EU Horizon 2020 - Contracts No 644051 and 101017743

3D Tune-In Toolkit Binaural Test Application v4.0 User Manual

This software was developed by a team coordinated by Arcadio Reyes-Lecuona (University of Malaga) and Lorenzo Picinali (Imperial College London). The members of the team are María Cuevas-Rodríguez, Carlos Garre, Daniel González-Toledo, Luis Molina-Tanco, Ángel Rodríguez-Rivero and Ernesto de la Rubia-Cuestas, (all affiliated to the University of Malaga).

Other collaborations: BRIR and near field simulation filters provided by David Poirier-Quinot (Imperial College London). Gammatone filter implementation developed by Mike Krzyzaniak (Imperial College London). Image Source Method developed by Arcadio Reyes-Lecuona and Fabián Arrebola. Adaptation of the user interface to include virtual sources developed by Paula García-Jiménez

The 3DTI Toolkit is a standard C++ library for audio spatialisation and simulation using headphones initially developed within the 3D Tune-In (3DTI) project (http://www.3d-tune-in.eu), and evolved within the CONICOM project (http://www.sonicom.eu), which aims at using 3D sound and simulating hearing loss and hearing aids within virtual environments. Technical details about the 3D Tune-In Toolkit spatialiser are described in:

> Cuevas-Rodríguez M, Picinali L, González-Toledo D, Garre C, de la Rubia-Cuestas E, Molina-Tanco L and Reyes-Lecuona A. (2019) 3D Tune-In Toolkit: An open-source library for real-time binaural spatialisation. PLOS ONE 14(3): e0211899. https://doi.org/10.1371/journal.pone.0211899

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Date of last revision: 27th Oct 2022

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Section 1: Introduction

This document presents the 3D Tune-In Binaural Test Application (Copyright © 2018-2019 University of Malaga and Imperial College London) user manual. This application is a technical demo for testing the features provided by the binaural 3D Tune-In Toolkit and the 3D Tune-In Resource Management Package (both Copyright © 2017 University of Malaga and Imperial College London). The 3D Tune-In Toolkit is a standard C++ library for audio spatialisation and simulation of hearing loss and hearing aids (http://3d-tune-in.eu/toolkit-developers). The 3D Tune-In Toolkit, together with 3D Tune-In Resource Management Package are released open-source under GPL v3 license (the code is available in GitHub https://github.com/3DTune-In/3dti_AudioToolkit).

Third party libraries.

- The 3D Tune-In Toolkit uses:
  - Takuya OOURA General purpose FFT (http://www.kurims.kyoto-u.ac.jp/~ooura/fft.html)
  - Eigen - A C++ template library for linear algebra (http://eigen.tuxfamily.org).
- The 3D Tune-In Resource Management Package uses:
  - Libsofa (Copyright © 2013-2014, UMR STMS 9912 - Ircam-Centre Pompidou / CNRS / UPMC. URL: https://github.com/sofacoustics/API_Cpp)
  - Cereal - A C++11 library for serialization (Grant, W. Shane and Voorhies, Randolph (2017). URL: http://uscilab.github.io/cereal/)
- This app also uses OpenFrameworks (Copyright © 2004- openFrameworks Community. URL: http://openframeworks.cc/)
- Together with this application, some HRTF files extracted from the LISTEN database are included: http://recherche.ircam.fr/equipes/salles/listen/index.html

The user interface of the Binaural Test App is divided into six panels:

1. Audio spatialisation configuration panel
2. Sources and listener layout chart
3. Audiometry panel
4. Hearing Loss Configuration panel
5. Hearing Aid configuration panel
6. Limiters configuration panel

The following figures show the different panels:

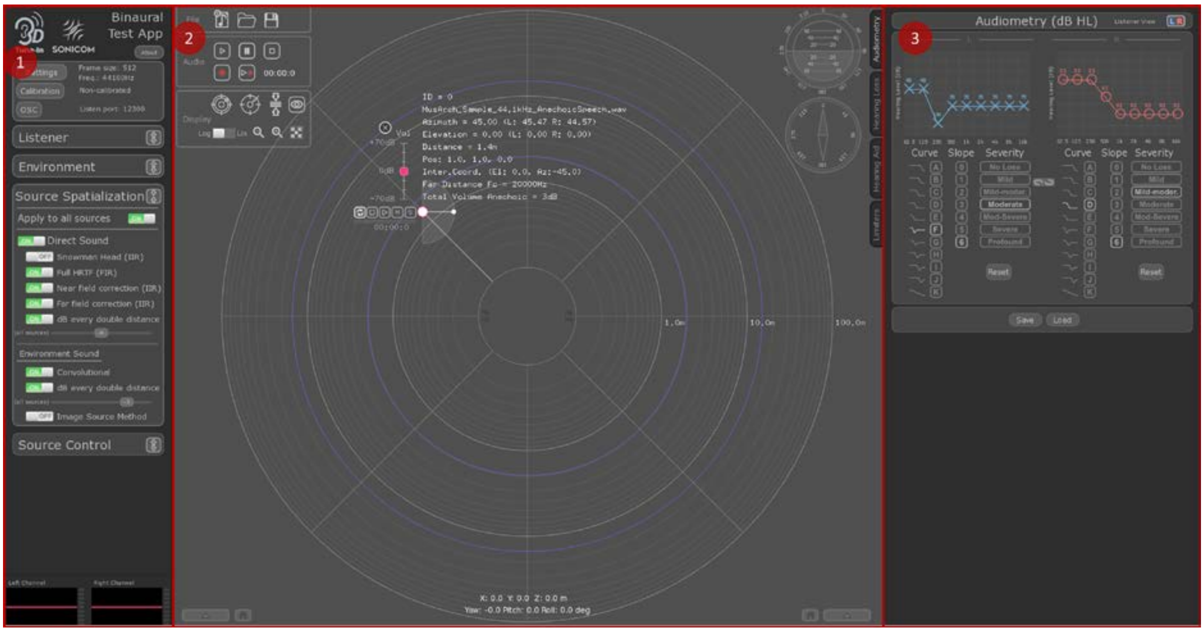


Figure 1. Binaural Test App showing spatialisation configuration (1), source and listener layout chart (2) and audiometry (3)

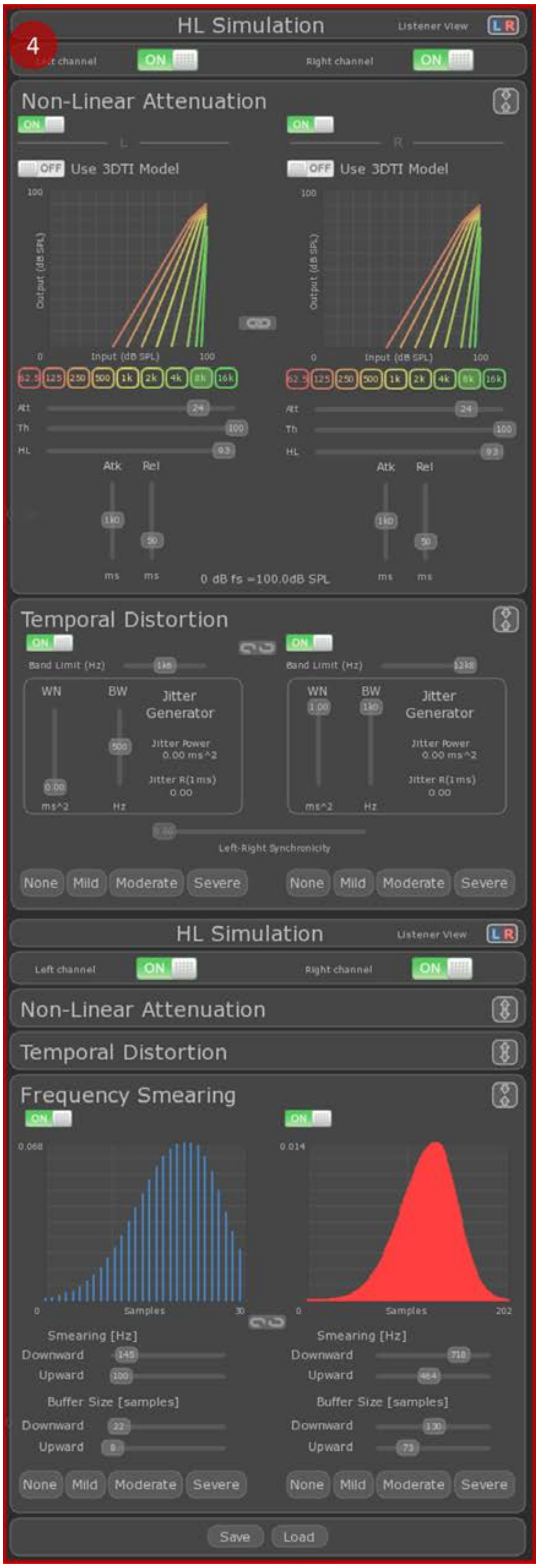


Figure 2. Hearing loss simulation controls (4)

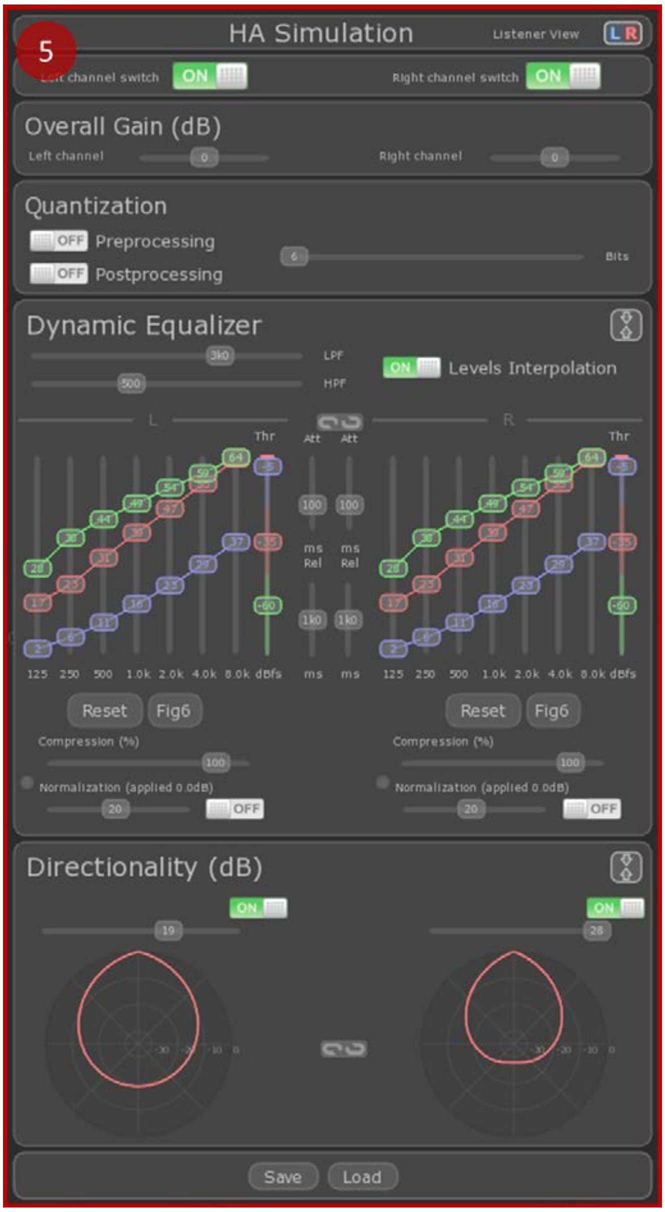


Figure 3. Hearing aid simulation controls (5)

Figure 4. Limiter controls (6)

The application interface is organised in three areas. The *Configuration panel*, to the left, and the *Source and listener layout chart*, at the center, are always visible. The *Audiometry, Hearing Loss simulation, Hearing Aid simulation and Limiter panels* share the rightmost area, which is hidden by default to give more room to the layout chart. These panels are hidden or shown by clicking the associated tabs.

