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

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Terminology

AVE	Auditory virtual environment
DSL	Digital subscriber line
HRIR	Head-related impulse response
HRTF	Head-related transfer function
HTTP	Hypertext Transfer Protocol
I-AVE	Internet-based interactive auditory virtual environment
IKA-SIM	Auditory virtual environment generator of IKA (Institut für Kommunikationsakustik der Ruhr-Universität Bochum)
JRE	Java runtime environment
PC	Personal computer
PDA	Personal digital assistant
RAM	Random access memory (main memory)
UDP	User Datagram Protocol

Pre-Amble

This deliverable describes the enhanced version of the Internet-based interactive auditory virtual environment (I-AVE) generator developed for the HearCom project. It builds upon deliverable D-3-2 which describes a stand-alone version and D-3-4 which describes the implemented localisation test using an I-AVE over the Internet. The enhanced version was used for the usability study described in deliverable D-11-4.

1 Executive Summary

Since the publication of D-3-2 and D-3-4 the I-AVE generator IKA-SIM which is used for the localisation self-screening test over the Internet has been continuously enhanced. This deliverable shall give an overview of the improvements and new applications of the implemented system.

Several valuable suggestions which arose from D-11-2, "Usability trials of the screening localisation test", have been adopted. A new test structure which allows skipping further demonstrations or repeating the instructions was implemented and the user interface of the self-screening applet was modified to become more user orientated, especially for those with little experience with PCs and the Internet. On the server side a new late reverberation generator was implemented and an interface to a MySQL database was added. In combination with the username which can be extracted now from an HTTP cookie personal test results can be stored for later analysis. The protocol which is used to control the I-AVE generator was extended to take advantage of these new features. The robustness of the audio streaming in the case of packet loss was further improved by a new packet retransmission mechanism.

In addition to the improvements of the self-screening test a new Java applet was written for demonstration purposes. This applet can be used to demonstrate ambient acoustics, spatial effects, and in general the potential of auditory virtual environments over the Internet. This applet uses the same server as developed for the self-screening localisation test and an extended client-server protocol.

2 Introduction

Auditory virtual environments (AVE) are used in a variety of applications, for example, in architectural acoustics, in entertainment and in 'edutainment' systems. Examples for such applications are Odeon¹, a professional software tool for acoustical room simulation, first person computer games, or the X-Fi sound card from Creative Labs² which uses a powerful digital signal processor (DSP) for spatial sound rendering. The purpose of such systems is to generate signals for binaural or loudspeaker reproduction which are perceived as being close to the auditory percept in natural environments. This requires the simulation of sound propagation in rooms and the generation of binaural signals for sound playback over headphones.

The real-time generation of an AVE is computationally demanding and cannot be easily implemented on end-user devices with limited computational power. On the other hand, it is desirable to make interactive AVEs accessible for low-end PCs, PDAs, Smart Phones etc. and thus extend the possible use cases. An interactive guide through virtual exhibition halls or edutainment software for mobile devices are examples for such a applications. While the purpose of the conventional World Wide Web was to serve and link documents, the second generation World Wide Web ("Web 2.0") is orientated towards providing services and linking people. An implementation of an AVE generator as web service could generally allow for platform independent "AVE services" for these use cases.

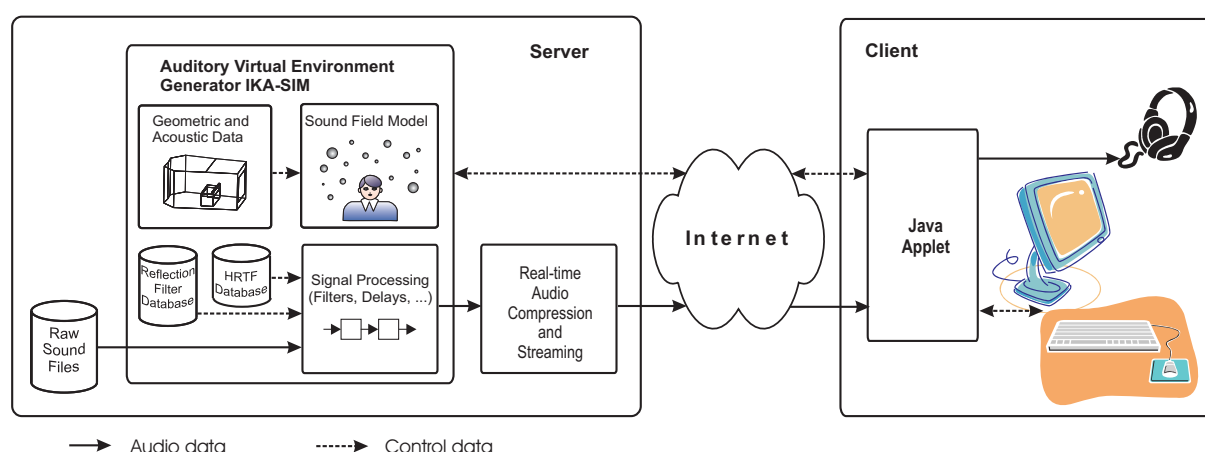


Figure 1: Block diagram of the Internet-based AVE generator

¹ <http://www.odeon.dk/>

² <http://uk.euro.creative.com/>

For the HearCom project a client-server approach to the generation of the AVE was chosen. While the computational power needed for real-time generation resides in the server, the client will execute only the user interface implemented as a Java Applet. The overall structure of the system is shown in Figure 1. The implemented AVE service can be used to demonstrate issues of ambient acoustics without high computational requirements or prior software installation on the client.

3 General Improvements to the Client-Server Architecture

3.1 AVE Generator (Server)

An AVE generator simulates the sound propagation from a sound source to a specific point in a room. In reverberant rooms numerous propagation paths exist. The output signal is composed of the direct path signal and signals reflected at the room boundaries. Along the transmission paths these signals are attenuated depending on the length of the transmission path, the number of reflections, and the reflecting material. Figure 2 shows a so-called “echogram” of the simulated living room. The abscissa marks the arrival times of the reflections and the ordinate the level of the reflections with respect to the direct sound. It can be partitioned into two parts, the early reflections and the dense late reverberation tail which results from overlapping reflections coming from – in principal - all directions [1,2]. The late reverberation is defined as the part of the reverberation tail which can be described by pure statistical means. For virtual rooms the simulation of early reflections and late reverberation can be done using different models. For physical rooms the transition from early to late reverberation is smooth.

