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Hearing in the Communication Society

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1 Executive Summary

This report will give an overview of the specifications required for the presentation of sound via the Internet for use in the HearCom Internet services. With this approach, the Internet sound access profile will be used as a structure for describing the sound environment at the end-user side and the server requirements for applications.

First, an overview is given on all possible applications and configurations that may be part of the HearCom Internet services.

Next an inventory is made on existing approaches for defining and controlling the sound reproduction system for the end-user. The end-user herself is characterized by the individual auditory profile on which the first outlines are given (the full auditory profile will be defined later in SubProject 1).

From that, the Internet sound access profile is structured by means of an XML application defined via an XML Schema. This deliverable describes the initial structure of the profile, which will be extended and tested during the project lifetime.

The Internet sound access profile is intended to be used for requirement specification of Internet sound applications and for special tools that can determine the Internet sound profile.

2 Introduction

One of the objectives of the HearCom project is to provide Internet services and applications that make use of sound, such as screening tests for the auditory function. For this, characterizations of the utilized Internet connections and the sound reproduction systems are required. In addition, the type and intentions of the users need to be specified.

In this report, a characterization is given of sound presentation systems with respect to the equipment available at the end-user, the characteristics of the Internet connection, and the individual hearing characteristics of listeners with and without hearing disabilities.

The report includes an overview of existing methods for controlling the Internet sound system. The specification of future methods is deferred to the development of HearCom eServices, scheduled for later project years. For this reason the title of this report does not refer to new control methods.

This report deliverable will serve as input for the specification and construction of the HearCom Internet eServices and other applications.

3 Applications via the Internet

This chapter presents an inventory of applications that present sound via the Internet or that are suitable for sound presentation via the Internet.

3.1 Internet sound applications

3.1.1 Entertainment

Many gaming and music-listening applications can present sound over the Internet. However, their sound control methods are designed for amusement purposes and usually not suitable for HearCom. Applications for entertainment fall beyond the scope of the HearCom project. These applications make use of streaming technologies, which are described in deliverable D10-3.

3.1.2 Screening tests

Below follows an inventory of existing tests that may be applicable for initial screening of hearing impairments that are suitable, or can be adapted, for execution via the Internet. This list of tests will be specified more extensively in WP1, deliverable D-1-1.

- Speech reception in noise test using 3-digit tokens as stimuli (Miller *et al*., 1951; Rudmin, 1987; Elberling *et al*., 1989; Smits *et al*., 2004). This test requires at least one speaker and a comfortable listening level.

- Spatial hearing/localization test using a virtual space implemented with standard head-related transfer functions (Wenzel *et al*., 1993; Begault *et al*., 2001). This test requires a set of headphones, a quiet listening environment, and a comfortable listening level.

- Measurement of the absolute threshold using tone detection in quiet (ISO 8253-1, 1989; ISO 389-7, 1998; ISO 389-8, 2004). This test requires a set of headphones, calibrated sound reproduction, and an extremely quiet listening environment.

3.1.3 Diagnostic tests

Below follows an inventory of existing tests to assess auditory impairments that are suitable, or can be adapted, for execution via the Internet. This list of tests will be specified more extensively in WP2, deliverable D-2-1.

- Measurement of the absolute threshold using tone detection in quiet (ISO 8253-1, 1989; ISO 389-7, 1998; ISO 389-8, 2004). This test requires a set of headphones, calibrated sound reproduction, and a quiet listening environment.

- Detection of sounds in noise test using the detection of pure tones in Gaussian noise. This test requires at least one speaker or headphone, a quiet listening environment, and a comfortable listening level.

- Language-independent speech-recognition tests such as syllable or CVC tests (e.g., Hirsh *et al*., 1952; Miller and Nicely, 1955). Such tests exist with open and closed-response sets. For application via the Internet, tests with closed-response sets might be the more suitable alternative. These tests require at least one speaker or headphone, a quiet listening environment, and a comfortable listening level.

- Speech reception in noise tests (ISO/TR 4870, 1991; ISO 8253-3, 1996), such as a 3-digit test (Rudmin, 1987; Elberling *et al*., 1989; Smits *et al*., 2004), a closed-response set rhyme test (House et al., 1965; Williams et al., 1973; Sotscheck, 1982; von Wallenberg and Kollmeier, 1989; Kollmeier et al., 1992; Kliem and Kollmeier, 1995; Brand et al., 1999), and a closed-response set sentence test (Hagerman, 1982, 1995; Wagener, 1999a, 1999b, 1999c). These test require at least one speaker a quiet listening environment, and a comfortable listening level.

- Spatial speech reception test using a multi-loudspeaker setup (Duquesnoy, 1983; Bronkhorst and Plomp, 1990; see also Plomp and Mimpen, 1981). Such a test can require five or more speakers, a quiet listening environment, and a comfortable listening level.

- Spatial speech reception test using a virtual space implemented with standard head-related transfer functions. (Arrabito *et al*., 2001; MacDonald et al., 2002). Such a test requires a set of headphones, a quiet listening environment, and a comfortable listening level.

- Horizontal spatial hearing/localization test using a multi-loudspeaker setup (Sanchez Longo *et al*., 1957; Blauert, 1997). Tones, noise bursts, and everyday sound can be used as stimuli. Such a test requires five or more speakers, a quiet listening environment, and a comfortable listening level.

- Vertical spatial hearing/localization test using a multi-loudspeaker setup (Blauert, 1969, 1997; Hebrank and Wright, 1974; see also Asano *et al*., 1990). Tones, noise bursts, and everyday sound can be used as stimuli. Such a test requires five or more speakers, a quiet listening environment, and a comfortable listening level.

- Spatial hearing/localization test using a virtual space implemented with standard head-related transfer functions. Tones, noise bursts, speech sounds, and everyday sounds can be used as stimuli (Wenzel *et al*., 1993; Begault and Wenzel, 1993; Begault *et al*., 2001; see also Loomis *et al*., 1990; Constan and Hartmann, 2003). Such a test requires a set of headphones, a quiet listening environment, and a comfortable listening level.

Next to variants of the aforementioned existing tests, the following new tests could provide possibilities:

- Measurement of spectral resolution
- Measurement of temporal resolution
- Measurement of cochlear compression

These measurements can be performed with the detection of tone sweeps in continuous and gapped-noise bursts, a relatively easily applied test that is presently in development at the VU Medical Center, Amsterdam (Hilkhuysen *et al*., 2005).

A powerful option is to use the Internet platform as a research and development tool in which new auditory test will be verified and evaluated. When modified for home use, tests can be presented via the Internet to address a large group of users. By logging all results, this allows the generation of large sets of data that can be used for test evaluation and verification.

3.1.4 Rehabilitation

The Internet provides a convenient channel for providing rehabilitation programs. Below follows an inventory of existing rehabilitation programs that are suitable, or can be adapted, for execution via the Internet.

- Training programs (e.g., speech reception for new cochlear implant wearers).
- Instructions (e.g., for various rehabilitation devices).
- Counseling (e.g., concerning tinitus, or aimed at the social environment of hearing-impaired listeners).
- Verification tests of training programs (either specific or the tests mentioned in sections 3.1.2 and 3.1.3).

This list will be specified in more detail by WP6 in deliverables D-6-1 and D-6-2.