This manual is organized as follows. The following section is a quick start guide for the user requiring just an overview of the application, which allows to test how spatialisation works. The remaining sections describe in detail the functionality and the user interface of the application.

Section 2: Quick start guide

The aim of this section is to explain the initial state of the application and how to start hearing spatialised audio.

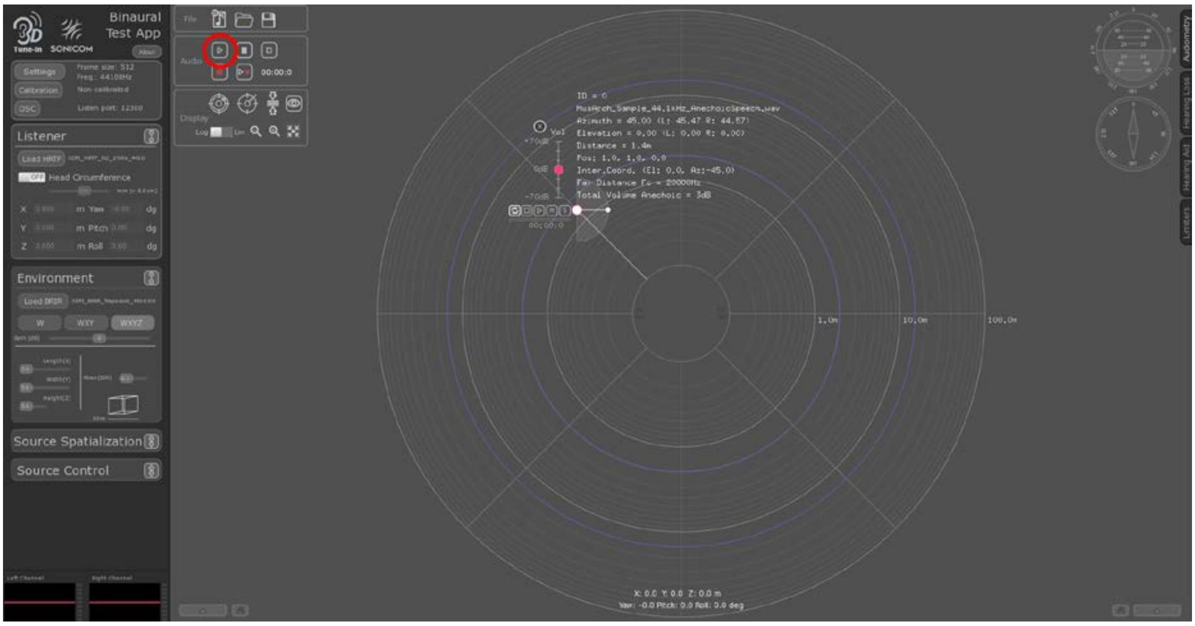


Figure 5. Binaural Test Application initial screen

By default, the application loads the initial configuration shown in Figure 5. This includes one pre-loaded sound source, a default listener head (convolution with HRTF loaded from a SOFA file) and a default environment configuration (convolution with BRIR loaded from a SOFA file). Click the *Play* button, shown circled in Figure 5, to test hearing spatialised audio including room reverberation (see Section 4 – subpanel 3: Audio clips control).

Each audio source is represented on the layout chart by a small white dot, together with a set of controls, which turn pink during audio reproduction. To move the sound sources, click and drag the dot to the desired position. The centre of the chart always represents the listener position. The listener can be also moved to change the relative position of the sources, which will approach, move away or turn around the listener, using the controls at the bottom corners of the chart (see Section 4 for more detail).

To add more sound sources to the simulation, load them from disk using the corresponding button at the top left corner of the chart (see Section 4 – subpanel 2: File controls). To change the spatialisation configuration by loading different HRTF or BRIR files, please refer to Section 3.

Section 3: Configuration panel

The Configuration panel is the leftmost area of the Binaural Test App. It is always visible and allows configuration of spatialisation and other global parameters. Controls are arranged into several subpanels

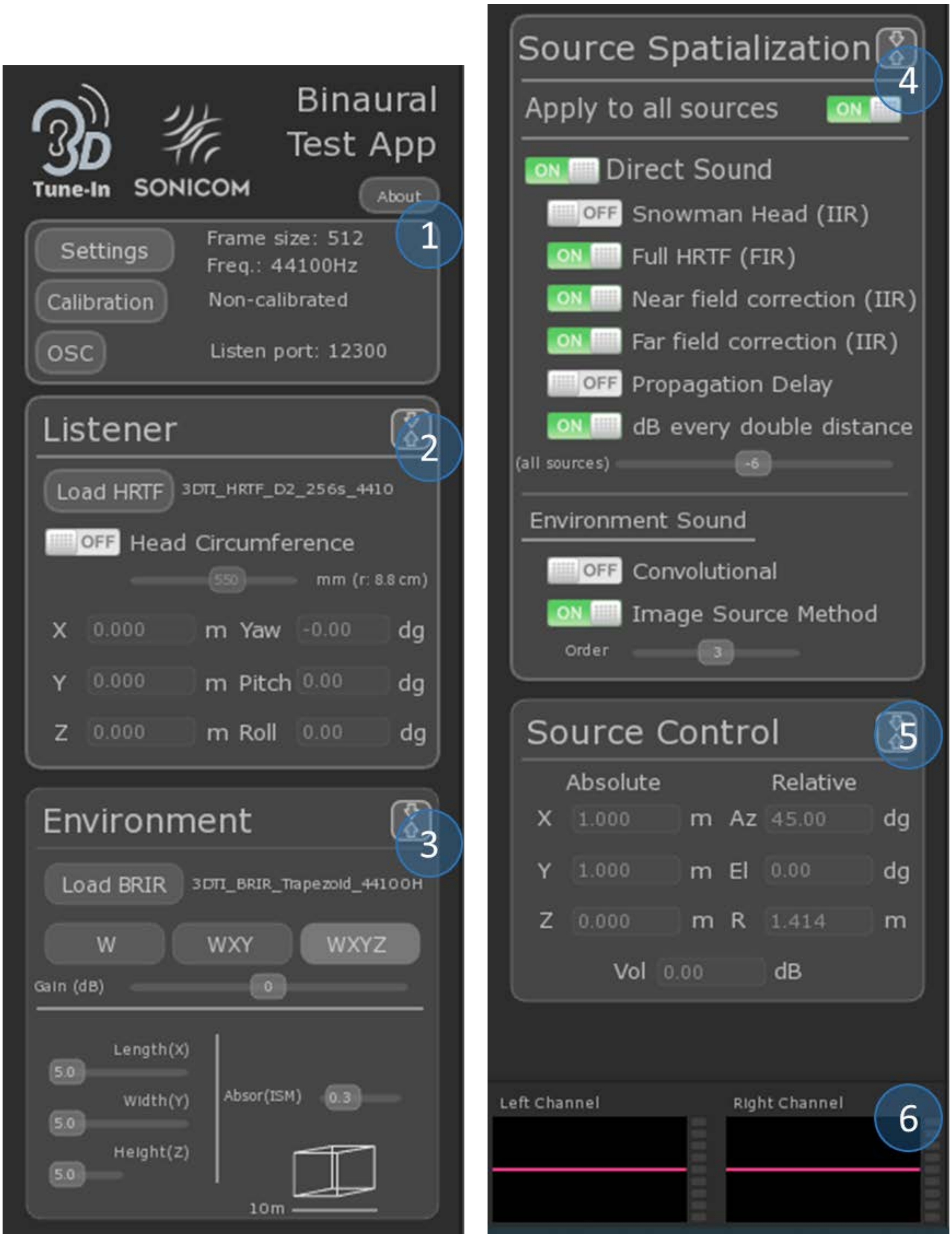


Figure 6. Configuration panel

The different controls of the panel are listed below. The numbering corresponds to the numbers in Figure 6:

1. Settings, calibration and OSC

The settings button opens a dialog window that allows changing many audio configuration settings, as shown in Figure 7.

The Frame Size, expressed as number of samples per frame, can be chosen within the different pre-sets (although the 3D Tune-In Toolkit internally supports other frame size values). The choice of the frame size depends on a compromise between latency and CPU cost (smaller values have smaller latencies but higher computational cost). Some frame sizes might not be supported by some audio interfaces (sound cards).

The sampling frequency at which all the sound sources and processing elements will operate can also be chosen from a set of values.

The HRTF resampling step parameter is used by the Toolkit during the HRTF interpolation process, after reading the HRTF data from file. For high spatialization quality, the HRTF resampling step should be small, while low memory (RAM) usage would demand large values. The default value is 15 and generally works well. A value of 5 would bring higher quality at the cost of an increase in memory usage.

The speed of the sound can also be specified.

The Available Audio Interfaces list shows all audio interfaces detected, allowing to change the audio interface to be used by the application.

The Folder to save recorded files is used by the record functionality to avoid the need of asking the user where to store recorded audio for each recording. The File name defines a pattern for the recorded files. An increasing numeral will be added to this file name for each recording. In the example of Figure 7, where File name is *Recorded.wav*, the first two recordings will be named *Recorded001.wav* and *Recorded002.wav*, respectively. The recordings will be standard wav files which format is defined by the setting Resolution, allowing to write either 16 or 24 bit (per sample) files.

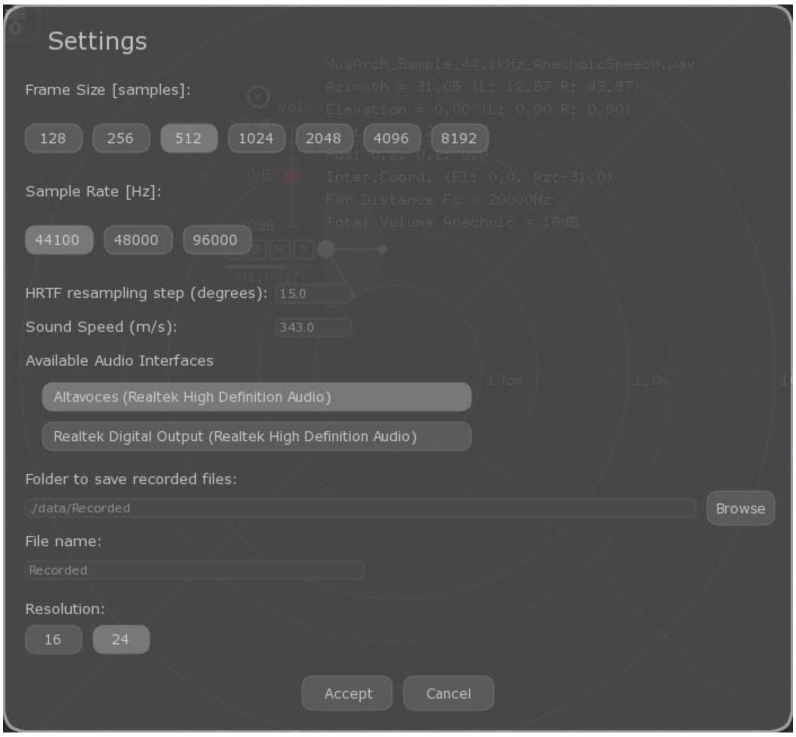


Figure 7. Settings dialog window

The calibration button opens a dialog window providing access to the calibration procedure, as shown in Figure 8. This procedure requires using a sound meter, in order to measure the Sound Pressure Level, in decibels, produced by your headphones. This process will characterize your amplifier and headphone system for internal calibration of the dynamic range. This process of calibration is needed to obtain realistic levels when using the Hearing Loss and Hearing Aid simulators.