Section 3.1.1 describes the generation of the early reflections and section 0 a new late reverberation generator. The output of the AVE generator is obtained by the sum of the output signal of both signal processing blocks.

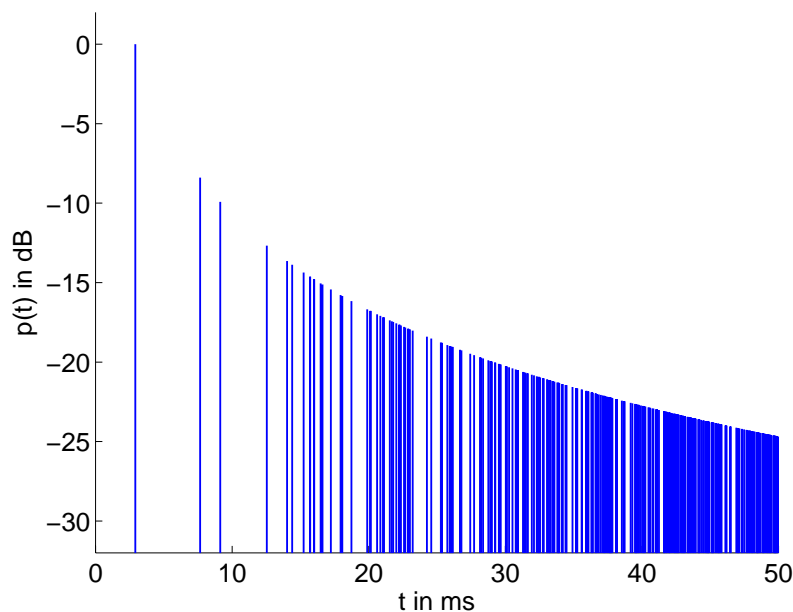


Figure 2: Reflection diagram for a rectangular room (5.5 x 4.5 x 2.7 m). The reflection level is plotted with respect to the direct sound level.

3.1.1 Early Reflections

The AVE generator comprises one or several sound sources, a single human receiver, and reflective boundaries. Under the constraints of a geometric propagation model the reflections may be computed using a mirror image model [3]. The simulation of the early reflections thus takes the directivity of the source, the reflections at the boundaries, the sound attenuation, and the propagation delays into account. The directional characteristics of human hearing are modelled by head-related impulse responses (HRIRs) which characterise the sound transmission from a source in the environment to a reference point in the ear canals under anechoic conditions [4, 5]. For headphone reproduction, these impulse responses are necessary for an out-of-head localisation. Off-line and real-time AVEs of this kind have been developed and described thoroughly [6, 7, 8].