3.1.5 Applications using sound contents

Most of the information provided by the internet applications will be in written text and graphics, such as instruction for running a specific application, help files, frequently asked questions, etc. But also more general information will mainly be presented in written text and in graphics; this concerns, for example, hearing and hearing impairments, rehabilitation devices, possible treatments, instructions for hearing-impaired listeners and their social environment, etc.

In a number of cases additional sound information will be useful. For example:

- In all the auditory tests described above in sections 3.1.2 to 3.1.4.
- When presenting sound demonstrations concerning, for example, different types of hearing impairment; the effects of environmental acoustics; various sound processing schemes (e.g., compression, noise suppression, cochlear implant processing strategies); examples of everyday sounds that often create problems for hearing-impaired listeners; etc.
- When providing spoken instructions and spoken information to supplement the provided visual information. In many situations, such an additional information channel can be useful.

A special set of circumstances occurs when the Internet is used to provide spoken information and spoken instructions to visually disabled users. In principle, such circumstances fall outside the scope of HearCom.

3.1.6 HearCom specific applications

- AVE (special requirements for AVE)
- Odeon (special requirements for Odeon)

The above applications are quite complex and will include audio streaming techniques. These applications will be realized during the second HearCom Project year. For a detailed description the reader is referred to the specific HearCom reports.

3.2 Sound-presentation configurations

The following classes of sound-presentation configurations can be identified:

- Headphones
- Simple speakers systems (mono and stereo)

- Multi speaker systems (surround systems)
  - Dolby surround 4.1, 5.1, and 7.1
- Professional sound systems
  - Dolby multi-speaker (more than eight) surround systems (e.g. in movies and theatres)
  - Other and Custom-made systems (e.g. as used in auditory laboratories)

In the HearCom project, only those sound-presentation configurations that are likely encountered in home situations will be taken into account.

4 Sound-presentation approaches

An inventory of currently used methods to control the presentation of sound using PC computers. Since many different soundcards are being used, special care needs to be taken to ensure good control over the presentation of sounds.

4.1 Internet and PC

A number of Internet sites and professional PC applications have been explored for methods on:

- Mixer control (channel selection, level, balance, advanced controls)
- Speaker setups (layout or geometry)
- Control of the sound quality (e.g., frequency characteristic, distortions, speaker polarity, etc)
- Control of the listening environment (e.g., noise, reflections)

Appendix A gives a list of the explored Internet sites and PC applications.

4.2 Soundcard control methods

Various applications used so-called wizards or mixer tools to control PC sound presentation independent of the installed sound card. Many soundcards are supplied with additional mixer and control tools to enable easy access to, and control of, all the features it offers.

- Sound control tools for Windows: mixers

  With such mixers the various channel levels and the muting of channels can be controlled. Sound control tools often run a check of the installed soundcard for all the options it offers. They can then be used to switch those options on and off.

- Soundcard wizards for Windows

  Wizards or often used to check specific features, for example whether simultaneous playing and recording works successfully with the installed soundcard.

4.3 Overview of methods:

An overview of existing, interesting, sound-control methods.

4.3.1 Balance and control of channels and speakers

- These depend on the soundcard used, so mixers are used to control the presentation of sounds. In addition, wizards are used to analyze specific properties of the soundcard.

4.3.2 Calibration of sound levels

- Sound levels controlled by user. Either the user is asked to adjust the sound level to a comfortable level, which is subsequently used in the application, or the user is asked to find a level at which test tones are just audible, after which a level increase is performed to achieve the level used in the application.

- Absolute calibration of sound levels, using an (accurate) sound meter, or a reference sound source and a microphone.

4.3.3 Sound quality

- Measurement of the frequency characteristics of the speakers, using specific test signals, such as chirps or maximum length sequences in combination with an omni-directional high-quality microphone (Audiolec Instruments, Behringer Technologies, Brüel og Kjær, Hewlett-Packard, MLSSA). If a PC is to be used for such an approach (e.g., Speaker Workshop) the soundcard in the PC will need to be able to play and record sounds simultaneously.

- Speaker polarity tests using correlated noises presented through the left and right channels (TestDisk.com, Rives Audio, Audiolec Instruments, eTesters).

- Distortion measurements of the sound production system using test tones and a dedicated measurement device such as a frequency analyzer. This analyzer can be either hardware (e.g., Stanford Research Systems, Hewlett-Packard), or software (e.g., Hung Chang). Alternatively, a high-quality microphone can be used in combination with a software analyzer (PC based digital storage oscilloscope, e.g., Hung Chang).

4.3.4 Environment

For free-field listening situations, some applications provide instructions concerning speaker positions, reflections from walls, and environmental noise (e.g., Beltone AVE., Dolby Laboratories).

5 The sound-presentation system

Different applications have different sound-presentation specifications. In this section a list is compiled of all the sound-presentation schemes required for the HearCom project, and it is indicated whether the necessary tools and instructions are available or have to be constructed.

Using a soundcard control tool (mixer), the levels and muting status of the input channels can be analyzed and controlled. With such a tool, the undesired channels can be muted while the desired ones can be enabled and have their volumes set to the correct values. Using a soundcard wizard, special features of soundcards can be analyzed, such as simultaneous playback and recording.

5.1 Transmission Characteristics

- Signal Bandwidth is determined by the speed of data transmission of the Internet connection. The end-to-end guaranteed data-rate should be determined. The use of Internet data streaming techniques will improve transmission speed characteristics for improved audio quality (see deliverable D-10-3).
- Signal delay is important for interactive applications. Some applications may have requirements on maximum data-delay for response or feedback.
- Data loss should be controlled to avoid hick-ups in sound presentation. Sound reproduction applications by Internet may have specific strategies and tolerances.

5.2 Channel and level control

- Control of channels and levels may depend on the soundcard used, so, a control tool (mixer) will be needed to check and set channel muting and levels. In addition, it may be necessary to provide a wizard that analyses the installed soundcard for specific features.
- Comfortable listening level controlled by used. For a number of applications (audio instructions, sound demonstrations, speech-in-noise tests) it is sufficient that the user sets the volume of the sound reproduction system to a comfortable listening level.
- Calibration of sound levels using a calibrated sound meter, or a reference source and a high-quality microphone. When absolute sound levels are required, usually in more professional applications, the sound reproduction system will need to be calibrated. This can

be achieved using a calibrated sound-level meter (e.g., Brüel og Kjær), or a much cheaper, non-calibrated but reasonable accurate, sound-level meter (e.g., Conrad, RadioShack), the latter is used for a CD for home measurements of audiograms (Digital Recordings). Alternatively, this can be achieved in a more expensive manner using a high-quality microphone (e.g., Brüel og Kjær, Behringer) in combination with a reference source (e.g., Brüel og Kjær, Conrad). For home use, the more popular, cheaper, solution will probably be the application of a non-calibrated but reasonably accurate sound-level meter, with an analog display these are available from €25, with a digital display from €50.

5.3 Sound quality

- Speaker frequency characteristics, using test signals (sweeps, chirps, maximum length sequences) and a high-quality omni-directional microphone (e.g., Brüel og Kjær, Behringer). Freeware software tools, like Speaker Workshop, may proof to be helpful in this issue.

- Distortion measurements using test tones (pure tones, tone complexes) and a high-quality omni-directional microphone (e.g., Brüel og Kjær, Behringer), and a software analysis tool. Alternatively, application of a condenser microphone of reasonable quality may be an option. Such microphones are more low cost and are available from approximately €25 (e.g., Conrad, RadioShack). An appropriate software tool has not yet been encountered and may need to be developed. Possibly, freeware software tools, like Speaker Workshop, can be helpful here.