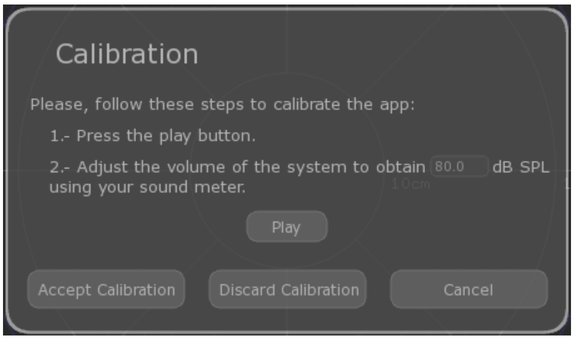


Figure 8. Calibration dialog window

The OSC button opens a dialog window that provides access to the OSC configuration menu. The Binaural Test application implements the Open Sound Control (OSC) protocol, which is a protocol for networking sound synthesizers, multimedia devices and others. The Binaural Test Application allows other applications implementing this protocol to control position and orientation of the sound sources ant the listener. In addition, the Binaural Test Application is able to control the position and play state of sound sources and listener position and orientation of a third application.

The OSC configuration menu, shown in Figure 9, allows configuration of the most important parameters[1] needed to use the OSC protocol, namely:

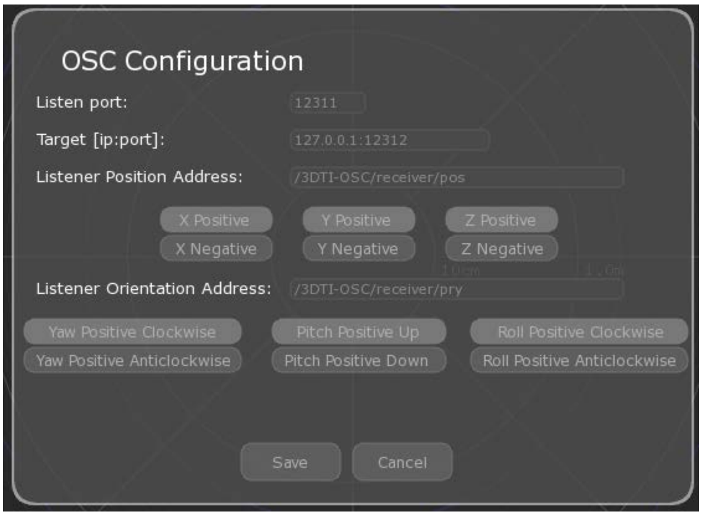


Figure 9. OSC Configuration dialog window

- **Listen port:** allows setting the UPD port on which the Binaural Test Application will listen for the reception of OSC messages.
- **Target [ip:port]:** allows setting the IP and port of a third party application to which we want to send commands from the Binaural Test Application. The target is specified following the usual *ip:port* format (for example: 192.168.1.2:12100).
- **Listener Position Address**: allows setting the command string used to receive or send the listener position. Additionally, the sign of any of the axis can be inverted before being applied to the listener position. To this end, the OSC configuration dialog window offers six buttons: *X Positive, X Negative, Y Positive, Y Negative, Z Positive* and *Z Negative*.
- **Listener Orientation Address:** allows setting the command string used to receive or send the listener orientation. Additionally, the sign of the received rotations can be inverted before being applied to the listener orientation. To this end, the OSC configuration dialog window offers six buttons: *Yaw Positive Clockwise, Yaw Positive Anticlockwise, Pitch Positive Up (clockwise seen from listener's left), Pitch Positive Down, Roll Positive Clockwise* and *Roll Positive Anticlockwise*.

For a complete list of OSC commands, see

[1] Please, note that not all parameters can be configured. For a complete list of OSC commands, see

Section 9. OSC commands

2. Listener

The listener subpanel allows the user to load the desired HRTF as well as setup the radius of the listener's head. The position and the orientation of the listener can also be entered through textbox controls.

3. Environment

The contribution of the environment can be configured separately for each source using either a convolution with a Binaural room Impulse Response or the Image Source Method (ISM). See the Source Spatialisation subpanel to know how to choose the technique to render the environment for each source. In this Environment subpanel, the room used in both techniques are configured. The subpanel is divided into two parts. The upper part configures the convolutional reverb, while the lower part configures the Image Source Method (see Figure 10).

Figure 10. Environment subpanel

- **Convolutional**: The Load BRIR button allows the user to load BRIR data from a SOFA (using Simple Free Field HRIR convention) or 3DTI-BRIR file. The buttons labelled as W, WXY, and WXYZ configures whether the four channels of the first order Ambisonics reverb processing are used (WXYZ) or just the horizontal plane (WXY) or just the omnidirectional (W). It is also possible to set the overall gain of the reverb effect.
- ISM: for the sources which use the Image Source Method, a shoebox room is defined using three sliders for length, width and height. The overall absorption of all the walls can also be defined as an absorption coefficient (percentage of energy absorbed by the wall).

4. Source Spatialization

This panel setup the spatialization parameters for sound sources. See Figure 11 for several examples of configuration with this subpanel. When *Apply to all sources* is enabled, the changes will be applied to every sound source in the diagram. Otherwise, they will be applied just to the selected one.

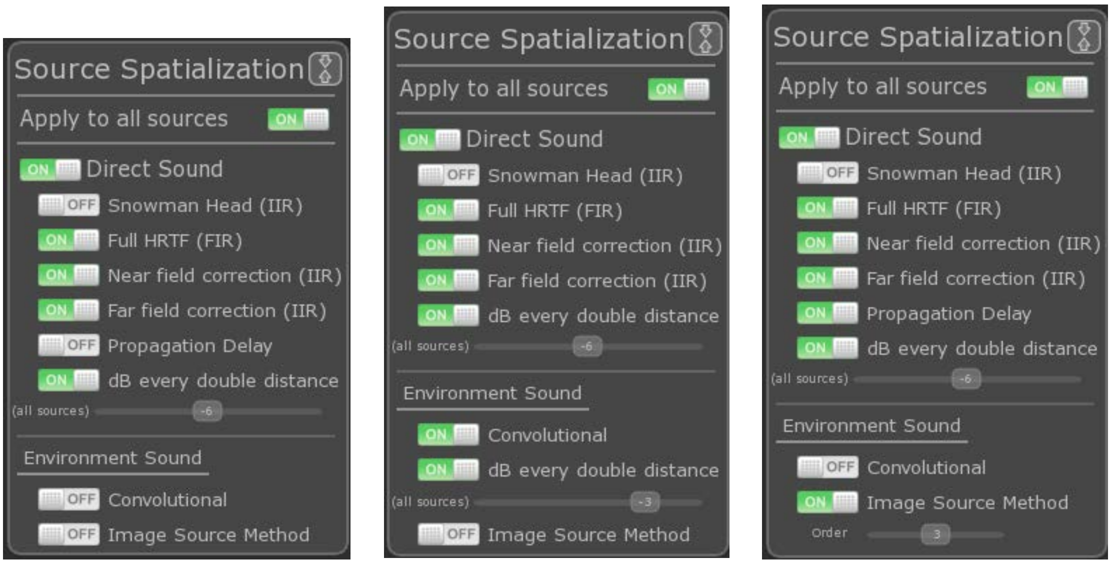


Figure 11. Source spatialization subpanel

- **Direct sound**. When the *Direct Sound* main button is switched off, all processes are switched off for the anechoic (direct) path. This includes both the direction simulation and the distance simulation.

  The *Snowman head (IIR)* and *Apply HRTF (FIR)* toggles are mutually exclusive and allow to select one of the two spatialisation methods provided by the application. In High Quality mode (*Apply HRTF (FIR)* enabled*)*, spatialisation is achieved through convolution with detailed HRTF models of the user head. In high performance mode (*Snowman head (IIR) enabled)*, the listener head is approximated with an ILD filter which cost (in terms of CPU and memory usage) is much lower, at the cost of a decreased spatialisation quality.

  The *Head Circumference* button allows switching On or Off the ITD simulation, which is based on the user head circumference. When switched on, the system does not use the ITD data stored in the HRTF and instead it calculates an ITD based on the listener's head size (configured with the slider below) and add the resulting delay to the signal after HRTF convolution (or ILD filtering, in High Performance mode). In High Performance mode, there is no ITD simulation unless this option is switched on. It's important to remark that ITD stored in HRTF can only be overridden when it is stored in the corresponding delay field in the SOFA or 3DTI file. In case it is just a delay inserted as part of the impulse response in every HRIR, it will not be removed.
  With all these options, you can configure the system to have anechoic spatialisation based on:
  - a measured HRTF (High Quality mode with HRTF from file) or
  - a measured HRTF customized to his/her head (High Quality mode with HRTF from file + ITD simulation) or
  - a ILD simulation (High Performance mode) or
  - a ILD simulation and ITD simulation customized to his/her head (High Performance mode + ITD simulation).

  Regarding the Distance simulation section of this subpanel, the *Near field correction (IIR)* switch enables/disables a frequency-dependent ILD effect simulation using bi-quad filters, which applies to any audio source when its distance to the listener is smaller than 1.95 meters. This switch is not available when the Snowman Head model is used.

The *Far distance (LPF)* switch enables/disables a low pass filter in order to simulate the far-away field effect. This effect applies to any audio source when its distance to the listener is bigger than 15 meters.

The Propagation Delay switch enables the simulation of the propagation delay depending on the distance between the source and the listener. When the propagation delay simulation is activated, the line between the source and the listener in the source chart is drawn in green. Otherwise it is drawn in white. This switch is not available when the convolutional reverb is used and, in that case, the propagation delay is not simulated

The *dB every double distance* switch enables/disables the attenuation with distance effect for the anechoic path, which applies to any source at any distance. The slider bellow allows to set for all sources the attenuation in decibels with double distance due to the sound propagation medium.

- **Environment Sound.** The Binaural Test App also implements environment (reverb) simulation using two techniques: (1) based on the convolution of the input signals coming from anechoic spatialisation with the binaural impulse response of a room (BRIR); and (2) using the Image Source Method, where a set of virtual sources are created in recursive mirror locations for each wall.

