EU Horizon 2020 - Contract No 644051

3D Tune-In Toolkit features

3D-games for TUNing and lEarnINg about hearing aids

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, Daniel González-Toledo, Carlos Garre, Luis Molina-Tanco and Ernesto de la Rubia-Cuestas, all affiliated to the University of Malaga.

Contact: areyes@uma.es, l.picinali@imperial.ac.uk

Date: 28 April 2017

Version: 2.0

Index

Section 1: Introduction

This document presents the features provided by the binaural and loudspeakers 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 will be released as open source under GPLv3 license for non-commercial use. Contact developers for commercial use.

Third party libraries.

- 3D Tune-In Toolkit uses:
  - Takuya OOURA General purpose FFT http://www.kurims.kyoto-u.ac.jp/~ooura/fft.html
- 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). cereal - A C++11 library for serialization. URL: http://uscilab.github.io/cereal/)

The features of the Binaural spatialisation, HA and HL simulators can be tested using the Binaural Test App, available both for Windows and Mac OSX operating systems.

The Loudspeakers-based spatialisation features can be tested using the Loudspeakers Test App, available both for Windows and Mac OSX operating systems.

The features of the Resource management package can be tested with the HRTF_SOFATo3DTI and BRIR_SOFATo3DTI tools, available for Windows operating system.

Section 2: Features of Binaural spatialisation

	Features	Description
Basic features	Multiple frame sizesallowed	The Toolkit allows selecting different frame sizes to process the audio input signal for binaural spatialisation. Allowed frame sizes are: 64, 128, 256, 512, 1024, 2048, 4096 and 8192.
	Moving listener	The Toolkit allows modifying listener position and orientation at run-time.
	Multiple moving sources	The Toolkit allows creating and removing sources in run-time. It also allows modifying source positions and orientations at run-time.
	Geometric calculations support	The Toolkit provides the listener polar and inter-aural relative coordinates to each source. It also provides classes for handling convention-safe rigid transformations, including position and orientation (quaternion-based, with conversion from/to angle-axis and Tait-Bryan).
	Physical magnitudes	The Toolkit provides a class to handle physical magnitudes such as distance attenuation factors and sound speed.
	Audio buffer handling	The Toolkit provides a class with methods and operands for basic handling of audio buffers.
Binaural Anechoic path simulation	Real time audio source spatialisation	The Toolkit implements algorithms for the convolution between audio source data and HRIR filters. The Toolkit allows enabling and disabling the HRTF simulation for each source, individually.
	HRTF selection	The Toolkit allows loading HRTF data. The data could be read using the Resource management package (see Section 5).
	HRIR interpolation with barycentric coordinates	The Toolkit implements offline HRIR calculation and storage (including the Interaural Time Difference) using a resampling algorithm with a configurable step based on barycentric interpolation. The Toolkit also implements On-line HRIR interpolation using barycentric coordinates with 1 degree of resolution.
	Efficient convolution with HRTF	The Toolkit implements fast convolution algorithms with uniform partitioning of the signal and the filter impulse response.
	ITD customization based on listener head circumference	The Toolkit allows users to select their head circumference size and to calculate their individualized HRIR Interaural Time Difference (ITD).
	Distance simulation of simple attenuation	The Toolkit allows simulating simple attenuation for every double distance. The attenuation value is configurable (set to 6dB by default). The Toolkit also allows enabling or disabling this distance simulation effect individually for each source.

	Distance simulation of near field effects	The Toolkit allows simulating a near field filter effect. This consist on a frequency-dependent ILD effect simulation using biquad filters where coefficients are read from a lookup-table. The data for the lookup-table could be read using the Resoure management package (see Section 5). This distance simulation effect can be enabled or disabled individually for each source.
	Distance simulation of far distance effects	The Toolkit allows simulating a far-away field effect using a low pass filter, with variable cutoff frequency inversely proportional to the distance. This distance simulation effect can be enabled or disabled individually for each source.
Binaural Reverb path simulation	Real time audio source reverb simulation	The Toolkit implements algorithms to carry out the convolution between audio source data and Binaural Room Impulse Response (BRIR) filters.
	Virtual Ambisonic approximation	The Toolkit implements an algorithm to make the reverb convolution process independent on the number of audio sources. This algorithm is based on measuring the direct impulse response between the B-Format of a first order Ambisonic approximation and the stereo binaural output. The output signal of the Ambisonic simulation is convolved with the computed Ambisonic Binaural Impulse Response (AmbiBinIR).
	BRIR selection	The Toolkit allows loading BRIR data. The data could be read using the Resource management package (see Section 5).
	Efficient convolution for very large BRIRs (for very long reverb effects).	The Toolkit implements fast convolution algorithms that use a uniform partitioning of the filter impulse response according to the buffer size.
	Distance simulation for the reverb effect.	The Toolkit allows simulating simple distance-dependent attenuation for the reverb effect, independent of the attenuation of the direct (anechoic) path between sound source and listener. The attenuation value is configurable (set to 3dB by default). Furthermore, the Toolkit allows enabling or disabling this distance simulation effect individually for each source.