5.4 Speaker layout and polarity

- Instructions concerning a polarity check of the speakers for two, five, and eight speaker systems. A test CD for stereo and 5.1 surround is available from Rives Audio. Basically, what is needed are pink-noise tracks that are in and out of phase in successive well-indicated burst, presented from all pairs of adjacent speakers in the system. From the perceived location of the noise the speaker polarity can be established, in phase produces a percept located between the two speakers, while out of phase produces a diffuse sound image.

- Instructions concerning the layout of the speakers for two, five, and eight speaker systems. Depending on the application, various speaker layouts may be needed. For every situation a clear description of the layout will be needed, preferably with clear maps. For some applications like localization tests, the acoustics of the room may need to be examined. This issue is dealt with in the next section.

5.5 Reflections and room acoustics

- Instructions for screening a setup for the influence of reflections. When a two-speaker virtual 3D system, or a multi-speaker setup is used in a localization task, the room will need to meet certain reverberation specifications. Below methods to estimate the reverberation decay time of a room are given. A quick and easy way to check a room for direct reflections is a mirror test (Beltone AVE.). This test requires two people and a mirror. One person sits in the position of the subject in the task, while the other slowly slides the mirror along the walls at eye height of the seated person. The latter can now record the positions where a speaker of the sound setup is visible. Next, small sound absorbing panels can be introduced at those locations to dampen the ensuing direct reflections.

- Calculation of the reverberation decay. The rt60 is an acoustical measurement used to calculate reverb time decay. The rt60 is the measurement of time it takes a given audio signal to fall -60db (decibels). The formula is rt60 = k*(V/Sa). In this formula, k is a constant that equals 0.161 when the units of measurement are metric, V is the volume of the room, and Sa is the total surface absorption. Companies like CSG Networks and Trinity Sound Company have similar Java engines available on the Internet, that can be used to calculate an estimated rt60 reverberation decay times from room dimension, wall types and surfaces, windows, etc (http://www.csgnetwork.com/acousticreverbdelaycalc.html).

- Impulse test: simple test by clapping hands or clicking fingers. A very simple way to quickly obtain a rough estimate of a room's acoustic properties is to produce a clicking sound by once clapping hands or clicking finger and then listening to the reverberations. After trying this in a couple of different rooms and circumstances it need not take very long to learn to compare them and thus roughly assess a novel environment.

- Measuring the reverberation time (rt60) using a recording test with a test sounds and a microphone. For the measurement of room acoustics, many alternatives have been developed. A simple and direct method uses the recorded decay of a white noise burst to estimate the rt60 (Schroeder, 1965). More sophisticated methods involving maximum-length sequences (mls) are also in much use (Schroeder, 1979). For the latter measurements it will be necessary to play and record a sound simultaneously, using a speaker and a microphone of reasonable quality. A software wizard may be needed to check whether the installed soundcard it suitable for simultaneous recording and playback. The produced recording will need to be analyzed by a software tool to produce a rt60 estimate.

5.6 Noise

- Measuring the noise level in the room using a sound level meter or a recording test with a calibrated microphone. To assess the level of background noise, a sound level meter with sufficient sensitivity (Conrad, RadioShack, Brüel og Kjær) can be used. Alternatively, and probably much more expensively, a calibrated high-quality omni-directional microphone setup (Brüel og Kjær, Behringer) can be used in combination with a software tool to analyze the recording.

6 Individual auditory profile

6.1 Hearing and communication profile

For many applications, it will be necessary to obtain an auditory profile of the listener. This profile will affect the presentation levels used in the application. In addition it can affect which tests, from the main body of auditory tests, will be applicable for each listener. The hearing and communication profile will be defined in SupProject 1. At this moment this profile is thought to consist of:

- Hearing profile:
    - The age of the listener. For normally hearing and hearing-impaired listeners age may affect some test outcomes. For this, a self-reported indication of their age will be needed.
    - Normal hearing in both ears, to be indicated by the listener.
    - Normal hearing in one, and impaired hearing in the other ear, to be indicated by the listener.
    - Impaired hearing both ears, to be indicated by the listener. In addition, many hearing-impaired listeners may have a copy of their audiogram. When present, that will constitute an important information source for adjusting the used setup.
- Language profile:
    - Non-native speakers, to be indicated by the listener. For non-native speaker, auditory tests using open response-set sentences lists are not suitable for application. In other cases it may be decided to work with non-speech tests only.
- Cognitive profile:
    - Memory span can affect the outcome of some tests. In the case of children it may be possible to correct the test outcomes using the average memory spans of children. This can be performed using the self-reported age. For some listeners, in particular elderly listeners, in may be necessary to include a test of memory span to be able to correct the test outcomes.
    - Language abilities. In the case of children it may be possible to correct the test outcomes using the average language abilities of children. This can be performed using the self-reported age.

6.2 Other accessibility/usability issues

The Consortium will analyse during the evaluation phase other issues related to accessibility and usability of the tests. In particular, it will consider in detail rendering issues of the HearCom portal as described in the Web Content Accessibility Guidelines 1.0 (Chisholm et al., 1999) and 2.0 (Caldwell et al., 2004) of the Web Accessibility Initiative.[1]

These requirements will be reflected in Section 13.3 (see also Velasco et al., 2004) when connecting user and device profiles.

[1] http://www.w3.org/WAI/

7 HearCom Internet Sound Access Profile

For supporting the HearCom Internet sound applications the Internet sound access profile will be defined. This profile will describe the characteristics for the use of HearCom Internet sound applications.

This profile will be used to describe the user characteristics at the user side and the requirements for a specific Internet application at the server side.

A match between the 2 sides is obtained when the profiles at both sides are equal, or more generally, when the user profile exceeds the application (requirement) profile.

A mismatch between the 2 sides can be resolved by:

- Additional characterizing of the user profile by adding additional information; For this special tools can be designed (e.g., by questions, by measurements)
- Upgrading the user profile by addition, change, adjustment and/or replacement of equipment.
- Downgrading the application (by less demanding requirements if possible)

This Internet sound access profile consists of:

- Internet Sound Profile, which includes user equipment and transmission profiles
- Internet Access Profile, which includes a hearing & communication profile and general accessibility profiles

These profiles will be formalized in next sections.

7.1 HearCom Internet Sound Access profile

The profiles will be formalized using structured descriptors based on XML (eXtensible Markup Language; Bray, et al., 2004) as frequently used in Internet applications. An introduction can be found in (Mark Johnson, 1999). The XML application will be expressed by means of a XML Schema (Biron, Malhotra, 2004).

XML will allow the exchange of data between Internet applications in a standard language. XML is extendible and thus allows future developments. The XML application can be analyzed by means of XML parsers and converted into specific application parameters.

The XML based profiles can be extended during the Project and beyond. In the sequel the initial minimum descriptors will be specified for use of HearCom Internet sound applications.

The HearCom Internet Sound Access profile (HearCom_ISAP) will be composed by the following major elements:

- HearCom Internet Sound Profile (HearCom_ISP): defined in Section 7.2, which describes the sound interface at the location of the user.
- HearCom Internet Access Profile (HearCom_IAP): defined in Section 7.3, which represents user profiles.