  The switch *Convolutional* allows switching On or Off the reverb rendered using the convolution, in the first order Ambisonics domain, with the Binaural Room Impulse response (BRIR) following the configuration set in the Environment subpanel. When convolutional reverb is On, the propagation delay simulation of the direct sound is disabled, and the corresponding switch is hidden. At the same time, the *dB every double distance* switch is shown. It enables/disables the attenuation with distance effect for the reverb path, which applies to any source at any distance. The slider bellow allows to set for all sources the attenuation in decibels with double distance due to the sound propagation medium.

  The switch *Image Source Method* allows the activation of this technique to simulate early reflections. When it is activated, the slider below allows to select the maximum reflection order in the technique (maximum number of reflections considered). When the Image Source Method is used, the corners of the room are represented in the source chart. If, in addition, the sound is being played, the location or all the virtual sources is represented as a cloud of green dots, including their elevations.

  Convolutional and *Image Source Method* are mutually exclusive. They cannot be activated at the same time, but it is possible to deactivate both, producing only anechoic rendering.

> **Warning:** When the source spatialization configuration is controlled via OSC commands, all parameters are set for each source independently. Therefore, if the switch *Apply to all sources* is switched on, the state of the switches in this subpanel may not represent the actual state of all the sources

5. Source Control

The *Source control* subpanel, allows the user to adjust the position and the volume of the currently selected sound source. The position can be expressed in Cartesian coordinates or using spherical coordinates. In the case or Cartesian coordinates, the position is absolute while spherical coordinates are relative to the listener's position and orientation.

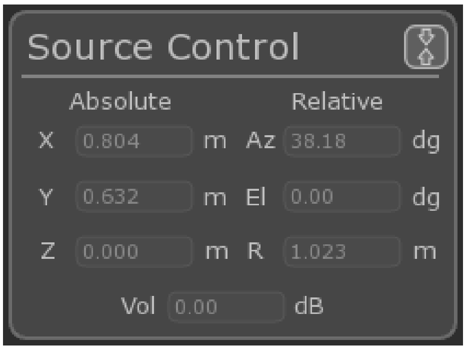


Figure 12. Adjustment of source position and volume

6. Output signal.

This subpanel shows the output signal wave graph and a level meter for each channel (left and right), as shown in Figure 13. The level meter indicates the digital level of the signal, with respect to internal full scale. When digital saturation happens, the sound meter will indicate it with red colour.

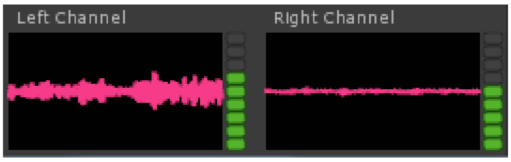


Figure 13. Output signal wave graph and level meter

Section 4: Source and listener layout chart

The layout chart is located at the centre of the Binaural Test App, as shown in Figure 14.

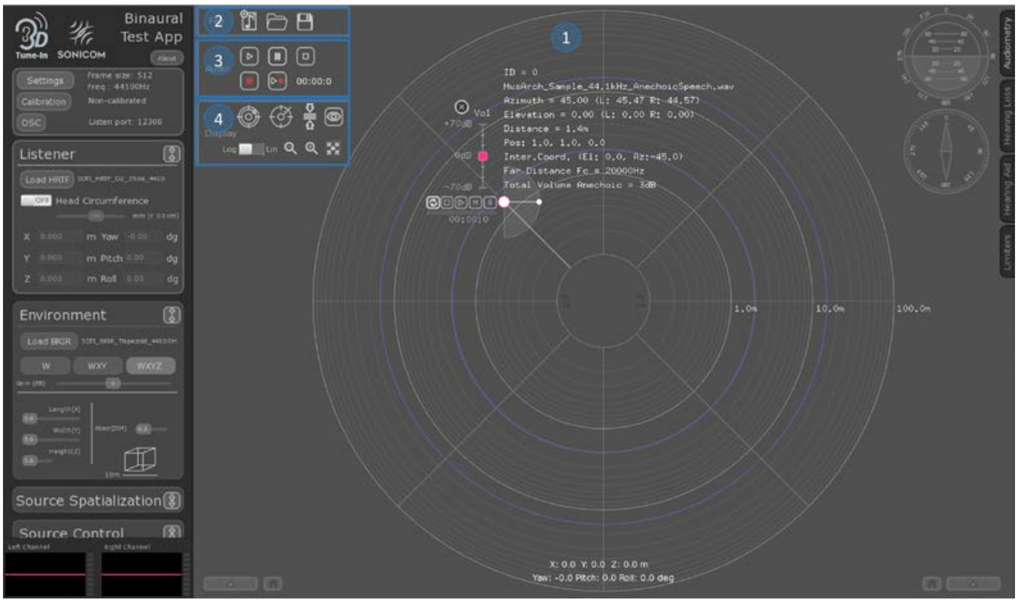


Figure 14. Sources and listener layout chart panel distribution

The different controls of the panel are listed below. The numbering corresponds to the numbers in Figure 14:

1. Source layout chart

The distribution of the sources and listener layout is shown in the Figure 15.

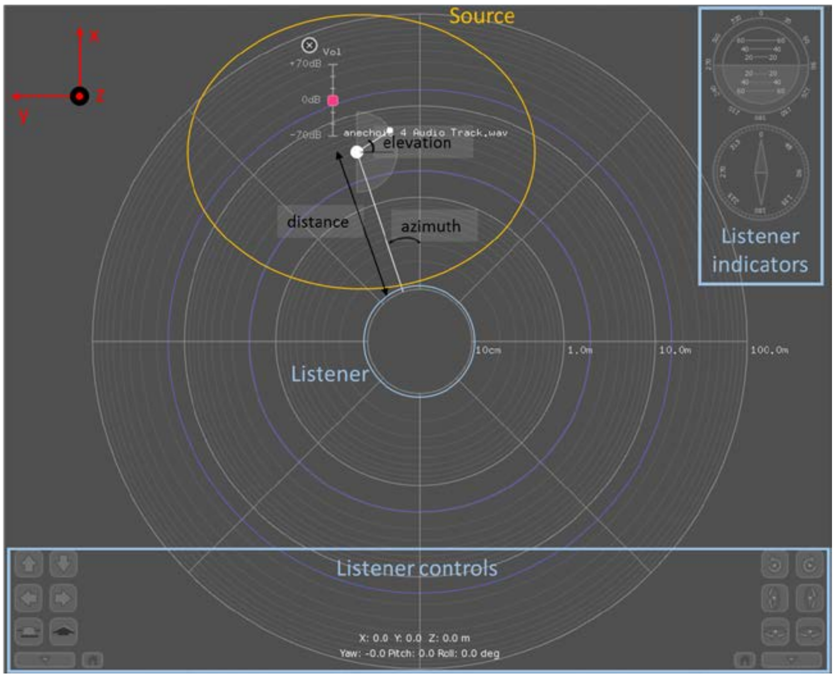


Figure 15. Sources and listener chart

The Cartesian coordinate-system of the virtual world is shown in the top-left corner of Figure 15. The listener is placed in the middle of the chart and the sources (represented with white circles) are placed around it. The remainder of this section explains how sources and listener are represented on the chart and how to move them.

Source representation

The chart shows the position of the sources relative to the listener position. Sources are represented with spherical coordinates decoupled, which means that the modification of one coordinate does not modify the representation of the other coordinates on the chart.

Each source is represented with two white dots (which turn to pink when audio is playing). The larger dot is connected to the listener with a white line (when no propagation delay is simulated) or a green line (when the propagation delay of that source is simulated). The angle between the connection line and the horizontal axis of the chart indicates the azimuth angle, from 0.0 to 359.9 degrees anti-clockwise. The smaller dot represents the elevation of the source. The angle between the line that connects the two white dots and the horizontal axis of the chart represents relative elevation of the source. The elevation varies between 0 and 90 degrees anti-clockwise (top-right quadrant), and clockwise from 360 to 270 (bottom-right quadrant).

Each source includes also a volume control (mark 1 in Figure 16) and a button (mark 2 in Figure 16) to remove it from the chart. A set of controls (mark 3 in Figure 16) to stop, play and pause the sound source are provided. A button to make the clip loop when reaching the end or not, as well as Mute y solo buttons are also available in the same group of buttons. Below them, a progress bar and the playing time for the clip are also displayed.

In addition, shown beside (mark 4 in Figure 16) each audio source is the information regarding the source ID, audio clip file name, source position (azimuth and elevation in degrees and distance in meters), source position in cartesian coordinates, inter-aural coordinates of the source (in degrees), cutoff frequency being applied by the far distance simulation effect (20KHz for not-far sources) and source volume for anechoic spatialisation, which depends both on the volume slider and on the attenuation of the sound which depends on the distance to the listener.

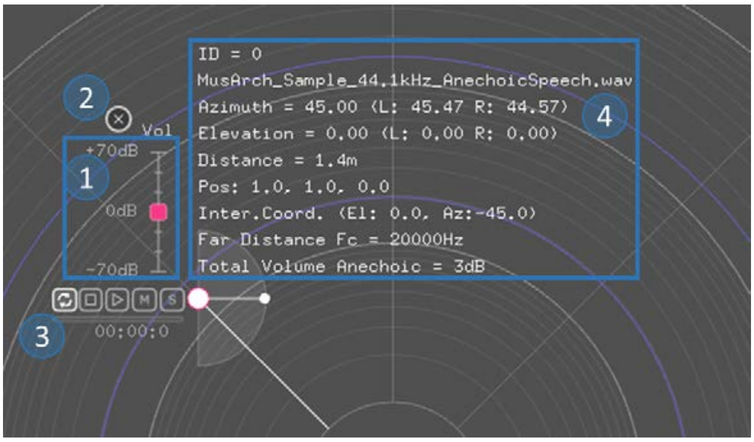


Figure 16. Sound source controls and indicators

The concentric circles of the chart serve as a ruler to measure the distance between the listener and the source. They could be in a logarithmic or linear scale. The blue circles indicate two significant distance values. The smaller blue circle indicates 2 meters, which is the reference at which the Toolkit considers that the original anechoic samples were recorded. The bigger blue circle indicates 15 meters, which is the threshold from which far distance simulation starts taking effect.

Listener representation

To move the listener use the controls at bottom corners of the central chart (Figure 15). The listener is always in the centre of the chart. When the listener position changes, the visual representation of all sources is moved in accordance with the new listener position, so that the relative position between the listener and the sources is always correctly represented on the chart.

With the initial listener orientation, the movements of the listener correspond to the following movements along the Cartesian coordinates:

Backward:	X positive
Forward:	X negative
Left:	Y positive
Right:	Y negative
Up:	Z positive
Down:	Z negative

Table 1. Listener movement axes with initial listener orientation

It is important to highlight that all listener movements are done over local coordinates. In practice it means that, once you change the orientation of the listener, the axes of movement may not coincide anymore with the world Cartesian coordinates as shown in Table 1.

The buttons at the bottom of the chart allow moving the listener, see Figure 17. The left group of buttons is used to change the listener position. From top to bottom: backward and forward movements, left and right movements, and up and down movements.

