPU time needed to process one block of audio data, divided by the duration of one block. For real-time systems, this relative CPU time needs to be below one. For most operating systems, the maximum relative CPU time depends also on the maximum system latency and the absolute block length. A detailed discussion of relative CPU time and real-time performance can be found in [5]. The relative CPU time of the PHS running the five respective signal enhancement algorithms is shown in Table 1.

For a portable PHS, a long battery runtime is desirable. The battery runtime of the PHS prototype has been measured by continuously measuring the battery voltage while running the PHS with the respective signal enhancement algorithms. The battery was fully charged before each measurement. The time until automatic power-down is given in Table 1. During the test, the netbook lid was closed and the display illumination was turned off.

5.5.2 Benefit for the End User

The algorithm performance in terms of speech recognition thresholds (SRT) and preference rating has been assessed in the HearCom project in a large multi-center study [3]. As an example, speech recognition threshold improvement data from [3] is given in Table 2. The standard deviation of the results across four different test sites is marginal, which proves the reliability of the PHS prototype as a research and field testing hearing system. While the speech intelligibility could not be improved by the algorithm COH, it was preferred by most subjects against identity processing, see Fig. 5. Hearing impaired subjects show a clearer preference for COH than normal hearing subjects do. The listening effort can be reduced by COH if the SNR is near 0 dB. Even if the SRT can not be improved by the algorithm, the reduction of listening effort is a major benefit for the user.

	SRT improvement / dB			
Test site	SC1	SC2	MWF	COH
1	0.0	-0.2	7.1	0.2
2	0.0	-0.1	6.7	-0.2
3	-0.1	-0.1	6.2	-0.5
4	0.2	-0.3	6.7	0.1
average	0.1	-0.1	6.7	-0.1
std. dev.	0.14	0.09	0.36	0.32

Table 2: Speech Recognition Threshold (SRT) improvement (i.e., difference to identity processing with hearing loss correction 'REF') in dB SNR for the five algorithms, measured at four test sites. The standard deviation across test site is marginal, which proves the reliability of the PHS prototype as a research and field testing hearing system. Data from [3].

Furthermore, a combination with the MWF algorithm is possible and indicated, since both methods exploit different signal properties (directional versus coherence properties). An improvement of the beamformer performance is likely if the coherence filter is preceding the beamformer [4].

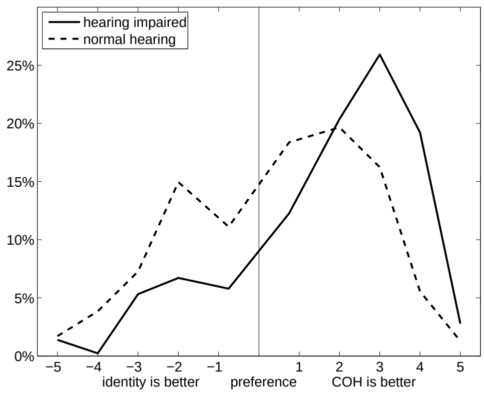

Figure 5: Preference histogram. COH is preferred against identity by 80.6% of the hearing impaired and by 61.1% of the normal hearing subjects. The categories 1-5 are "very slightly better", "slightly better", "better", "much better" and "very much better".

5.5.3 Requirements towards the PCL

The requirements towards the wireless link between headsets and central processor varies with the algorithms. The link bandwidth required to transmit all six microphone channels is 384 bytes per block in the direction from one headset to the central processor and 128 bytes per block in the other direction. With two headsets and 500 blocks per second this leads to a required (uncompressed) bandwidth of 375 kBit/s from headsets to the central processor and 125 kBit/s back to the headsets. The requirements for the coherence filter COH towards the link bandwidth in three scenarios are presented. The trivial scenario is the condition where the algorithm is running on the central processor, where the full audio signal of both sides is required, i.e., 125 kBit/s in each direction. Lower bandwidth is required if only signal analysis is performed in the central processor: The phase information in nine frequency channels of each side and for each cycle is required, leading to a headset-to-processor bandwidth requirement of 35.2 kBit/s. For the other direction nine gains are transmitted, with identical gains for both sides. This results in a bandwidth requirement of 17.6 kBit/s. The third scenario is a situation where signal analysis and filtering are processed in the audio headsets, and the link is only used for data exchange between the audio headsets. Then only the phase information is exchanged, i.e. 17.6 kBit/s are required in each direction. These bandwidth requirements do not include data compression. With special signal coding strategies, the bandwidth requirements can be further reduced.