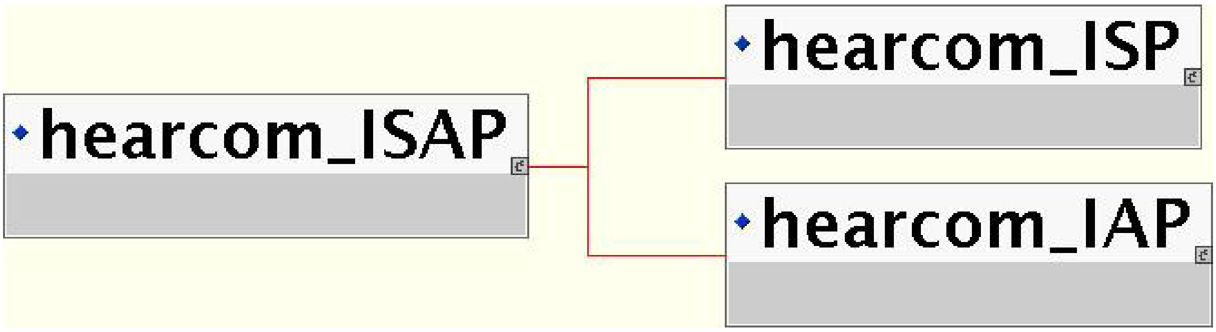

Figure 1. General structure of the HearCom ISAP.

7.2 Internet Sound Profile (HearCom_ISP)

The Internet Sound Profile describes the sound interface at the location of the user. The sound profile will be defined for HearCom applications but can be generalized for general purposes. The Internet Sound Profile will consist of the following components:

- Transmission
- Reproduction Devices
- Channel Control
- Sound Geometry
- Sound Quality
- Sound Environment

Note: the current specification will not include the sound profile for situations in which the user side will send sound input to an Internet application. The need and use for this will be specified in SP4/WP8 and will be added later to the above profile.

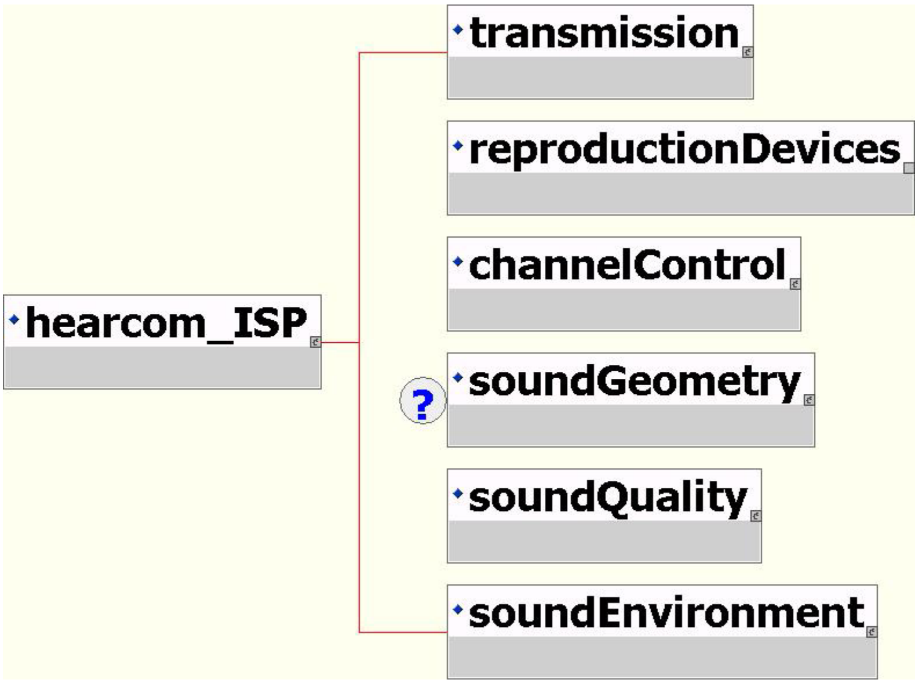

Figure 2. General structure of the HearCom ISP.

7.2.1 Detailed descriptors of the Internet Sound Profile

The detailed descriptors for the Internet Sound Profile will evolve during development of Internet sound applications. The initial detail descriptors for HearCom are given in next sections.

7.2.1.1 Transmission

Transmission relates to the characteristics of audio transmission. This profile will be characterized by the audio coding-method, the signal data rate (bps of the agregate bitstream) and quality as described by delay and errors. The main components are:

- audio_coding
- audio_rate
- audio_data_quality (composed of audio_delay and audio_data)

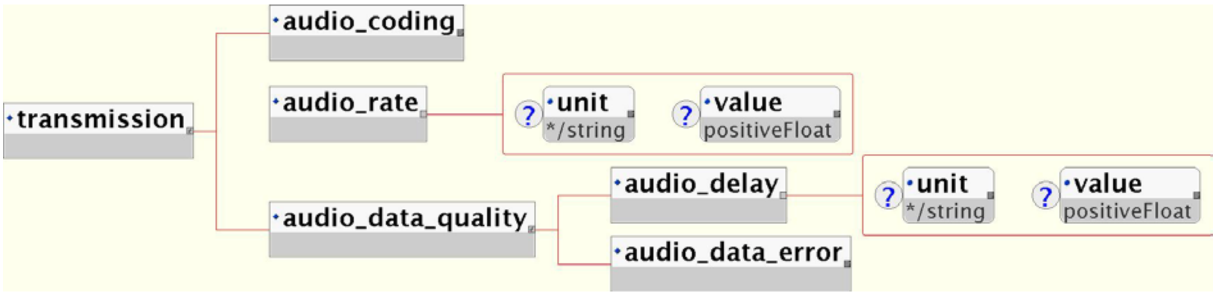

Figure 3. Transmission components

These components are described in Section 13.2.

7.2.1.2 Reproduction Devices

Reproduction devices relates to the type of sound output devices and the number of output (channels).

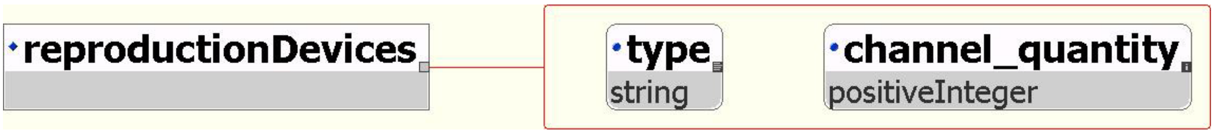

Figure 4. Reproduction Devices components.

These components are described in Section 12.3

7.2.1.3 Channel Control

Channel Control relates to how the output-channel is controlled and calibrated. Sensitivity refers to the sound level as measured at the place (middle of the head) of the listener for a test tone of 1 kHz full scale amplitude (16-bit) from a specific channel. Note that the actual calibration may be performed at normal sound levels and is converted to full scale units.

Accuracy refers to the error of the sound calibration method. This can be measured by a reference microphone, being professional or simple. Also a simple estimations can be applied. For instance an error rate of +/- 10 can be obtained when a test signal is adjusted to be comparable to a normal speech level. The actual numbers will be defined by the calibration method used or offered by the application.