The right group of buttons is used to change the listener orientation. From top to bottom: yaw negative and positive movements (positive is clockwise seen from above the listener), pitch negative and pitch positive movements (positive is going up, or clockwise as seen from the listener's left) and roll negative and positive movements (positive is clockwise as seen from behind the listener).

The *home* buttons are used to reset the listener position and orientation to zero (the initial position and orientation). Between the two groups of controls there is a display where the position of the listener (in world coordinates) and the orientation (yaw, pitch and roll values relative to the listener initial position and applied in that order) are indicated.

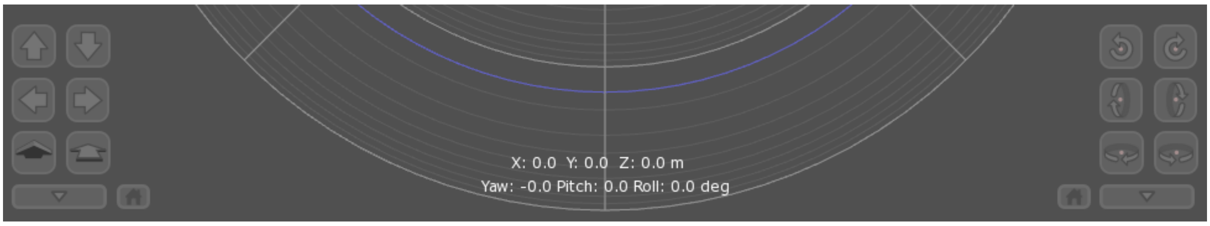


Figure 17. Listener position and orientations controls and indicators

Finally, at the top-right corner, two indicators show the listener orientation, see Figure 18. The upper indicator is an inclinometer that indicates the listener's pitch and roll angles. The lower one is a compass that represents the listener's yaw angle.

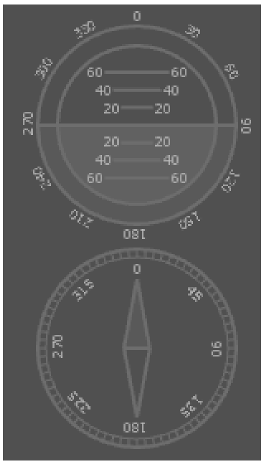


Figure 18. Listener orientation indicators

2. Files load and save files

To add a sound source or load and save a specific scenario (layout of sources in the chart), the interface offers the following buttons, grouped in the Files subpanel:

Figure 19. Scenarios subpanel

The first button allows the user to load a new audio file and add it to the scenario. The file must have 16 or 24 bits, 44100Hz 48000Hz or 96000Hz sample rate WAV format without compression.

The button in the middle allows users to load a previously saved scenario.

The right button allows users to save the current scenario.

Scenarios are stored as XML files. Various examples of these files are provided with the installation in the *Configurations* folder.

3. Audio clip control

To load, play and record sound sources use the following buttons, grouped in the *Audio* subpanel:

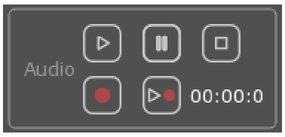


Figure 20. Audio subpanel

The First button starts playing all sources that have been loaded into the scenario. The second button pauses all the sound sources.

The Third button stops play. When playing again after stop, the audio clips will start from the beginning.

The buttons in the row below are for recording. The first button opens the dialog window shown in Figure 21 to start an offline recording. The offline recording allows to record a standard wav file of any

Duration, specified in minutes (*mm*) and seconds (*ss*) from the current scenario. The wav file will be stored in *Folder to save recorded files* with the configured *File name*. The *Resolution* of the recorded wav file can be set either to 16 or 24 bits per sample. When the user clicks on *Start*, the offline recording process will start and a dialog window will show the progress (in percentage) of the record until it reaches the specified duration. By default, the recordings will be stored in the folder specified in the *Settings* dialog window (see Section 3 – subpanel 2: Settings).

The offline recording functionality can be useful in two situations: when you want to record a long clip in only a few seconds, or when you want to record a clip which cannot be reproduced in real-time because the current scenario demands too many CPU resources from your system (in this case, the offline recording process will probably take longer than the specified duration).

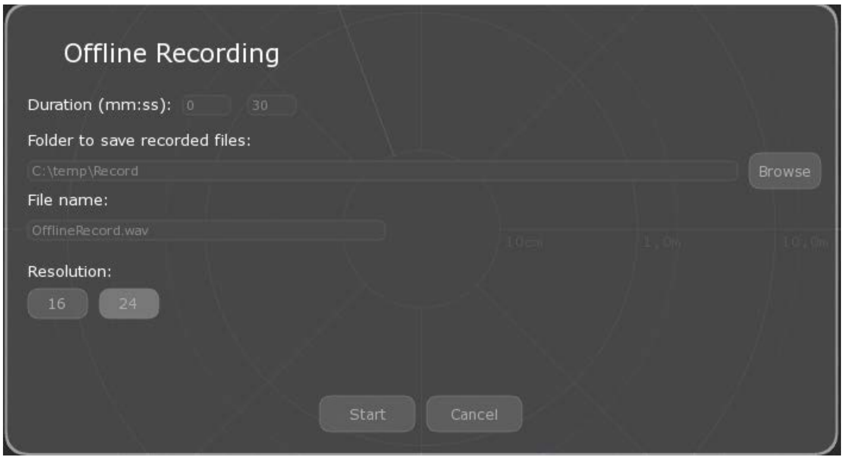


Figure 21. Offline recording dialog window

The last button of the *Audio* subpanel starts real-time recording. While offline recording is not interactive and records the output of a fixed scenario (without the ability to move the sources or change any settings in the middle of the recording), the real-time recording allows to change all settings and source positions while recording. The real-time recordings will be automatically stored in the folder configured in the Settings dialog window, following the specified file name pattern and resolution (see Section 3 – subpanel 2: Settings). When real-time recording starts, the recording progress is shown in the time counter at the right of the real-time recording button, and this button will change its appearance to show a Pause rather than a Play symbol. When recording is paused using the real-time recording button, play will continue without recording to wav. On the other hand, if recording is paused using the standard pause button, play will also be paused as usual.

Warning: some operating systems includes compression in their default audio drivers that can reduce or even avoid distortion due to high volume in the played audio. However, as compressing is not applied during the recording process, distortion could be present in the recorded wav file.

4. Display options

The Display subpanel panel shows different options related with the layout display:

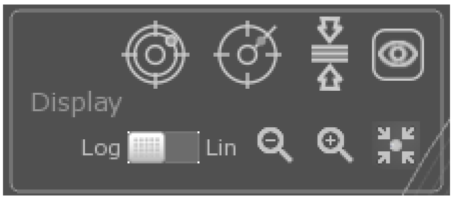
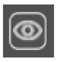


Figure 22. Display subpanel

 The first button in the first row constraints the distance from source to listener when moving sources. When this button is enabled, you can move the sources around the listener (changing its azimuth and elevation), but not change the distance.

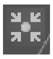
 The second button in the first row constraints the azimuth angle between source and listener when moving sources. When this button is enabled, users will be able to change the elevation angle and the distance to the listener, but not to change the azimuth angle.

 The third button in the first row collapses the information visualized by the sources, which can be handful when the chart is populated with a lot of sources

 The fourth button in the first row hides/shows the virtual sources when the Image Source Method is being used.

The second row contains a switch to choose whether the polar chart is in logarithmic or linear scale.

 When a linear scale is selected, two buttons allow to zoom in or zoom out. These buttons have no effect in logarithmic scale.

 Finally, the last button brings all the sources to a known distance which ensures that all of them are visible in the chart. This distance is 10m in the logarithmic scale, and 60m, 6m and 60cm in the linear scale, depending on the zoom level.

Section 5: Audiometry configuration panel

The panel for configuring the Audiometry is located in the right column of the Test App and is shown in Figure 23. This panel is split into two columns. The link button shown between the two columns allow linking the configurations for both ears (left and right). If the button image is , each ear is configured separately, if the button image is all configuration changes will affect both ears simultaneously. If a specific configuration is set for each ear and the user links the two configurations, the values of the right ear will be matched to the values of the left ear. The top right of the panel shows a control for setting the placement of the left and right ear controls. In Listener View mode, the left ear (in blue) controls are placed in the left column of the panel and the right ear (in red) controls in the right column. In Audiologist View mode, the left ear controls are shown in the right column and the right ear controls in the left column, as it is the usual practice in the representation of audiometries.

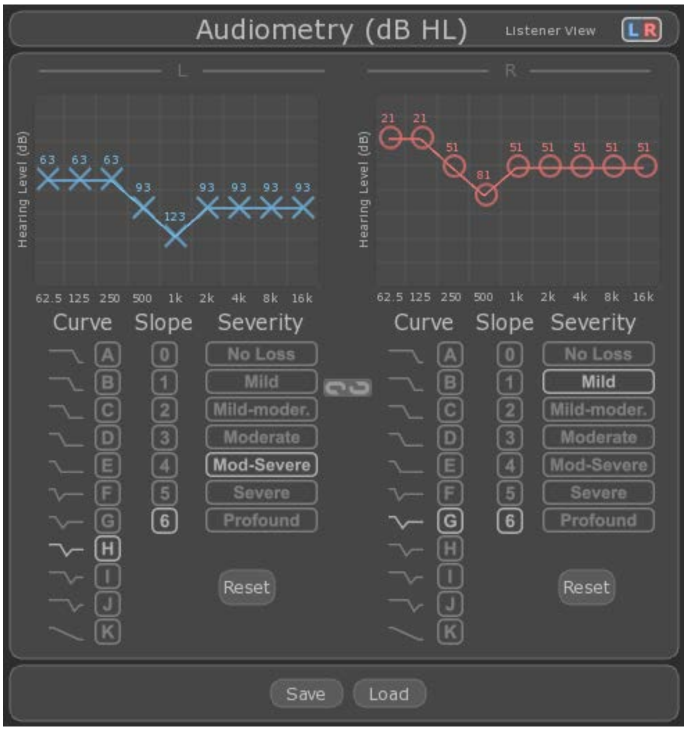


Figure 23. Audiometry Panel distribution

This panel shows an audiometry for both ears. The audiometry for each ear consists of a set of hearing loss levels for 9 bands (62.5Hz, 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 8KHz, 16KHz), in dB HL units. The levels can be dragged from the points in the audiometry as if they were sliders. The value in dB HL is shown above each point in the audiometry.

The buttons below the audiometry allows the user to choose between sets of curves, slopes and severities of the impairment. In addition, each mark on the audiometry can be dragged to customize the audiometry. The reset button, set the values of every band to zero.

Save and Load buttons allows the user to create an xml file to keep the current settings. These two buttons do also appear in hearing loss and hearing aid panels.

Section 6: Hearing Loss configuration panel

The panel for configuring Hearing Loss Simulation is located in the right column of the Test App and is shown in Figure 24. This panel is split into four rows and two columns. Rows group different subsets of configuration options. The rows can be collapsed or expanded by clicking the corresponding and buttons, in order to fit all desired controls in your screen. The link button shown between the two columns allow linking the configurations for both ears (left and right). If the button image is , each ear is configured separately, if the button image is all configuration changes will affect both ears simultaneously. If a specific configuration is set for each ear and the user links the two configurations, the values of the right ear will be matched to the values of the left ear. The top right of the panel shows a control for setting the placement of the left and right ear controls. In Listener View mode, the left ear (in blue) controls are placed in the left column of the panel and the right ear (in red) controls in the right column. In Audiologist View mode, the left ear controls are sown in the right column and the right ear controls in the left column, as it is the usual practice in the representation of audiometries.