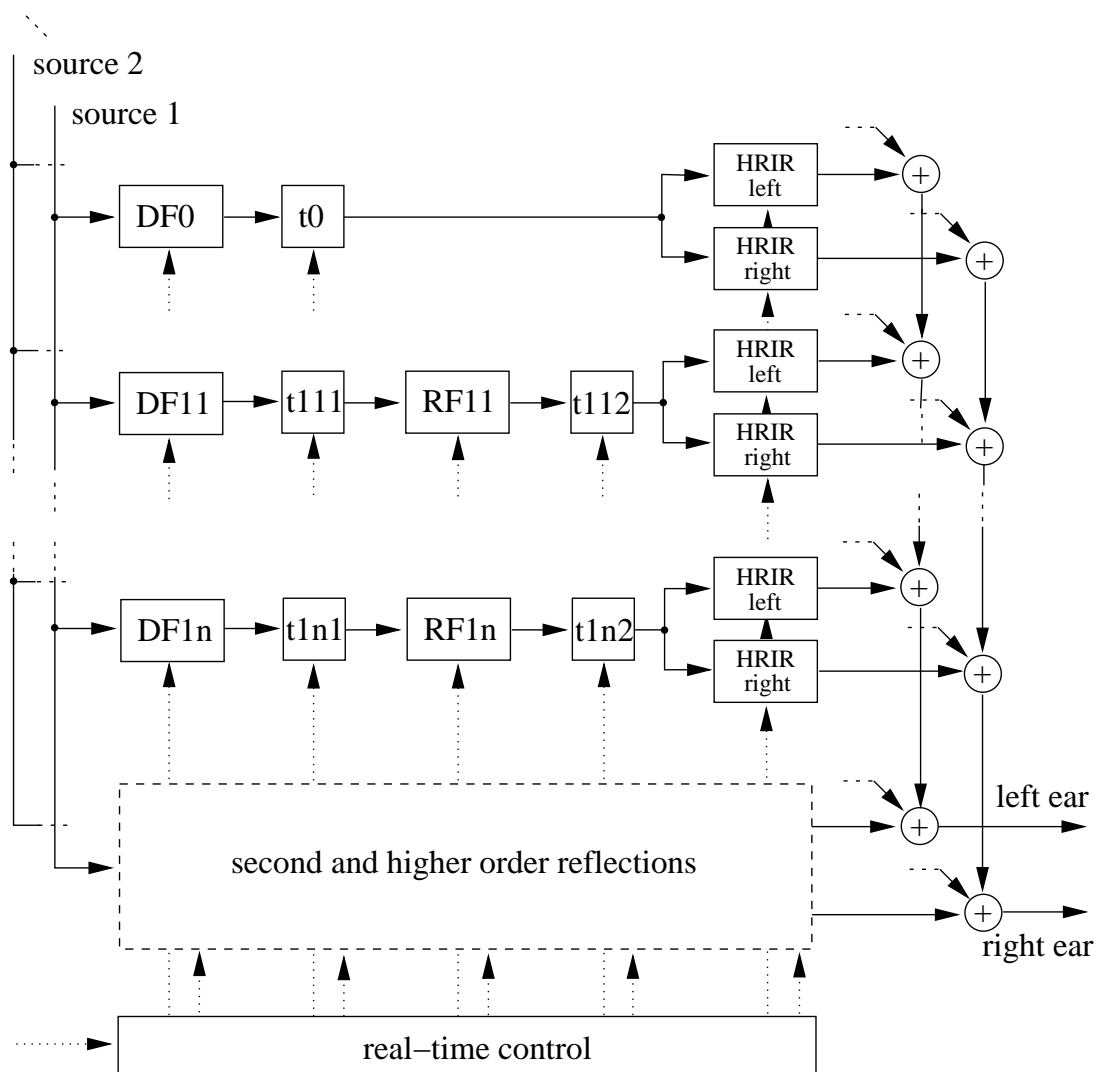


Figure 3: IKA-SIM signal processing model

Figure 3 details the signal processing model of the AVE generator IKA-SIM [9]. The source directivity and the boundary reflections are modelled by filters DF_x and RF_x , respectively, where x denotes the order of the reflection (only the direct path and the first order reflections are explicitly shown). Propagation delays are denoted by tx . In order to allow fractional delays, these are also realised by means of FIR filters. Note that the number of simultaneously active sources and the number of parallel branches, each of which simulates an acoustic path from the source to the ear canal, is limited by the computational power available for the AVE generator.

Despite several simplifications and powerful computers, the implementation of an interactive AVE generator is still a computational demanding task. The rendering of the mirror sources is the computationally most expensive part of a real-time generator. The computation of each mirror image source requires a directivity filter, reflection filters, time interpolation filters, and the HRIR for the left and the right ear.

For a simple rectangular room with six surfaces the number of mirror sources $M(o)$ up to order o can be directly calculated. Equation 1 results from the geometrical structure of the mirror image rooms with an order of less or equal o (a discrete double pyramid structure). The equivalence of equation 2 can be derived from the consideration that the limited sum over a second order polynomial yields a third order polynomial whose coefficient can be determined from 4 values obtained from equation 1 (e.g., $M(0)=1$, $M(1) = 7$, $M(2)=25$, $M(3)=63$),

$$M(o) = 2o^2 + 2o + 1 + 2 \sum_{i=0}^{o-1} (2i^2 + 2i + 1) \quad (1)$$

$$= \frac{4}{3}o^3 + 2o^2 + \frac{8}{3}o + 1 \quad (2)$$

For an implementation with uniform filter length this yields an algorithmic complexity of $O(N^3)$ for mirror sources up to order N . For more complex or non-rectangular rooms with S surfaces there are a maximum of S^o order o mirror sources. In this case a visibility check has to be performed for each mirror source resulting in an exponential algorithmic complexity $O(S^N)$. For the implemented system a rectangular room (5.5 x 4.5 x 2.7 m) was used because of the lower computational complexity which is needed for the generation of the mirror image sources. Informal listening tests with normal hearing expert listeners have shown that an extension of the simulated mirror sources to an order of 3 or above did not enhance the localisation accuracy significantly. Thus $o=2$ was chosen for the implemented system, i.e. 25 early reflections are simulated with the mirror image model.

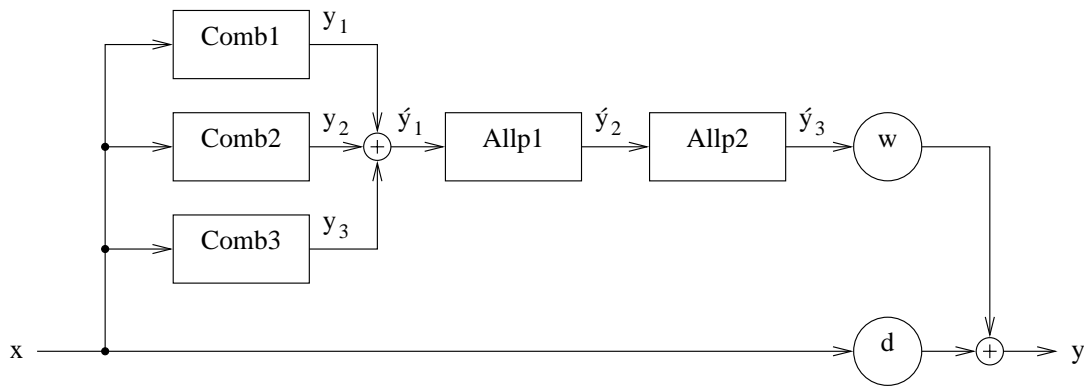


Figure 4: Parametric reverberation generator

3.1.2 Late Reverberation

The late reverberation results from dense reflections from all directions, has a diffuse character, and thus can be generated by a parametric reverberation generator [2]. While the stand-alone version described in D-3-2 used an external reverberation generator the current IKA-SIM version includes a new parametric reverberation module derived from Freeverb, a publicly available reverberation software³. Parametric reverberation generators of this kind have been developed and described thoroughly [10,11,12]. Although the structure of such reverberation generators is generally quite simple, tuning the parameters to gain a natural sounding result is not trivial. The newly developed reverberation generator module was optimised to be used in combination with the AVE generator.

While Schroeder used 4 parallel comb filters and 2 serial allpass filters [10], the new reverberation generator module shown in Figure 4 is constructed from 3 modified comb filters and 2 modified allpass filters, each with different parameters. A circle is used in these figures to denote

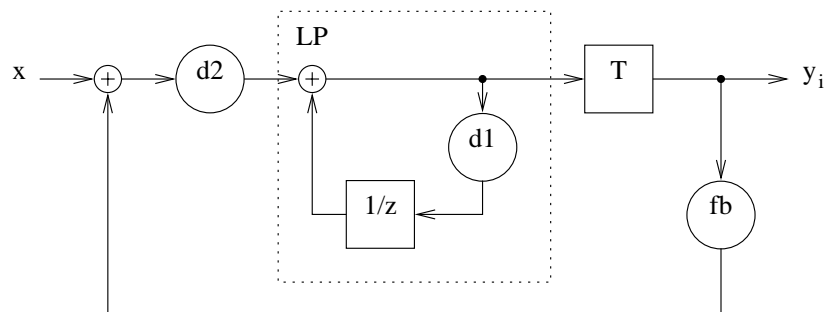


Figure 5: Modified comb filter element

³ Author: Jezar, Dreampoint Design and Engineering, <http://web.archive.org/web/20020327230357/www.dreampoint.co.uk/>

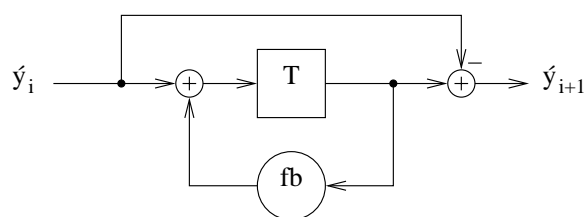


Figure 6: Modified allpass element

a multiplication (amplification) of a signal with a given value while rectangular boxes are used to denote general filter elements with internal input/output memory.