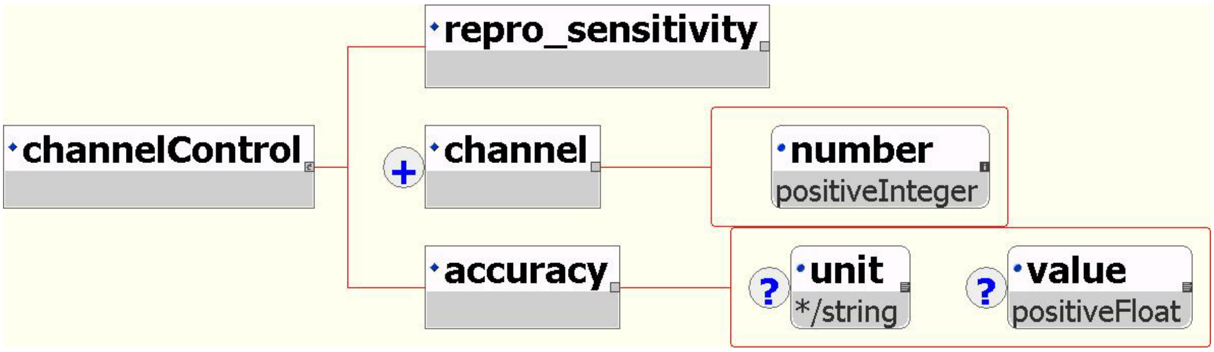

Figure 5. Channel Control components.

These components are described in Section 12.4.

7.2.1.4 Sound Geometry

The sound geometry refers to the placing of the speakers. Sound geometry is not relevant for systems with headphones and inearphones

The geometry can be defined by indicating the direction angle of each speaker and distance of the speakers. Standard setups can be defined by Stereo, Dolby x.x, front_only, front_and_back or otherwise. The elements will be defined during the project.

In case of multiple speakers it is possible to perform localization tests, but depending on the accuracy of the sound geometry.

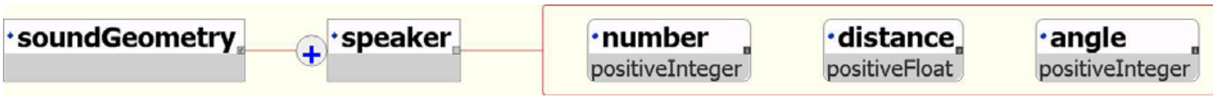

Figure 6. Sound Geometry components.

These components are described in Section 12.5.

7.2.1.5 Sound Quality

The sound quality is the quality of the reproduction devices. It will relate to frequency characteristics and distortion of each output device. A basic level can be obtained by self reporting. More sophisticated levels can be applied for good quality sound applications.

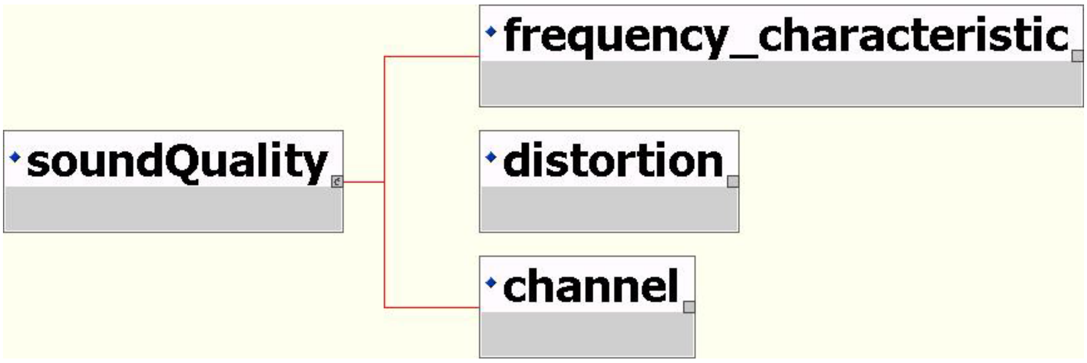

Figure 7. Sound Quality components.

These components are described in Section 12.6. The method(s) for defining characteristics and distortion are to be defined.

7.2.1.6 Sound environment

Sound_environment refers to the environment of the sound reproduction set-up and environment (e.g. room). Main relevant characteristics are the ambient noise level and reverberation time.

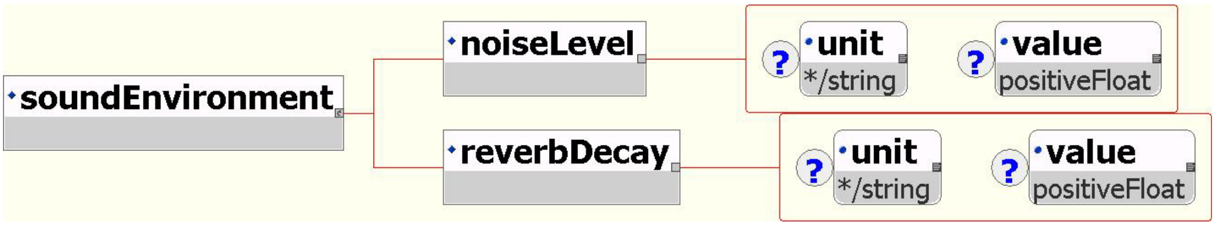

Figure 8. Sound Environment components.

These components are described in Section 12.7.

7.2.1.7 Integration

All these components are integrated in the Schema described in Section 12.8.

7.3 Internet Access Profile descriptor

The Internet Access Profile (HearCom_IAP) will consist from a user auditory profile (UAP) and a general Internet access profile (see Section 13.1).

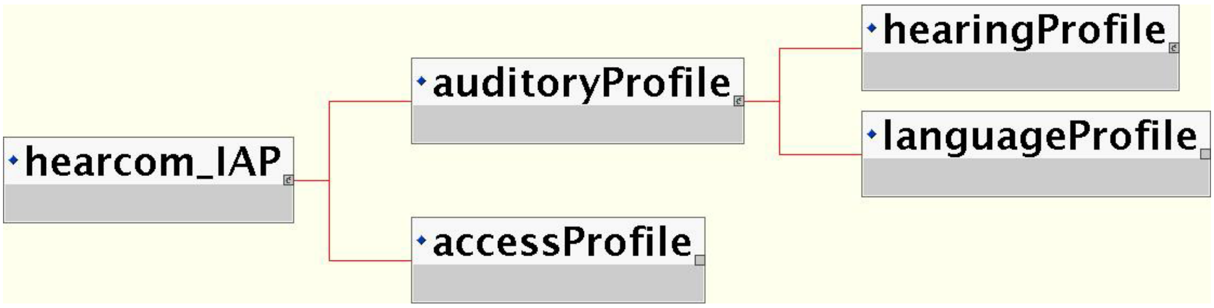

Figure 9. General structure of the HearCom IAP.

7.3.1 Auditory profile

The auditory user Profile is under development in SubProject 1. For the moment it may consists of: hearing (audibility, dynamic range, distortion..), language and cognition effects. A preliminary structure is presented in Section 13.2.

The cognitive and Language profile will be left open and will be defined when the auditory profile will be available from SubProject 1.

7.3.2 General Access Profile

In these first steps, the general access profile is going to be based upon the IMS Metadata specification for accessibility (see Section 13.3 for references).

7.4 Example Profile

Below an example is given of a profile used to specify the requirements of a typical headphone application. This example will concentrate on the HearCom_ISP profile.