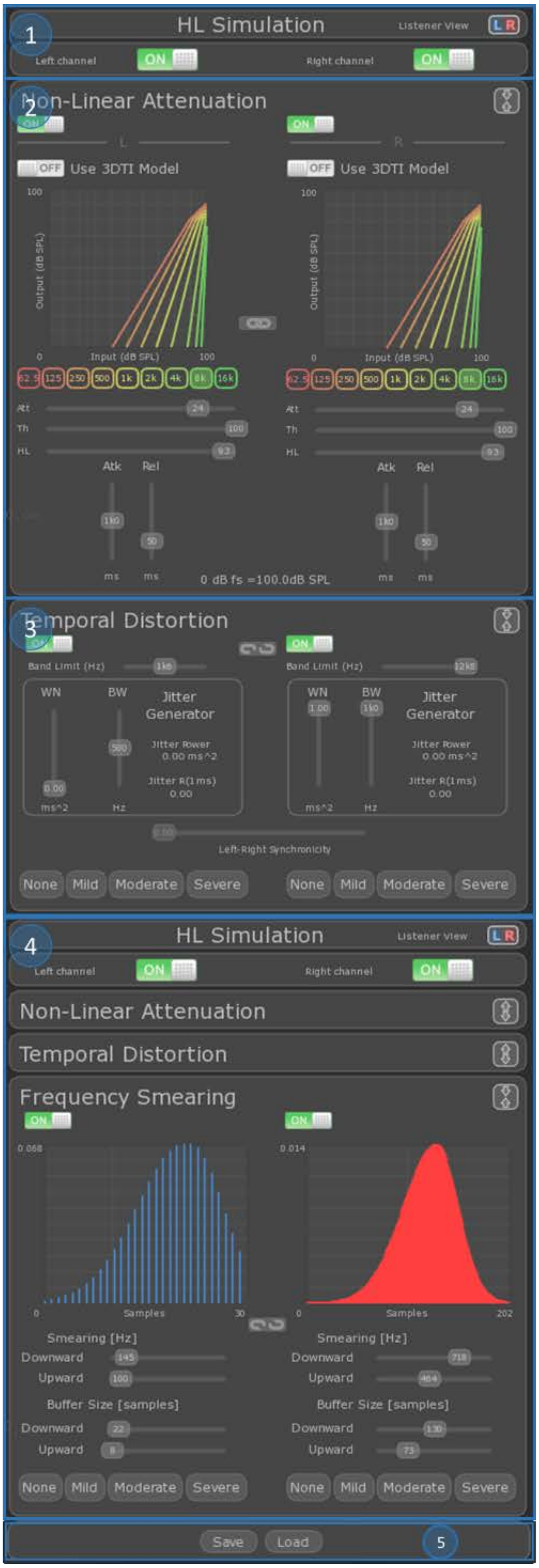


Figure 24. Hearing Loss Panel distribution

The different controls of the panel are listed below. The numbering corresponds to the numbers in Figure 24:

1. Global effect switches

This subpanel allows switching On/Off the global Hearing Loss simulation effect for each ear.

2. Non-linear attenuation

This subpanel, shown in Figure 25, allows the user to customize the model used for non-linear attenuation in HL simulation. The non-linear attenuation HL simulator consists of a multiband dynamics expander with an additional attenuation for each band. The multiband expander (threshold and ratio for each band) and the attenuations can be automatically configured from the audiometry data based on a model extracted from the work of Rasetshwane et al.[2] (called the 3DTi Model), but the application allows also for fine adjustment of each parameter of the simulator, overriding the 3DTi Model.

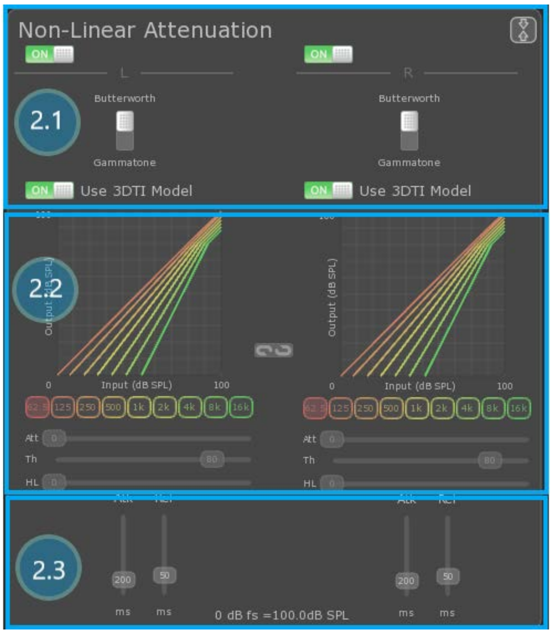


Figure 25. Non-linear attenuation subpanel distribution

In section 2.1, the first controls allow switching On and Off the non-linear attenuation process. Then, a couple of switches allow to select the type of filters used in the filter bank.

Selecting “Butterworth” will configure the Toolkit to use a set of 27 second order Butterworth filters (BW: 1/3 octave), and group them in nine groups (three filters per group). Each group corresponds to one expander which corresponds to each octave.

[2] Rasetshwane, D. M., Trevino, A. C., Gombert, J. N., Liebig-Trehearn, L., Kopun, J. G., Jesteadt, W., Gorga, M. P. (2015). “Categorical loudness scaling and equal-loudness contours in listeners with normal hearing and hearing loss”. The Journal of the Acoustical Society of America, 137(4), 1899. doi:10.1121/1.4916605

On the other hand, selecting “Gammatone”, a set of 42 Gammatone filters distributed according to the auditory filters in the cochlea. These filters are grouped into 9 groups (one per octave) and associated to an expander, as in the previous case.

Finally, the *Use 3DTi Model* controls for each ear allow, respectively, to switch On and Off the 3DTi Model for that ear, which configure automatically the expanders from the audiogram, using the 3DTI model already mentioned. Setting one of these controls to Off will override the automatic configuration of the HL simulator from the audiometry data and, instead, all controls in section 2.2 will be activated for full customization of the HL model.

Section 2.2 shows, from top to bottom for each ear, the following controls: first, a graphical representation of the multiband expander and attenuation for all bands is shown in a coloured graph. This graph shows the dynamics curve for each band with a different colour (from red for lower bands to green for higher bands). Each curve shows the input level in dB SPL and the output level produced by the HL simulator in dB SPL.

Below the graph, a set of 9 buttons allows the user to select the band to customize. After one band is selected with these buttons, the *Att*, *Th* and *HL* sliders will set, respectively, the attenuation, expander threshold and hearing level for that band (the ratio of the expanders will depend on the threshold and hearing level settings and can be seen in the graph as the slope for the curve of that band).

Figure 26 shows an example of HL customization with exaggerated values that are not usually found in real hearing loss, but are useful for didactic purposes. In the example, all bands except two correspond with normal hearing and thus their curve is perfectly linear and starting at 0. The red (62.5 Hz) band has a high attenuation value (23 dB), a hearing level of 47 dB (the crossing of the red curve with the horizontal axis) and expander threshold in 82 dB (the point in which the curve is split in two segments). The green curve (16 KHz) has a small attenuation (4 dB) and values on threshold and hearing level which are pretty close (62 dB and 56 dB, respectively), causing the expander of that band to present a high ratio, close to that of a noise gate.

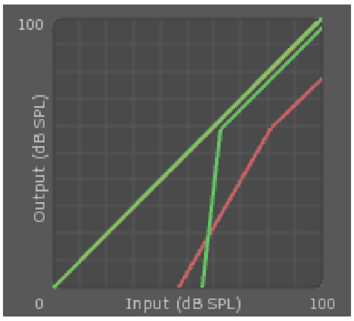


Figure 26. Example of HL simulation customization

Section 2.3 shows controls to configure the envelope detectors associated to the multiband expander. Since the 3DTi Model does not configure the attack and release of the envelope detector, they can be both configured, using the *Att* and *Rel* sliders for each ear, regardless the 3DTi Model is being used or not. Between these sliders, the calibration setting is written for convenience (for example: 0 dB fs = 100.0dB SPL) as a reminder while configuring the HL simulation.

3. Temporal distortion

This subpanel, shown in Figure 27, allows the user to customize the model used for temporal distortion simulation. Temporal distortion is a potential symptom of age-related sensorineural hearing loss, where cortical speech processing may be limited by age-related decreases in the precision of neural synchronization in the midbrain. The temporal distortion simulator consists of a jitter generator for applying random displacements of low-frequency samples.

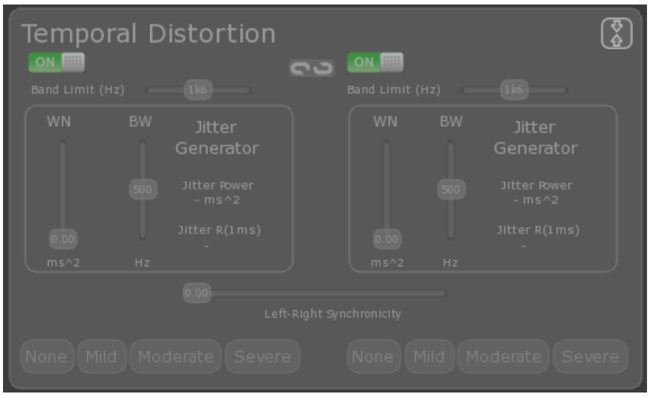


Figure 27. Temporal distortion subpanel

The first controls allow switching On and Off the temporal distortion process. The *Band Limit* control allows setting the frequency band where temporal distortion is applied; since temporal distortion is always applied over low frequencies, this parameter refers to the maximum frequency (in Hz) that will be affected by this effect (in other words, the cutoff frequency of a high-order low-pass filter used to split the signal ant get only the low frequencies).

Inside the *Jitter Generator* section: the *WN power* control allows setting the power of the Gaussian white noise used as jitter source (since the mean is 0, it refers to the standard deviation of the Gaussian distribution of the noise); the *BW* control allows setting the bandwidth of the jitter noise source (the source white noise goes through a low-pass filter for autocorrelation, and this control refers to the cutoff frequency of the autocorrelation filter); The *Jitter Power* indicator shows in real-time the power of the noise source (before autocorrelation), while the *Jitter R(1ms)* indicator shows in real-time the autocorrelation of the noise source after 1 millisecond (which depends on the settings of the low-pass autocorrelation filter).

The *Post-Jitter LPF* control allows switching On and Off a low-pass filter after the low frequency band goes through the jitter process, to ensure that no high-frequency artefacts have been introduced by the process before restoring the original signal by mixing with the unaffected high frequency band (actually, this high band is unaffected, except for a delay added to be coherent with the delay introduced by the jitter process in the low frequency band).