Figure 5 details the structure of the comb filters which are responsible for the frequency dependent decay of the impulse response. The feedback gain fb parameterises the reverberation time and the delay constant T the reflection density. The gain factors $d1$ and $d2$ parameterise the lowpass filter and by this the frequency dependence of the reverberation.

The modified allpass elements which are detailed in Figure 6 scatter the comb filter impulse response and increase the echo density, because a too low echo density leads to an unpleasant metallic or “fluttering” sound [10].

3.2 Database (Server)

The main purpose of the I-AVE used within the HearCom project is to use it for self-screening localisation tests. For a screening test which is conducted regularly it is useful to store the test results of each subject so that the development of the results over time can be analysed.

This functionality is implemented within IKA-SIM using an interface to a MySQL database. The personal test results are stored in a database together with the user ID which is identical to the username which will be assigned to the users of the portal for login. The test results of subjects who did not login to the portal are stored anonymously in the database. Table 1 gives details about which data is stored in the database.

3.3 Client-Server Communication

Two protocols are used to control the AVE generator on the server. The *IKA-SIM control protocol* and the *audio streaming control protocol*. Both are developed for the HearCom project and described in D-3-4, section 6.3. The following subsections specify the extensions to these protocols.

Variable	Comment
Date	Conduction date
Time	Conduction time
UserID	Unique username
TestNum	Identifier for the conducted test ("42" = DE-RUB localisation test)
Stimulus	String representation of the stimulus, telephone ring order
Answer	String representation of the given answers
ClassValue	Numerical value of the classifier output (floating point)
ClassString	String representation of the classifier output (e.g., "OK")

Table 1: Data stored in the database

3.3.1 IKA-SIM Control Protocol

Along with the client and server improvements the *IKA-SIM control protocol* was extended. This protocol is used by the client to control the server-side AVE generator (e.g., to set the position of a sound source). Listing 1 shows an example for the new "TestResult" command which instructs the server to store the given data in the database. For the date and time values the system clock of the server is used to avoid inconsistency due to wrong client system clocks.

A further extension of the protocol was necessary in order to remotely control the new late reverberation generator described in section 0. Listing 2 shows an example for the new "SetReverbSettings" command which sets the reverberation parameters accordingly.

```
<TestResult sUserID="Peter Mustermann"
nTestNum="42"
sStimulus="Stimulus sequence: 6 3 3 5 2 6 4 2 1 5 1 4"
sAnswer="Answer sequence: 3 3 3 3 3 3 3 3 3 3 3 3"
dClassValue="0.009057212590956125"
sClassString="no result" >
</TestResult>
```

Listing 1: Test result report

```
<SetReverbSettings nReverbMode="2"  
  fWetLevel="-6.0"  
  fDryLevel="-4.0"  
  fTimbre="-8.0"  
  fFeedbackGain="-6.0"  
  fEnvelopment="0.8"  
  nReverbDelayInSamples="1167">  
</SetReverbSettings>
```

Listing 2: Late reverberation generator parameters

3.3.2 Audio Stream Control Protocol

The binaural audio stream is sent to the client using the user datagram protocol (UDP). On the one hand this protocol provides for low latency transmission, but on the other hand it lacks a retransmission mechanism which can improve the transmission robustness in the case of packet loss. Such a retransmission mechanism was added to the *audio stream control protocol* which was developed for the self-screening test. When the client receives an audio packet with a non consecutive sequence number the missing packets are requested for retransmission immediately. As the client buffers the audio data to compensate network transmission jitter, a retransmitted audio packet can reach the client in time if the transmission takes less than the buffer latency. If the missing packets are not received in time the last packet is simply repeated. This best-effort mechanism cannot eliminate the problem of packet loss but can increase the probability that missing packets are retransmitted in time. This new feature improves the audio transmission robustness for Internet connections with low packet loss rates.

If a too high packet loss rate is detected, the audio streaming is aborted with an error message ("Error: Too much packet loss! Aborting test.") to inform the user that the current Internet connection is not appropriate for low-latency audio streaming.

3.4 Client

Several new features have been added to the java applet to adopt the outcome of the usability study from SP5. The test procedure, the kind of interaction, and all user instructions are defined in XML formatted scene definition files (see D-3-2, section 6.2.4). Each section in this file specifies a new screen which was shown to the user consecutively. To achieve more flexibility the latest version of the applet supports a non-linear execution of the specified scene instructions. By this, a certain part of the test, e.g. additional instructions, can be omitted or repeated.

Especially for instruction pages a demonstration mode was added which changes the virtual head orientation by automatically moving the slider. This feature can be used to explain the function of the slider and to demonstrate how the slider is linked to the visual and auditory environment.

As described in D-3-4 the virtual head orientation could be adjusted via the slider, via the *left* and *right* buttons, and via the cursor keys. This variety of possible interface elements made the whole interface more complex and thus less intuitive. The *left* and *right* buttons which changed the head orientation by a fixed angle step on each click were removed to simplify the user interface and thus to enhance the usability of the user interface.

Figure 7 shows the new user interface. The red triangles which mark the front position were added to support the user's orientation.