The example for a headphone setup could be used for a screening test that estimates the ability of understanding spoken language in background noise. For this, no exact level calibration is needed and therefore the requirements are relatively broad allowing a large range of compatibility with the client environment (client is the Internet connection, PC, audio-components and the end-user). A typical XML representation of this screening test auditory profile would be:

```
<?xml version = "1.0" encoding = "UTF-8"?>

<hearcom_ISP xmlns:xsi = "http://www.w3.org/2001/XMLSchema-instance"

  xsi:noNamespaceSchemaLocation = "HearCom_ISP.xsd">
```

```
    <transmission>
        <audio_coding>MP3</audio_coding>
        <audio_rate unit = "bps" value = "64000"/>
        <audio_data_quality>
            <audio_delay unit = "ms" value = "5"/>
            <audio_data_error>0.01</audio_data_error>
        </audio_data_quality>
    </transmission>
    <reproductionDevices type = "headphone" channel_quantity = "2"/>
    <channelControl>
        <repro_sensitivity unit = "dBA" value = "100"/>
        <channel number = "1">
          <accuracy unit = "dB" value = "10"/>
  </channel>
        <channel number = "2">
          <accuracy unit = "dB" value = "10"/>
  </channel>
    </channelControl>
    <soundGeometry>
        <speaker number = "1" distance = "0" angle = "90"/>
        <speaker number = "2" distance = "0" angle = "270"/>
    </soundGeometry>
    <soundQuality>
        <frequency_characteristic value = "10"/>
        <distortion value = "0.01"/>
        <channel number = "1">
```

```
            <accuracy unit = "dB" value = "10"/>
  </channel>
    </soundQuality>
    <soundEnvironment>
        <noiseLevel unit = "dBA" value = "50"/>
        <reverbDecay unit = "ms (RT60)" value = "10"/>
    </soundEnvironment>
</hearcom_ISP>
```


The end user (client) side will confirm this profile by returning the settings that have been set at the required profile. In case client cannot meet the required profile the returned profile may deviate from the required profile. The server will accept these or will negotiate other values by a procedure to be defined in that application.

Dissemination and Exploitation

The Internet sound access profile is intended as the standard structure for specifying the requirements of Internet sound applications.

Special tools can be developed to detail the end-user sound access profile. By this the performance level of application and user requirement can be optimized.

By using Internet sound access profile the quality of the internet sound services can be optimized and its quality be assured.

The HearCom Internet services will be the first application of the sound access profile. Upon evaluation in the first HearCom Internet services the access profile can be used as the general interface for Internet sound applications.

The architecture of the developed systems will ensure that they are interoperable and therefore will guarantee its marketability and exploitation opportunities.

8 Conclusions

The HearCom Internet sound system profile will be applied for design and implementation of the HearCom internet services. The profile will serve as the formal interface to specify the characteristics of the user side and of the application side. The profile as such will facilitate the proper match between both sides. The development of adaptation procedures for achieving a best match will be part of the application. The inventory on existing approaches can be explored for an overview on existing approaches.

The Profile as described in this report will serve as a framework specification and is likely to evolve during the Project lifetime. The profile has been formalized using standard XML to allow the easy inclusion and adaptation into the HearCom Internet applications.

A specification for a user-source sound profile was not planned for this report. However SP4/WP8 may require such a profile, which extension will be included into D-8-1 (planned September 2005).

It was not possible to fully define the auditory Profile. The reader is referred to SP1/WP2 for a preliminary version of the auditory profile, which is planned to be available September 2005. The present report only gives a first outline.

9 References

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10 Appendix A: Soundcard tools, measurement tools, and hearing tests:

Audiolec Instruments, a company supplying a range of tools for acoustic measurements: http://www.audiolec.com/

Behringer Technology, a company supplying a range of tools for acoustics: http://www.behringer.com/

Beltone AVE™, Audio Verification Environment. A patient-focused fitting system. ©2000 Beltone Electronics Corporation, Chicago IL, USA.

Brüel og Kjær, a company supplying a range of tools for acoustic measurements: http://www.bksv.com/

Calibrate, a freeware tool to control PC sound and channel levels via Active-X. This tool has been written and is supplied by Dr. R. Cusack: http://www.mrc-cbu.cam.ac.uk/personal/rhodri.cusack/

Conrad Electronics and Technology, a mail order company supplying a range of tools: http://www.conrad.com

CSG Networks, a company specializing in network and computer support have a Java engine available on the internet, that can be used to calculate an estimated rt60 reverberation decay times from the room dimension: http://www.csgnetwork.com/acousticreverbdelaycalc.html

Digital Recordings, a company supplying various tools for hearing tests: http://www.digital-recordings.com/product.html

Dolby Laboratories, company specializing in “spatial” multi-speaker sound reproduction: http://www.dolby.com/

eTesters, website dedicated to testing methods for electronic and electrical equipment: http://www.etesters.com/

Hewlett and Packard, a company supplying intelligent analysis devices: http://www.hp.com

Hung Chang, a company supplying a range of tools for acoustic measurements: http://www.hungchang.com

MLSSA, DRA Laboratories, a company supplying a range of tools for acoustic measurements: http://www.mlssa.com

MMHotKeys (shareware), to control PC sound volume and channels at Library Smith Multimedia: http://www.librarysmith.co.uk/

Nationale Hoortest, a screening test for hearing by telephone. Smits, C., Kapteyn, T. S., and Houtgast, T. (2004). "Development and validation of an automatic speech-in-noise screening test by telephone," Int. J. Audiol. 43:15-28.

PowerMixer (shareware), to control PC sound volume and channels at Actual Solution: http://www.actualsolution.com/

RadioShack, a mail order company supplying a range of tools: http://www.radioshack.com

Rives Audio, a company specializing in acoustic engineering and associated tools, for home environments: http://www.rivesaudio.com

Snd_mixer, to control PC sound volume and channels at Matlab Central open exchange for Matlab and Simulink users: http://www.mathworks.com/matlabcentral/fileexchange

Speaker workshop, a freeware program for designing and analyzing loudspeakers that offers analysis possibilities by measurements via the soundcard: http://www.speakerworkshop.com/

Stanford Research Systems, a company supplying intelligent analysis devices: http://www.thinksrs.com

TestDisk.com, mail order company supplying a range of CDs for testing audio equipment: http://www.testdisc.vista.com

Trinity Sound Company, a company specializing in sound recording and sound recording systems have a Java engine available on the internet, that can be used to calculate an estimated rt60 reverberation decay times from the room dimension: http://www.trinitysoundcompany.com/rt60.html

11 Appendix B: XML Schema for Individual Hearing Profile

11.1 User profile for the Individual Auditory Profile

The needs for the user profile within this section are rather simple. The schema reflects for this case only the needed demographic data requested from Section 6.1, i.e., age and mother tongue, to modify the profile.

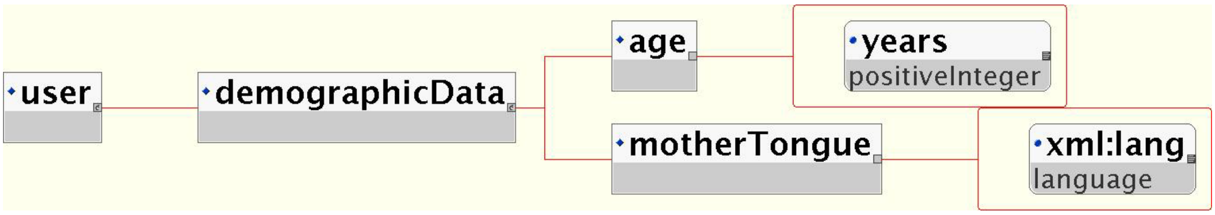

Figure 10. Graphical representation of the user profile schema.