The link button shown between the two columns allow linking both configurations and both temporal distortion processes. If the button image is , each ear is configured separately, if the button image is  all configuration changes will affect both ears simultaneously and the jitter noise sources used for each ear can be correlated. When the two ears are linked, the *Left-Right Synchronicity* control is enabled, allowing to set the amount of correlation between the jitter noise sources of both ears; a setting of 0 means no correlation (independent jitter noise sources for each ear) while a setting of 1 means full correlation (the same jitter noise source used for both ears).

Some presets are available through the set of buttons labelled as None, Mild, Moderate and Severe for each ear to model different levels of hearing loss.

4. Frequency smearing

This subpanel, shown in Figure 28, allows the user to customize the model used for frequency smearing simulation. Frequency smearing is a potential symptom of age-related sensorineural hearing loss, which consists in the broadening of the auditory filter shapes. The frequency smearing simulator consists of a non-linear process of convolution in frequency domain with a configurable smearing window.

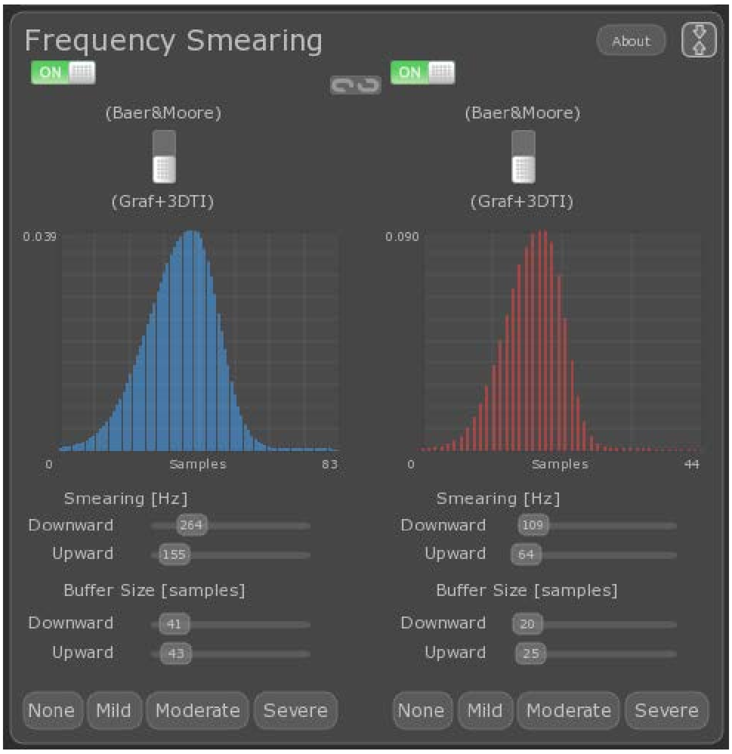


Figure 28. Frequency smearing subpanel

The first controls allow switching On and Off the frequency smearing process. Then you have a couple of switches to choose for each ear one of the two implemented models for frequency smearing. "

- (Graf+3DTI) is a re-elaboration of the implementation presented in: Badri, R., Siegel, J.H., Wright, B.A. (2011). Auditory filter shapes and high-frequency hearing in adults who have impaired speech in noise performance despite clinically normal audiograms. The Journal of the Acoustical Society of America, 129(2), 852-63.
- (Baer&Moore) implementation (Baer, T., & Moore, B.C. (1993). Effects of spectral smearing on the intelligibility of sentences in noise. The Journal of the Acoustical Society of America, 94(3), 1229-1241), which has been directly translated in C++ from the original Matlab code, courtesy of Michael Stone @ University of Manchester.

If the Graf+3DTI option is selected, below the mentioned controls, a graphical representation of the smearing window is shown. The smearing window is normalized to have a total area of 1, and the maximum value (value at mean) after normalization is shown in the y-axis. The x-axis represents each sample in the smearing window and the total number of samples depends on the Buffer Size parameters, which will be described later.

- The *Smearing [Hz]* controls allow setting the amount (in Hertzs) of smearing effect in Downward and Upward directions. Downward smearing amount represents how much each frequency is

smeared by (affected by) lower frequencies, while Upward smearing amount represents how much each frequency is smeared by (affected by) higher frequencies. These controls change the graphical representation of the smearing window accordingly.
- The *Buffer Size [samples]* controls allow setting the number of samples reserved for the Downward and Upward directions in the smearing window buffer. Although the effect is noticeable only with 1 sample for each direction, the effect gets richer (but more computationally expensive) with bigger buffer sizes.
- The *Broadening Factor* (or BF) will not be able to change because it is not needed for this algorithm.

If Baer&Moore Algorithm is selected:

- The *Smearing [Hz]* and *Buffer Size [samples]* controls will not be able to change because it is not needed for this algorithm.
- The *Broadening Factor* (or BF) controls allow setting the amount of smearing in Downward and Upward directions, used by Moore’s Algorithm.

Some presets are available through the set of buttons labelled as None, Mild, Moderate and Severe for each ear to model different levels of hearing loss for each smearing algorithm.

5. Save/Load

These buttons allow the user to save and load their hearing loss configuration in a XML file.

Section 7: Hearing Aid configuration panel

The panel for configuring Hearing Aid Simulation is located in the right column of the Test App and is shown in Figure 29. This panel is split into six rows and two columns. The rows group different subsets of configuration options. The rows can be collapsed or expanded by clicking the corresponding and buttons, in order to fit all desired controls in your screen. The link button shown between the two columns allow linking the configurations for both ears (left and right). If the button image is , each ear is configured separately, if the button image is all configuration changes will affect both ears simultaneously. If a specific configuration is set for each ear and the user links the two configurations, the values of the right ear will be matched to the values of the left ear. The top right of the panel shows a control for setting the placement of the left and right ear controls. In Listener View mode, the left ear (in blue) controls are placed in the left column of the panel and the right ear (in red) controls in the right column. In Audiologist View mode, the left ear controls are sown in the right column and the right ear controls in the left column, as it is the usual practice in the representation of audiometries.

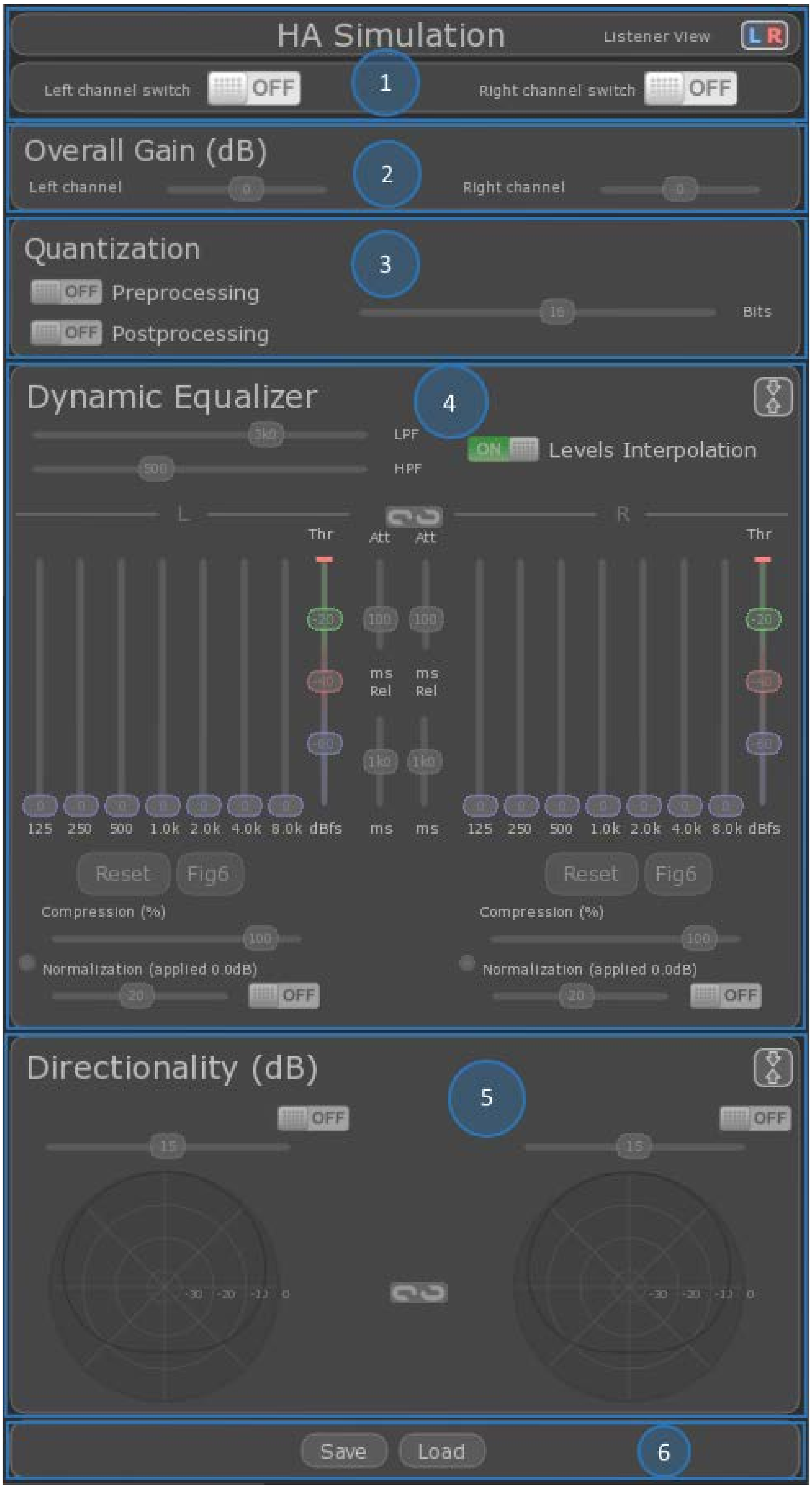


Figure 29. Hearing Aid Simulator panel distribution

The different controls of the panel are listed below. The numbering corresponds to the numbers in Figure 29:

1. On/Off controls

These switches control the overall hearing aid simulation process. When switched off, there will be no hearing aid simulation processes for that ear. When switched on, at least the Overall Gain and Dynamic Equalizer options will be available. Quantization Noise and Directionality processes have their own enabling switches.

2. Overall gain

These sliders allow setting a gain (attenuation or amplification) applied at the end of the hearing aid simulation process chain.

3. Quantization noise

These controls allow simulating the quantization noise produced by some low-end hearing aid devices. Quantization noise can be added at the beginning (*Quantization Before*) or at the end (*Quantization After*) of the hearing aid simulation process chain. The slider allows setting the number of bits of the hearing aid device (less bits will produce more quantization noise).

4. Dynamic equalizer

The 7-band dynamic equalizer has three different response curves (red numbers in Figure 30) linked to three different level thresholds (blue numbers in Figure 30).

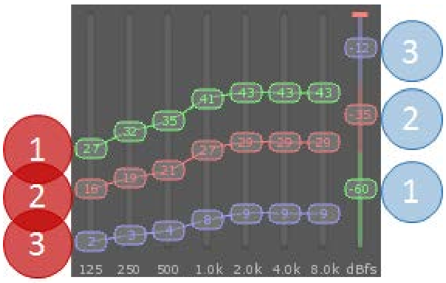


Figure 30. Dynamic equalizer

The equalization curve applied to the signal depends on the level detected by an envelope detector. Louder sounds (in the blue zone around blue number 3), are equalized following the third curve (red number 3), which usually implies less amplification. Sounds with moderate loudness (red zone around blue number 2) are equalized following the second curve (red number 2) and last, softer sounds (in the green zone around blue number 1) are equalized with the first curve (red number 1), which is usually the one with greater amplification.