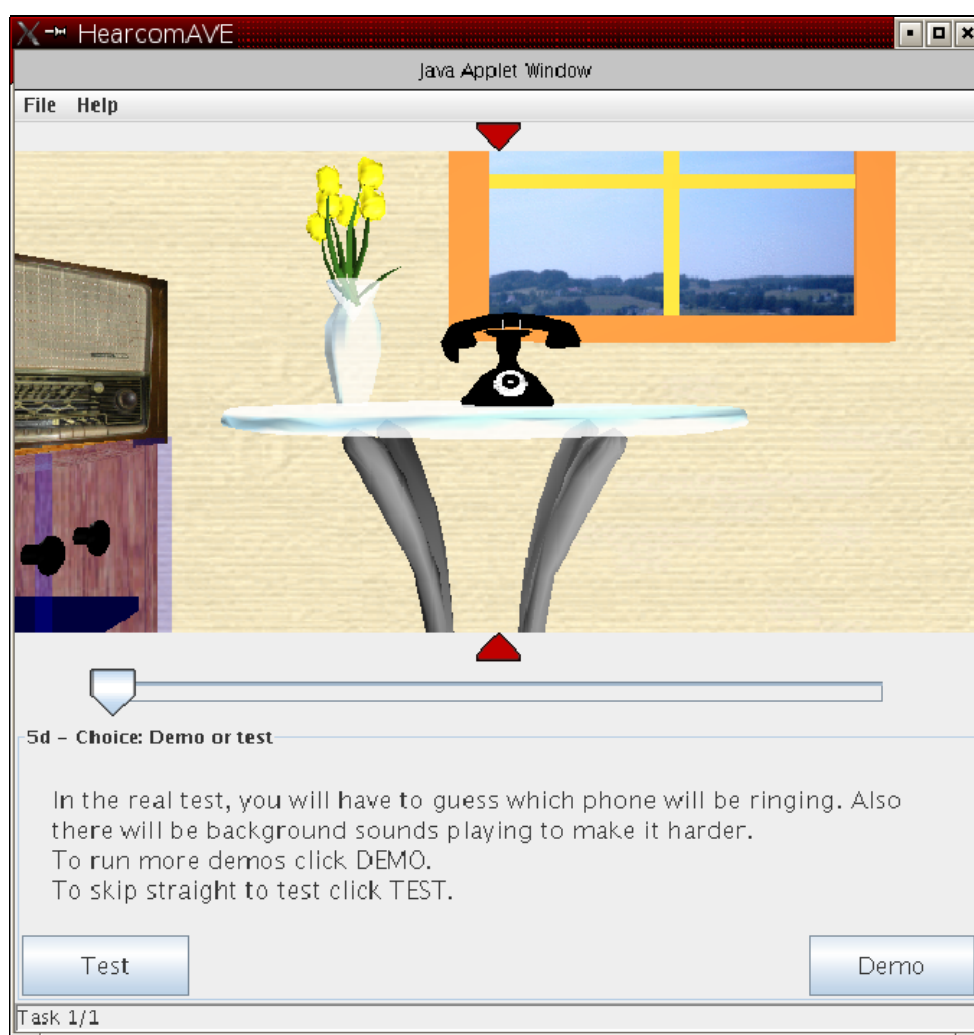


Figure 7: Screenshot of the DE-RUB localisation test applet

When subjects login to the HearCom portal their self-screening results can be stored in a database. A so-called *HTTP cookie* which is created by the web server and stored on the client contains the entered username. The latest version of the java applet extracts the username from the cookie so that personal test results can be sent to the server.

4 Improvements to the DE-RUB Localization Test

The test structure and the user instructions have been revised according to the suggestions of WP11. Figure 8 shows the new test structure. The displayed instruction of each screen is diagrammed by a rectangle, the user input by a parallelogram, and an input dependent fork by a hexagon. The actual test is highlighted by vertical twin lines.

The first instruction screen explains the function of the slider for users who are not familiar with sliders or computer games. The second screen directs the user's attention to the auditory environment and its correlation to the visual environment. The actual stimulus, the ringing phone, and the localisation strategy are introduced in the third screen. The fourth screen explains the function of the background music as interferer. The user has then the choice to proceed either directly to the test or to get more demonstrations. The first demonstration screen advises the user to localise a known sound source while the second demonstration screen instructs the user to localise an unknown sound source. After this the user can either proceed directly to the test or repeat the whole introduction.

In addition to the new displayed instructions the spoken instructions were updated with new recordings provided by NL-EMC.

4.1 Test result

The intention of a self-screening test is to give advice to its users. Based on the sequence of telephones the user has selected, the localisation performance in a virtual every day situation will be rated as "good", "marginal", or "weak". By estimating the probability that someone with good localisation performance has chosen the sequence of telephones under the condition of the stimulus sequence and estimating the same conditional probability for someone with weak localisation performance, an assignment to these classes can be done. A Bayesian classifier which simply compares these probabilities could be used for this task.

These conditional probabilities depend on the virtual room geometry, the position of the listener, the positions of the telephones, the reverberation, and the stimulus itself. For the implementation of the classifier, these probabilities have to be estimated for the chosen test setup. This can be done using the test results of a normal localising reference group and an impaired reference group. Practically, these probabilities can be estimated from the relative frequency of the chosen telephones during the validation

test which will be done by SP5. More details about this procedure and the theoretical background will be included in D-3-8.