This profile will be extended accordingly to studies like those of Velasco et al. (2004), to incorporate user and device profiling. The following listing presents formally the XML Schema:

```
<?xml version = "1.0" encoding = "UTF-8"?>
<xs:schema xmlns:xs = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xs:import    namespace    =    "http://www.w3.org/XML/1998/namespace"
schemaLocation = "http://www.w3.org/2001/xml.xs"/>
      <xs:element name = "user">
            <xs:complexType>
                  <xs:sequence>
                        <xs:element ref = "demographicData"/>
                  </xs:sequence>
            </xs:complexType>
      </xs:element>
      <xs:element name = "motherTongue">
            <xs:complexType>
                  <xs:attribute ref = "xml:lang" use = "required"/>
            </xs:complexType>
      </xs:element>
      <xs:element name = "demographicData">
            <xs:complexType>
                  <xs:sequence>
                        <xs:element ref = "age"/>
                        <xs:element ref = "motherTongue"/>
                  </xs:sequence>
            </xs:complexType>
      </xs:element>
      <xs:element name = "age">
            <xs:complexType>
                  <xs:attribute ref = "years" use = "required"/>
            </xs:complexType>
      </xs:element>
      <xs:attribute name = "years" type = "xs:positiveInteger"/>
</xs:schema>
```

11.2 Individual auditory profile

As described in Section 6.1, the individual auditory profile consists of two parts, one the hearing ability, and the other the user profile.

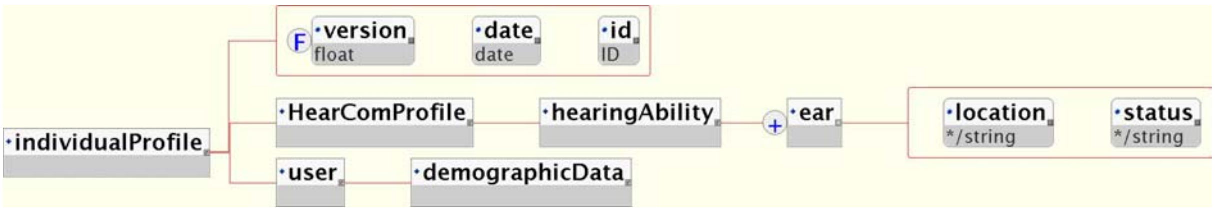

Figure 11. Graphical representation of the individual auditory profile.

The following listing describes the XML Schema:

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xs:schema xmlns:xs = "http://www.w3.org/2001/XMLSchema"
       version = "1.0"
       elementFormDefault = "qualified"
       attributeFormDefault = "unqualified"
       id = "hearcom_individal_profile">
      <xs:import namespace = "http://www.w3.org/XML/1998/namespace"
schemaLocation = "http://www.w3.org/2001/xml.xsd"/>
      <xs:include schemaLocation = "userProfile.xsd"/>
      <xs:element name = "individualProfile">
            <xs:complexType>
                  <xs:sequence>
                        <xs:element ref = "HearComProfile"/>
                        <xs:element ref = "user"/>
                  </xs:sequence>
                  <xs:attribute ref = "version" fixed = "1.0"/>
                  <xs:attribute ref = "date" use = "required"/>
                  <xs:attribute ref = "id" use = "required"/>
            </xs:complexType>
      </xs:element>
      <xs:element name = "HearComProfile">
            <xs:complexType>
                  <xs:sequence>
                        <xs:element ref = "hearingAbility"/>
                  </xs:sequence>
            </xs:complexType>
      </xs:element>
      <xs:element name = "hearingAbility">
            <xs:complexType>
                  <xs:sequence>
                        <xs:element ref = "ear" maxOccurs = "unbounded"/>
                  </xs:sequence>
            </xs:complexType>
      </xs:element>
      <xs:element name = "ear">
            <xs:complexType>
                  <xs:attribute ref = "location" use = "required"/>
                  <xs:attribute ref = "status" use = "required"/>
            </xs:complexType>
      </xs:element>

      <!-- Attributes list -->
```

```
        <xs:attribute name = "version" fixed = "1.0" type = "xs:float"/>
        <xs:attribute name = "date" type = "xs:date"/>
        <xs:attribute name = "location">
                <xs:simpleType>
                        <xs:restriction base = "xs:string">
                                <xs:enumeration value = "left"/>
                                <xs:enumeration value = "right"/>
                        </xs:restriction>
                </xs:simpleType>
        </xs:attribute>
        <xs:attribute name = "status">
                <xs:simpleType>
                        <xs:restriction base = "xs:string">
                                <xs:enumeration value = "normal"/>
                                <xs:enumeration value = "impaired"/>
                        </xs:restriction>
                </xs:simpleType>
        </xs:attribute>
        <xs:attribute name = "id" type = "xs:ID"/>
</xs:schema>
```

12 Appendix C: XML Schemas for the HearCom Internet Sound Profile

12.1 Common components

These are components reused by all other Schemas:

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
	 elementFormDefault = "qualified">
	<xsd:attributeGroup name = "units">
		<xsd:attribute ref = "unit"/>
		<xsd:attribute ref = "value"/>
	</xsd:attributeGroup>
	<xsd:element name = "channel">
		<xsd:complexType>
			<xsd:sequence>
				<xsd:element ref = "accuracy"/>
			</xsd:sequence>
			<xsd:attribute name = "number" use = "required" type =
"xsd:integer"/>
		</xsd:complexType>
	</xsd:element>
	<xsd:element name = "accuracy">
		<xsd:annotation>
			<xsd:documentation>unit must be set o "-dB" or "+dB".
Value is in the range 0 &lt;= value &lt;= 200</xsd:documentation>
		</xsd:annotation>
		<xsd:complexType>
			<xsd:attributeGroup ref = "units"/>
		</xsd:complexType>
	</xsd:element>
	<xsd:attribute name = "unit">
		<xsd:simpleType>
			<xsd:restriction base = "xsd:string">
				<xsd:enumeration value = "ms"/>
				<xsd:enumeration value = "bps"/>
				<xsd:enumeration value = "dB"/>
				<xsd:enumeration value = "-dB"/>
				<xsd:enumeration value = "+dB"/>
				<xsd:enumeration value = "dBA"/>
				<xsd:enumeration value = "ms (RT60)"/>
			</xsd:restriction>
		</xsd:simpleType>
	</xsd:attribute>
	<xsd:attribute name = "value" type = "positiveFloat"/>
	<xsd:simpleType name = "positiveFloat">
		<xsd:restriction base = "xsd:float">
			<xsd:pattern value = "([0-9]*\.)?[0-9]+"/>
		</xsd:restriction>
	</xsd:simpleType>
</xsd:schema>
```

12.2 Transmission

```
<?xml version = "1.0" encoding = "UTF-8"?>
```

```
<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "transmission">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "audio_coding"/>
                        <xsd:element ref = "audio_rate"/>
                        <xsd:element ref = "audio_data_quality"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "audio_coding" type = "xsd:string">
            <xsd:annotation>
                  <xsd:documentation>Text     string     of     max     26
chars</xsd:documentation>
            </xsd:annotation>
      </xsd:element>
      <xsd:element name = "audio_rate">
            <xsd:annotation>
                  <xsd:documentation>unit   attribute   must   be   set   to
"bps"</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "audio_data_quality">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "audio_delay"/>
                        <xsd:element ref = "audio_data_error"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "audio_delay">
            <xsd:annotation>
                  <xsd:documentation>unit  attribute  must  be  set  here  to
"ms"</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "audio_data_error" type = "xsd:string">
            <xsd:annotation>
                  <xsd:documentation>To be defined</xsd:documentation>
            </xsd:annotation>
      </xsd:element>
</xsd:schema>
```