For signal routing from the headsets via the central processor back to the headsets, a maximum delay of approximately 10 ms is acceptable. For larger delays, a signal analysis on the central processor is possible. However, when the sequence of filter coefficients is delayed relative to the signal to which it is applied, signal distortion will arise. Informal listening tests revealed that a link delay of 25 ms is acceptable for the COH algorithm. Because this algorithms represents the class of speech envelope filters, this margin might apply for the more general case, too.

5.5.4 Technical performance of dedicated audio interface

The technical performance of the dedicated audio interface is given in Table 3. During the evaluation of the dedicated audio interface, the following factors on the audio quality of the device have been found: (i) A notebook should be disconnected from power supply and used in battery mode, to avoid a 50 Hz distortion caused by the net power supply. (ii) Other USB devices should not be connected to the same USB controller/hub since the data transmissions of these devices could interfere with the USB power supply (crosstalk effects between data and power wires) and thereby degenerate signal quality. (iii) Front PC-USB-ports are often attached to the CPU's main-board by long ribbon cables. Running alongside a gigahertz processor this constellation introduces a vast amount of interference and noise.

6 Discussion

New technological developments make the development of a communication and hearing device with advanced and personalized signal processing of audio communication channels feasible. User inquiries underline that such a development would be accepted by the end users. Such a device has the potential of being accepted as an assistive listening device and 'beginner' hearing aid. However, the introduction depends on availability of consumer audio headsets with microphones and a bi-directional or only uni-directional low-power and low-delay link. If the link to the audio headsets is not low-delay or uni-directional, then environmental sounds can not be processed on the central processor, and the benefit of the PCS would be reduced to personalized post-processing of audio streams from telephony, multimedia applications and public announcement systems. This processing is usually not as computational demanding as algorithms for environmental audio processing, e.g., auditory scene analysis. As to whether the central processor could be used for processing computational demanding algorithms depends on whehter the data to be exchanged between central processor and headsets can be restricted to preprocessed signal-parameters and time-dependent gain values. The perspective of transmitting the full audio signal at very low delays seems unclear to date.

Sampling rates	16, 32, 44.1, 48, 96 kHz
Input	
Input sensitivity 0 dBFS	-17.5 dBu (0.3 V_{pp})
Impedance (1 kHz)	10 kΩ
Dynamic range (S/N)	-92 dB unweighted
Output	
Output level 0 dBFS	2.1 dBu (2.8 V_{pp})
Impedance (1 kHz)	7.4 Ω
Dynamic range (S/N)	-94 dB unweighted
Round trip	
Frequency response, -1.5 dB	8 Hz - 6.7 kHz @ 16 kHz
	9 Hz - 13.2 kHz @ 32 kHz
	10 Hz - 17.9 kHz @ 44.1 kHz
	11 Hz - 19.1 kHz @ 48 kHz
	14 Hz - 33.5 kHz @ 96 kHz
Frequency response, -0.5 dB	12 Hz - 5.4 kHz @ 16 kHz
	14 Hz - 10.5 kHz @ 32 kHz
	15 Hz - 14.1 kHz @ 44.1 kHz
	15 Hz - 15.1 kHz @ 48 kHz
	17 Hz - 27.8 kHz @ 96 kHz
THD (1 kHz, -32 dBu)	-75.1 dB

Table 3: Evaluation results of the dedicated audio interface.

The implementation of a prototype system revealed barriers and solutions in the development of a PCS as a concentrator of communication channels. The advantage of using high-level programming languages in algorithm development is partly reversed by the need of a floating point processor. Current SmartPhones and PDAs do have only fixed point processing. The continuous convergence of miniature notebook PCS and mobile phones might lead to a new generation of mobile phones providing floating point processing. Until then, the solution for the prototype was to chose a separate small notebook PC based hardware for the PHS with a network connection to the PCS. The processor performance of such a netbook is sufficient to host most recent advanced signal enhancement algorithms. The battery of a netbook computer provides a runtime which is sufficient for field testing. However, final realizations of the PHS for everyday use must provide significantly longer runtime before recharging. The audio quality of the dedicated audio interface is sufficiently high for field testing.

With the prototype system of the PHS a research and field testing hearing system is available. The reliability of the PHS as a field test hearing system was proven by the small variances in subjective evaluation of SRT improvement across audiological test sites. The system provides ways to subjectively assess new aspects of signal processing algorithms already in an early stage of algorithm development, e.g., preference rating and listening effort. The analysis of the example algorithm ‘COH’ shows that the concept of the PCS is feasible for the implementation of advanced signal processing schemes. Link delay and bandwidth are scalable for the broad class of signal processing schemes represented by the ’COH’ algorithm, and the protoype system allows for testing user benefit depending on the link constraints.

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