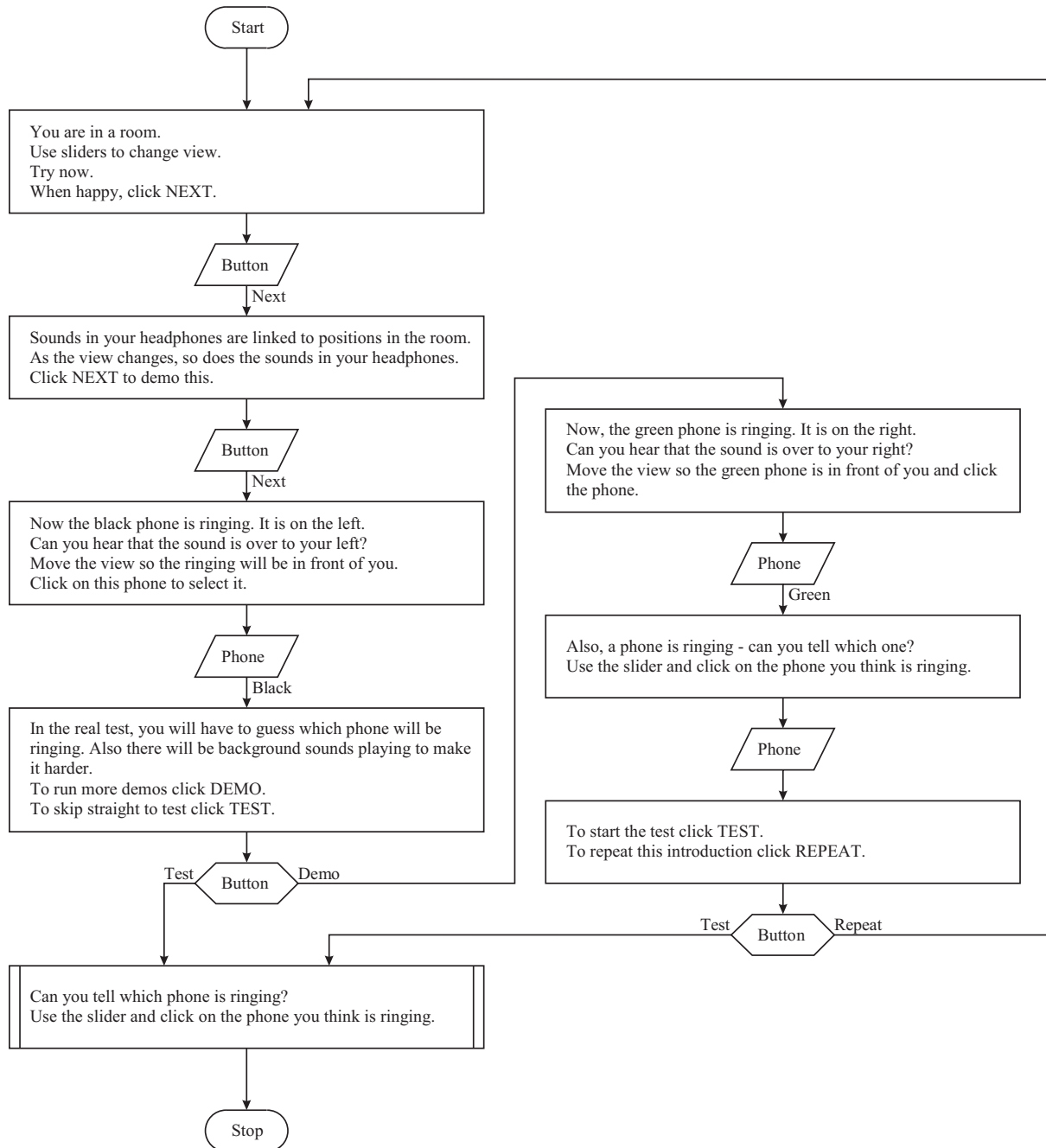


Figure 8: DE-RUB localisation test flowchart

5 Known Limitations

The purpose of the implemented system is to create a virtual environment which is perceived as being close to a natural environment. To enhance the level of immersion, a screen filling visual environment is used in addition to the auditory virtual environment. The level of immersion can further be enhanced by interactivity. Although a natural interaction with this environment is not possible, the system was designed to be as interactive as possible in consideration of the limited tools which are available at home. While all this enhances the quality of experience of the end user, it enlarged the technical requirements which are necessary to conduct the test. As a result, some users, who's local terminal equipment do not fulfil these requirements, will not be able to conduct the test.

The implemented system is based on some 3rd party software. The client uses Java, more precisely a Java Runtime Environment, and is executed in a web browser. The visualisation of the environment uses either Java3D or an image scrolling function from Java. The processed audio data is streamed over the Internet to the client and requires a fully functional Internet connection. The decoding of the audio stream is done by another 3rd party library, JLayer. A malfunction of these 3rd party software packages in conjunction with specific versions of operating systems, hardware, or web browsers can prevent some users to conduct the test. This serious problem can not be solved without a restriction of functionality.

5.1 Client

5.1.1 Java Runtime Environment

For the execution of the client a web browser with a Java Runtime Environment (JRE) version 1.4 or newer is required. Due to a more reliable JavaSound implementation, the usage of Java version 1.5 or 1.6, the current version, is recommended.

Unfortunately, the most recent JRE from Sun Microsystems, version 1.6.0_03, contains a software error⁴ which prevents the client from connecting to the server and thus makes it impossible to conduct the test in the Firefox browser. As long as this bug is not fixed by Sun Microsystems, this problem can only be solved by the user by downgrading to version 1.6.0_02 or older.

⁴ http://bugs.sun.com/bugdatabase/view_bug.do?bug_id=6622150

5.1.2 Java3D

If Java3D, an optional software package from Sun Microsystems which offers hardware accelerated 3D rendering, is available it is used for visualisation. It provides a faster and more realistic rendering of the visual environment.

Unfortunately, sometimes Java3D does not properly render the 3D scene and shows only a black screen instead. After changing the size of the window, the scene is properly rendered. The suggested workaround - to let the software automatically resize the window - was included in the java applet. This issue can only occur in combination with Java3D. De-installing Java3D solves the problem.

5.1.3 Graphic Card Driver / DirectX

If Java3D is not available, a section of a panorama image is used to create a spatial visual effect. This is internally implemented by a Java function which scrolls this image horizontally (without showing the horizontal scrollbars which are normally visible). On a Microsoft Windows PC the Java Runtime Environment uses DirectDraw functions to use hardware accelerated scrolling mechanisms. DirectDraw is part of DirectX, a software library from Microsoft which provides high-level access to graphic cards, sound cards, and network interfaces. Some users reported that the display was completely distorted ("encrypted") when the window was resized or the horizontal slider was used. This issue which is specific for single graphic cards was reported for PCs with a

Intel 82865G Graphics Controller or

ATI Radeon 9600 TX

graphic card. None of the tested PCs with a NVidia graphic card showed this error and it also did not show up if the PC was running under GNU/Linux. If this display error also occurs on other ATI Radeon cards, this would be a serious problem, because graphic cards from this series are quite common.