12.3 Reproduction Devices

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "reproductionDevices">
```

```
            <xsd:complexType>
                  <xsd:attribute ref = "type" use = "required"/>
                  <xsd:attribute ref = "channel_quantity" use =
"required"/>
            </xsd:complexType>
      </xsd:element>
      <xsd:attribute name = "type">
            <xsd:simpleType>
                  <xsd:restriction base = "xsd:string">
                        <xsd:enumeration value = "headphone"/>
                        <xsd:enumeration value = "inearphone"/>
                        <xsd:enumeration value = "speaker"/>
                        <xsd:enumeration value = "other"/>
                  </xsd:restriction>
            </xsd:simpleType>
      </xsd:attribute>
      <xsd:attribute name = "channel_quantity" type =
"xsd:positiveInteger">
            <xsd:annotation>
                  <xsd:documentation>Maximum is 256</xsd:documentation>
            </xsd:annotation>
      </xsd:attribute>
</xsd:schema>
```

12.4 Channel Control

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "channelControl">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "repro_sensitivity"/>
                        <xsd:element ref = "channel" maxOccurs =
"unbounded"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "repro_sensitivity">
            <xsd:annotation>
                  <xsd:documentation>unit must be set to "dBA". Value is in
the range 0 &lt;= value &lt;= 200</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
</xsd:schema>
```

12.5 Sound Geometry

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "soundGeometry">
            <xsd:annotation>
```

```
                  <xsd:documentation>This element is optional and needs
further refinement.</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "speaker" maxOccurs =
"unbounded"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "speaker">
            <xsd:complexType>
                  <xsd:attribute name = "number" use = "required" type =
"xsd:positiveInteger"/>
                  <xsd:attribute name = "distance" use = "required" type =
"positiveFloat">
                        <xsd:annotation>
                              <xsd:documentation>Distance in
meters</xsd:documentation>
                        </xsd:annotation>
                  </xsd:attribute>
                  <xsd:attribute name = "angle" use = "required" type =
"xsd:positiveInteger">
                        <xsd:annotation>
                              <xsd:documentation>Angle in
radians</xsd:documentation>
                        </xsd:annotation>
                  </xsd:attribute>
            </xsd:complexType>
      </xsd:element>
</xsd:schema>
```

12.6 Sound Quality

```
<?xml version = "1.0" encoding = "UTF-8"?>
<!--Generated by Turbo XML 2.4.1.100. Conforms to w3c
http://www.w3.org/2001/XMLSchema-->
<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "soundQuality">
            <xsd:annotation>
                  <xsd:documentation>These elements need to be refined
further.</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "frequency_characteristic"/>
                        <xsd:element ref = "distortion"/>
                        <xsd:element ref = "channel" maxOccurs =
"unbounded"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "frequency_characteristic">
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "distortion">
```

```
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
</xsd:schema>
```

12.7 Sound Environment

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:include schemaLocation = "../common/common.xsd"/>
      <xsd:element name = "soundEnvironment">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "noiseLevel"/>
                        <xsd:element ref = "reverbDecay"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "noiseLevel">
            <xsd:annotation>
                  <xsd:documentation>unit must be set to "dBA". Value is in
the range 0 &lt;= value &lt;= 200</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "reverbDecay">
            <xsd:annotation>
                  <xsd:documentation>unit must be set to "ms (RT60)". Value
is in the range 0 &lt;= value &lt;= 100,000</xsd:documentation>
            </xsd:annotation>
            <xsd:complexType>
                  <xsd:attributeGroup ref = "units"/>
            </xsd:complexType>
      </xsd:element>
</xsd:schema>
```

12.8 HearCom_ISP Schema

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:import   namespace   =   "http://www.w3.org/XML/1998/namespace"
schemaLocation = "http://www.w3.org/2001/xml.xsd"/>
      <xsd:include schemaLocation = "./HearCom_ISP/transmission.xsd"/>
      <xsd:include schemaLocation = "./HearCom_ISP/soundGeometry.xsd"/>
      <xsd:include                     schemaLocation                       =
"./HearCom_ISP/reproductionDevices.xsd"/>
      <xsd:include schemaLocation = "./HearCom_ISP/channelControl.xsd"/>
      <xsd:include schemaLocation = "./HearCom_ISP/soundQuality.xsd"/>
      <xsd:include schemaLocation = "./HearCom_ISP/soundEnvironment.xsd"/>
      <xsd:element name = "hearcom_ISP">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "transmission"/>
```

```
                        <xsd:element ref = "reproductionDevices"/>
                        <xsd:element ref = "channelControl"/>
                        <xsd:element  ref  =  "soundGeometry"  minOccurs  =
"0"/>
                        <xsd:element ref = "soundQuality"/>
                        <xsd:element ref = "soundEnvironment"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
</xsd:schema>
```

13 Appendix D: Internet Access Profile

13.1 General overview

```
<?xml version = "1.0" encoding = "UTF-8"?>
<!--Generated by Turbo XML 2.4.1.100. Conforms to w3c
http://www.w3.org/2001/XMLSchema-->
<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:import namespace = "http://www.w3.org/XML/1998/namespace"
schemaLocation = "http://www.w3.org/2001/xml.xsd"/>
      <xsd:include schemaLocation = "./HearCom_IAP/auditoryProfile.xsd"/>
      <xsd:element name = "hearcom_IAP">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "auditoryProfile"/>
                        <xsd:element ref = "accessProfile"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "accessProfile">
            <xsd:complexType/>
      </xsd:element>
</xsd:schema>
```

13.2 Auditory Profile

```
<?xml version = "1.0" encoding = "UTF-8"?>

<xsd:schema xmlns:xsd = "http://www.w3.org/2001/XMLSchema"
       elementFormDefault = "qualified">
      <xsd:element name = "auditoryProfile">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "hearingProfile"/>
                        <xsd:element ref = "languageProfile"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "hearingProfile">
            <xsd:complexType>
                  <xsd:sequence>
                        <xsd:element ref = "audibility"/>
                        <xsd:element ref = "dynamic_range"/>
                        <xsd:element ref = "dynamic_range"/>
                        <xsd:element ref = "cognitiveProfile"/>
                  </xsd:sequence>
            </xsd:complexType>
      </xsd:element>
      <xsd:element name = "languageProfile">
            <xsd:complexType/>
      </xsd:element>
      <xsd:element name = "cognitiveProfile">
            <xsd:complexType/>
      </xsd:element>
      <xsd:element name = "audibility">
            <xsd:complexType/>
      </xsd:element>
```

```
    <xsd:element name = "dynamic_range">
        <xsd:complexType/>
    </xsd:element>
    <xsd:element name = "distortion">
        <xsd:complexType/>
    </xsd:element>
    <xsd:element name = "localisation">
        <xsd:complexType/>
    </xsd:element>
</xsd:schema>
```

13.3 User Profile

The user profile is going to be based upon the AccessForAll metadata specification. In particular, we will use the XML Schema available at: http://www.imsglobal.org/xsd/AccessForAll_v1p0.xsd