Section 3: Features of Hearing Aid (HA) and Hearing Loss (HL) simulators

	Features	Implementation details
HA simulation	Directionality simulation	The Toolkit allows processing directionality with adjustable cardioid microphone response.
	Dynamic multiband equalization filter	The Toolkit provides a dynamic multiband equalization filter. This filter consists of an envelope detector with a set of configurable level thresholds and a set of equalization curves with an adjustable number of bands, initial frequency and octave step. The applied equalization curve depends on the signal level of the envelope detector and the configured thresholds, allowing for linear interpolation of the equalization curves. Each band consists in a bi-quad filter with adjustable gain and Q. Two additional filters are included for low pass and high pass filtering. Global compression percentage and normalization can be applied to the equalization curves.
	Quantization noise simulation	The Toolkit provides simulation of the quantization noise added by digital hearing aid devices, with adjustable number of bits and adjustable position in the HA simulation process chain (at the beginning/at the end).
	Fig6 algorithm	The Toolkit implements the Fig6 algorithm for fitting the dynamic multiband equalization filter using hearing loss data as input.
	Overall gain per channel	The Toolkit allows setting a global attenuation or gain for each channel.
HL simulation	Multiband equalization filter	The Toolkit provides a multiband equalization filter with adjustable number of bands, initial frequency and octave step. Each band is a bi-quad filter with adjustable gain and Q.
	Dynamics compressor with adjustable envelope detector	The Toolkit provides methods to set threshold, ratio and AR envelope for processing dynamic compression on each ear. The dynamics compressor is serially chained with the equalizer, allowing two different configurations (compressor first, or equalizer first)

Section 4: Features of Loudspeaker-based spatialisation

	Features	Implementation details
Basic features	Multiple frame sizes allowed	The Toolkit allows selecting different frame sizes to process the audio input signal for loudspeakers spatialisation. Examples of values allowed of frame size are: 64, 128, 256, 512, 1024, 2048, 4096 and 8192.
	Multiple moving sources	The Toolkit allows creating and removing sources in run-time. It also allows modifying sources position and orientation in run-time.
	Loudspeakers physical configuration	The Toolkit allows using different loudspeakers configurations using an Ambisonic decoding through projection, which means that it is exact only for layouts where loudspeakers are in the vertex of a regular (Platonic) solid.
	Geometric calculations support	The Toolkit provides the listener polar and inter-aural relative coordinates to the source. It also provides classes for handling convention-safe rigid transformations, including position and orientation (quaternion-based, with conversion from/to angle-axis and Tait-Bryan)
	Audio buffer handling	The Toolkit provides a class with methods and operands for basic handling of audio buffers.
Loudspeakers Anechoic path simulation	Real time audio source spatialisation	The Toolkit implements algorithms to do the audio sources spatialisation through a virtual Ambisonic approach.
	Virtual Ambisonic transformation	The Toolkit implements second-order encoder and decoder Ambisonic transformations.
	Distance simulation of simple attenuation	The Toolkit allows simulating simple attenuation for every double distance. The attenuation value is configurable (set to 6dB by default). Furthermore, the Toolkit allows enabling or disabling this distance simulation effect individually for each source.
	Distance simulation of far distance effects	The Toolkit allows simulating a far-away field effect using a low pass equalizer filter, with variable cutoff frequency inversely proportional to the distance. This distance simulation effect can be individually enabled of disabled for each source.
Loudspeakers Reverb path simulation	Real time audio source reverb simulation	The Toolkit implements algorithms to carry out the convolution between audio source data and Ambisonic Room Impulse Response (ARIR) filters.
	Virtual Ambisonic approximation	The Toolkit implements an algorithm to make the reverb convolution process independent on the number of audio sources. This algorithm is based on measuring the direct impulse response between the B-Format of a first order

		Ambisonic approximation and the anechoic spatialisation output. The output signal of the Ambisonic simulation is convolved with the Ambisonic Impulse Response (ARIR).
	ARIR selection	The Toolkit provides methods to store and manage ARIR data, coming from external resources.
	Efficient convolution for very large A-RIR.	The Toolkit implements fast convolution algorithms that use a uniform partitioning of the filter impulse response according to the buffer size.
	Distance simulation for the reverb effect.	The Toolkit allows simulating simple distance-dependent attenuation for the reverb effect, independent of the attenuation of the direct (anechoic) path between sound source and listener. The attenuation value is configurable (set to 3dB by default.)The Toolkit also allows enabling or disabling this distance simulation effect individually for each source.

Section 5: Features of 3D Tune-in Resource management package

	Features	Implementation details
SOFA format	SOFA files reader	The Toolkit implements methods to read HRTF and BRIR data from SOFA (Spatially Oriented Format for Acoustics) format files.
3DTI format	Conversion from SOFA to binary 3dti format	The Toolkit provides different tools to allow converting SOFA files with HRTF or BRIR into 3dti format files. 3dti files are binary files providing easy portability of data between the different platforms supporting the Toolkit and its wrappers.
	3DTI files reader	The Toolkit implements methods to read HRTF, BRIR and ILD data from 3dti format files.