The problem could be solved on the ATI based PC by installing the latest graphic driver from ATI⁵ and installing the latest version of DirectX⁶. Deactivating the DirectDraw functions also solves the problem. If the problem also occurs with the latest drivers, this can be done by reducing

⁵ <http://ati.amd.com/support/driver.html>

⁶ <http://www.microsoft.com/downloads/Browse.aspx?displaylang=en&categoryid=2>

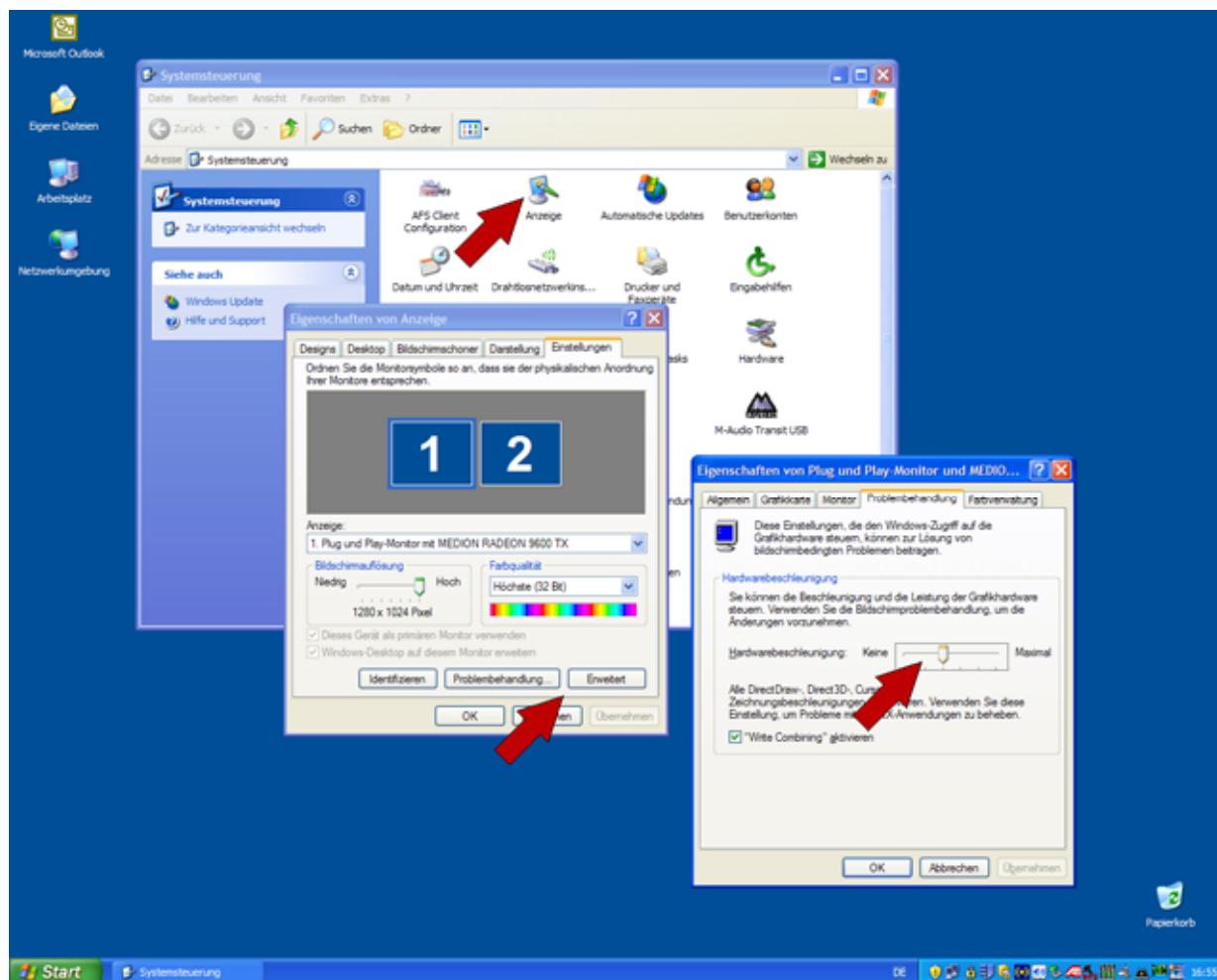


Figure 9: Solution for the Display Error

the value of the “hardware acceleration” slider under

Control Panel → Display → Settings → Advanced → Troubleshoot

as shown in Figure 9. Even though the suggested solutions solve this problem, they require some experience in installing software and drivers. This might be a problem for end-users with limited computer knowledge.

5.1.4 Internet Connection

The binaural audio signal which is streamed over the Internet to the client requires a fully functional Internet connection. This includes unrestricted bidirectional access to Internet services, low packet loss, and low transmission delay jitter. A Internet connection over a wireless network (WLAN) can be problematic due to a less reliable connection with respect to packet loss and jitter.

5.1.5 Firewall

Firewalls shall enhance the security by blocking network connections. A firewall software which protects only a single host is called “personal firewall”. We expect that a large number of end-users have installed such a packet filter, e.g. the Windows Firewall which is shipped with Windows XP. Such packet filters can prevent the usage of the DE-RUB Localisation Test if the end-user or the responsible administrator block – intentionally or unintentionally – essential network traffic. If the end-user has activated the Microsoft Personal Firewall, he is asked only once, if the browser shall get the permission to connect to the IKA-SIM server. If he denies this question, he won't be able to use the system. Because the HearCom portal can't include instructions for all kind of available firewall products in all kind of languages about how to configure the packet filter, it's the responsibility of the end-user to correctly configure their firewall. This is a difficult task for technically inexperienced end-users.

5.1.6 Hardware Requirements

The DE-RUB Localisation Test is intended to be conducted at home on an average PC (Intel Pentium-III/Pentium-IV/Core2 or AMD Athlon). Including a Firefox web browser (version 2.0.0.8), a Java Runtime Environment (version 1.6.0), and the memory usage of the applet, 70MB free main memory (RAM) is required to execute the test. This value will vary for other combinations. If not sufficient free memory is available, the resulting memory swapping can interfere with the audio playback. As the playback buffer increases for the case that the audio playback is interrupted, this also increases the memory requirements of the applet. The result is that the operating system needs to swap out even more memory pages to the swap disk and that the playback stops completely.