The dynamic equalizer is a complex element, with many controls that can be better understood in groups as shown in Figure 31.

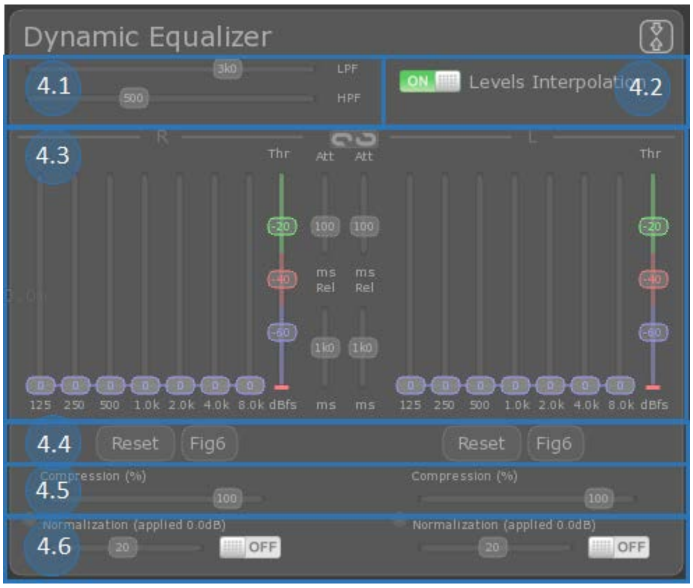


Figure 31. Dynamic equalizer subpanel of Hearing Aid simulator

3.1. Low pass and High Pass filters

These sliders allow setting the cutoff frequencies, in hertzs, of global low pass (*LPF*) and high pass (*HPF*) filters.

3.2. Level interpolation

When this switch is On, the actual equalization curve is interpolated between the two closest equalization curves depending on the current signal level detected by the envelope detector. For example, if the first threshold is at -20dB and the second threshold is at -40dB, a signal level of -30dB will be equalized with a response curve between the second and third curves set in the equalizer. This linear interpolation allows smooth transitions between the different signal level regions.

3.3. Band gains and envelope detector

These controls are the core of the dynamic equalizer. The seven bands of the equalizer have fixed centre frequencies, with one-octave increments starting at 125Hz (i.e. 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 8KHz). The gain for each band can be set using seven (for each ear) multi-handle sliders. Each multi-handle slider allows setting three different values: the gain for that band for the first curve (green handle), for the second curve (red handle) and for the third curve (purple handle). Interpolated curves are drawn between the handles for easy visualization of the three equalization curves.

The *Thr* multi-handle slider allows setting the three thresholds, in dBfs (0 dBfs means maximum signal level).

The *Att* sliders sets the attack time (in milliseconds) of the envelope detector for each ear. The *Rel* sliders sets the release time (in milliseconds) of the envelope detector for each ear.

3.4. Quick setting controls

Manual setting of each equalization curve (using the controls in group 3.3) is a tedious and complex procedure. These buttons allow for fast simultaneous setting of all band gains (all controls of group 3.3).

The *Reset* button sets all gains for all curves to 0dB. In this case, the dynamic equalizer will not do any amplification.

The *Fig6* button sets the three curves and the three thresholds following the Fig6[3] algorithm. The input data for Fig6 consists of the 7 central bands configured on the Hearing Loss Simulation panel.

3.5. Compression controls

These controls allow for a fast way of changing the amount of dynamic compression that naturally arises from the dynamic equalizer. When the slider is set to 100% compression, the equalizer curves follow exactly the values set with the controls in group 3.3. As the slider goes below 100%, the curves get closer until they collapse in a single curve at 0% compression. The curve used as reference is the second (red curve) and thus this slider controls how the other two curves (green and purple) converge towards the (usually central) red curve. The slider allows for values greater than 100% (up to 120%), what leads to an increased compression effect.

The sliders set the nominal value, but the actual gain applied is the one represented by the linear curves.

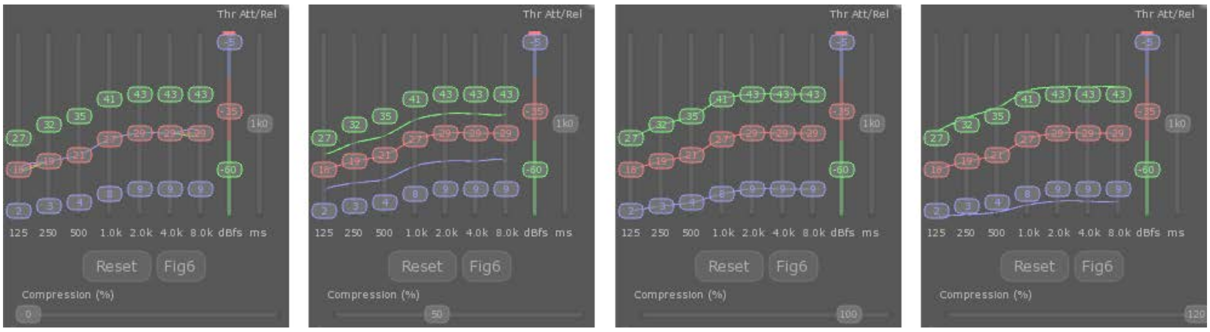


Figure 32. Dynamic equalizer curves displacement depending on compression percentage. From left to right: 0%, 50%, 100% and 120% compression.

The effect of compression can be seen in the curves drawn over the controls in group 3.3 (see Figure 32). Please, note that changing the gains for curves one and three (green and purple) will have no effect while compression is set to 0%.

3.6. Normalization controls

Although normalization is not a real feature of hearing aids, it is convenient for simulation, to avoid excessive amplification to go beyond full scale and to avoid hearing damage in users with no real hearing loss.

Normalization can be switch on and off and a slider allows setting the normalization level in decibels. The second curve (red curve) of the equalizer is used as reference for normalization. If any of the band gains of this curve goes above the normalization level, then this curve will be normalized so that its maximum gain equals the normalization level, applying an offset to all gains in all curves. If an offset is applied, the red led to the left of the *Normalization* label will lit and the applied offset, in decibels, will be shown to the right. For example, in Figure 33 an offset of -9.5dB is applied when setting normalization to 20dB level. When this normalization makes some curves to go below 0dB, they will

[3] Killion, M. C., and Fikret-Pasa S. "The 3 types of sensorineural hearing loss: Loudness and intelligibility considerations." Hearing journal 46 no11 (1993): 31-36.

stay at 0dB, avoiding negative gains (attenuation) in the HA. The effect of normalization can be seen in the curves drawn over the controls of group 3.3.

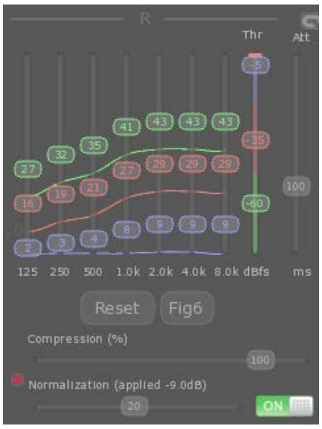


Figure 33. Dynamic equalizer curves displacement due to normalization

Please note that, although the same offset is applied to all curves, the first curve (green) has usually higher gains than those in the reference curve (red) and thus normalization will not guarantee that none of the gains within all curves is beyond the normalization level. Moreover, since it is not possible to predict the level of the input signal, normalization is just a quick way of lowering the output signal of the hearing aid rather than a guarantee to avoid clipping or excessive amplification.

All controls of the dynamic equalizer (except for the groups 3.1 and 3.2, which are shared by both ears) can be split or linked for both ears using the corresponding link button.

6. Directionality

These controls allow setting the directionality feature of directional hearing aids. This feature can be enabled/disabled using the corresponding switches of the panel. The slider allows setting the directionality extent, expressed as the attenuation in decibels applied to the sounds coming from the backward direction. The graph shows an approximate representation of the hearing aid microphones array directional response.

A setting of 0 corresponds with no directionality, showing an omnidirectional microphone response (Figure 34, Left). As the directionality extent growths, the response takes a cardioid shape, with progressive attenuation depending on the angle with respect to the front direction, or naso-occipital axis (Figure 34, Right).

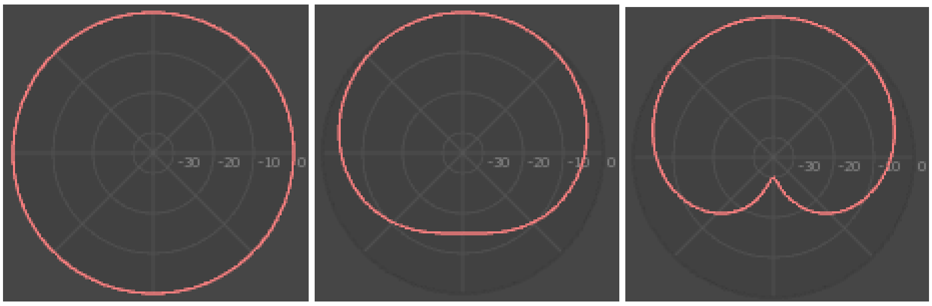


Figure 34. Microphone response depending on hearing aid directionality extent.
Left: 0dB or omnidirectional; Middle: subcardioid with 15dB maximum attenuation Right: cardioid with 30dB max. att.

The configuration of hearing aid directionality can be split or linked for both ears using the corresponding link button.

7. Save/Load

These buttons allow the user to save and load their hearing aid configuration in a XML file.

Section 8: Limiters configuration panels

The panel for configuration of Limiters is located in the right column of the Test App and is shown in Figure 35. This panel is split into three rows. The rows group different subsets of configuration options.

Figure 35. Limiters panel distribution

The different controls of the panel are listed below. The numbering corresponds to the numbers in Figure 35:

1. Limiter after spatialization

When switched on, this panel allows for control of a dynamic compressor applied to the signal after the spatialization (both anechoic and reverb) process. This compressor is intended to act as a dynamic limiter and thus high ratios and fast attack times can be set. The sliders set, respectively: attack time in milliseconds (*Att*), release time in milliseconds (*Rel*), threshold level in dBfs (*Thr*) and ratio (*Rat*). A led near the *Thr* label is lit whenever the signal of the envelope detector exceeds the threshold.

2. Limiter at the end of the process chain

When switched on, this panel allows for control of a dynamic compressor applied to the signal at the end of all process chain (spatialization, hearing aid and hearing loss). This compressor is intended to act as a dynamic limiter and thus high ratios and fast attack times can be set. The sliders set, respectively: attack time in milliseconds (*Att*), release time in milliseconds (*Rel*), threshold level in dBfs (*Thr*) and ratio (*Rat*). A led near the *Thr* label is lit whenever the signal of the envelope detector exceeds the threshold.

3. Clipping at the end of the process chain

When switched on, samples above 1 at the end of the process chain will be clamped to 1 while samples below -1 will be set to -1.