5.2 Server

During the usability study conducted for D-11-4 some issues related to the server were unveiled. E.g., IKA-SIM crashed reproducible when certain Windows clients connected to the system. This was caused by a different timing of the commands sent by the client. To start the playback of a sound source, the client sends a command to load a configuration file which describes the room dimension and the present sound sources, then instructs the server to start the sound renderer, and then sends the playback command. The playback command was received by the server while the loading procedure has not yet finished. This caused an error which terminated IKA-SIM and closed the network connection to the

client. This issue has been fixed by adding certain grace times to delay the execution of the received commands.

Another issue whose reason is not yet determined, is that in some rare cases single telephones are not played back. This needs further investigation.

6 New Applications of the Internet-based AVE

The DE-RUB localisation test applet with its special purpose makes only limited use of the I-AVE potential. To have an I-AVE generator with a higher degree of freedom a new java applet was written. This applet can be used to demonstrate ambient acoustics in closed environments. Figure 10 shows a screenshot of the new applet.

The horizontal projection of the room is displayed in the upper part of the applet. The listener is symbolised by a circle and a black dot which denotes the head orientation. Three sound sources are symbolised by numerated boxes. The position of the listener as well as the sound sources can be changed by “drag and drop”.

The lower part of the applet contains further control elements. The head orientation can be changed with a horizontal slider. Each sound source can be activated or deactivated with the corresponding “Play” and “Stop” buttons. In addition to the three movable sound sources, ambient noise

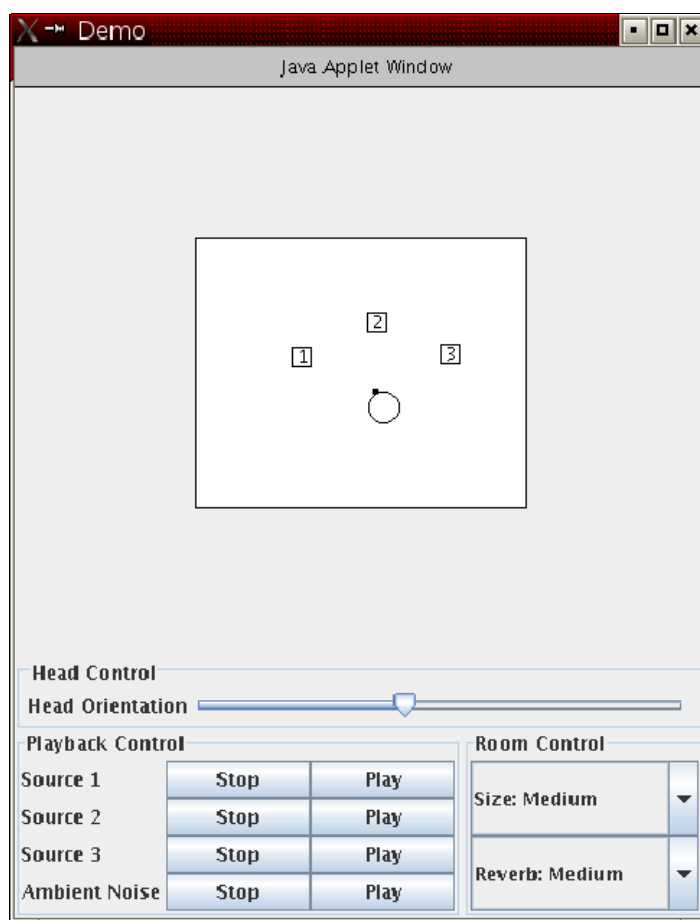


Figure 10: Screenshot of the DemoAVE applet

can be played back using pre-processed sound files. The room size which affects the generation of the early reflections can be changed with the "Size" drop-down list. It contains presets for a small, a medium, and a large room. The late reverberation generator can be influenced by the "Reverb" drop-down list which contains presets for low, medium, and high reverberation portion. This split of control elements for early and late reverberation makes it possible to distinguish between auditory effects based on early and late reflections.

This tool enables users in combination with a mechanism to upload wav files within the HearCom portal to process their own audio files in this demonstration applet. However, this mechanism is not yet implemented.

This applet can be used to demonstrate certain effects in architectural acoustics. It can also serve as a signal generation front-end for tests of hearing. More specifically, it could be combined with the Master Hearing Aid (MHA) to generate acoustic signals under adverse conditions and to improve them with the suite of HearCom enhancement algorithms for hearing-impaired listeners.

7 Dissemination and Exploitation

The approach for a localisation self-screening test based on an I-AVE has the potential to attract the general public. It will extend the services offered by the HearCom portal. A commercial exploitation depends on the marketing strategy chosen by the HearCom Internet and commercialisation board. In addition to commercial profit the results of the self-screening tests may be of value for researchers.

We believe that Internet-based AVE services have a high commercial potential. The implemented demonstration applet as well as the AVE based localisation test are just examples for Internet-based AVE services. Such a service could make an interactive AVE accessible for low-end PCs, PDAs, or Smart Phones. Due to the system independence the users are not limited to a special operating system or hardware. Thus the set of possible customers is not reduced by system requirements. Possible use cases for such a service are interactive guides through virtual exhibition halls, edutainment or communication software.

7.1 Ethical issues

The intention of a self-screening test is to give advice to its users. This advice must be carefully chosen as not to intimidate users, or to give faulty advice. A necessary next step therefore is to validate the localisation test. Validation tests will be conducted by WP11.

With the storage of personal data, i.e. the test result, in a database privacy issues have to be considered. A transparent privacy policy which specifies how long what data is stored and who has access to it is necessary. This policy has to be in accordance with national laws. As the username is used to store personal data, it should be chosen carefully. An acronym instead of the full name could provide anonymity to some extent. The topic which data can/shall/must be stored is an ongoing discussion within the HearCom project. On the one hand the test results can be of interest for the scientists. But on the other hand, the privacy of the test participants has to be assured and it must be clear what data is collected for which purpose.

8 Conclusions

The AVE generator on the server side, the java applet for the self-screening test on the client side, and the protocol for client-server communication has been improved. The self-screening test was updated

according to the outcome of the usability test conducted by WP11. After a validation the test is ready for public use.

A newly developed java applet can be used to demonstrate the potential of Internet-based AVE services.

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