A new Audacity feature: room objective acustical parameters calculation module

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Abstract

Audacity is a popular and powerful platform for audio editing. It has many useful features, a flexible plug-in subsystem, and is multiplatform too. The latter characteristic greatly facilitates the development of portable acoustic applications that will work regardless of the system employed by the final user. This conducted us to start the porting of Angelo Farina's Aurora plug-in family for *Adobe Audition*®to Audacity, beginning with the *Acoustical Parameters* module. In this article we describe this development work and the results obtained.

Keywords

Audacity, Acoustical parameters, Impulse response,

1 Introduction

During the evolution of the architectural acoustic science it became clear there was a need for a set of parameters describing the *acoustic* behavior of a room. It is well known that human sound perception is very subjective, but the design of an auditorium, for example, shouldn't be based on a subjective basis. An auditorium, or a theatre, should be a place where people can have a good perception of the concert, the show or whatever is performed there. To reach this very difficult goal a designer needs *objective* data to guide the design of room structures — these data are called *objective acoustical parameters* charachteristic of a particular room. Many parameters have been studied, from the 1950s up to today. It is impossible to ignore the great work done by Leo Beranek in this field: one of the results of his research is a set of parameters, releated to the reverberation time, that describe various aspects of a room from an acoustical point of view.
The most important of them are collected and defined in the ISO 3382 standard[1] [1].

2 ISO 3382 Acoustical Parameters

Before describing the software, a quick review of the subject matter may be required.[2]

2.1 Reverberaton time

This is the best known parameter, a sort of fundamental of the acoustic science, defined as the time - in seconds - needed by the sound pressure level to decrease by 60 dB; the rate of decay is measured by the linear least-squares regression of the measured decay curve from a level 5 dB below the initial level to 35 dB below. This is called T_{30}. When the final value is changed[3] to 25 dB we obtain T_{20}. It is important to note that, even if the decay rate is measured over a range of just 30 or 20 dB, the reverberation time is always expressed as the time required for a 60 dB decay. If the decay is linear, hence, $T_{60} = T_{30} = T_{20}$.

2.2 Clarity, Definition and Center Time

These parameters express a balance between early and late arriving energy, that is useful to measure the clarity as perceived by human ears. The C_{t_e} parameter is used with $t_e = 50$ ms, or $t_e = 80$ ms dependently on the destination of the room, speech or music listening, respectively. C_{t_e} is defined as

$$C_{t_e} = 10 \log \frac{\displaystyle\int_0^{t_e} p^2(t)\,\mathrm{d}t}{\displaystyle\int_{t_e}^{\infty} p^2(t)\,\mathrm{d}t} \quad [\mathrm{dB}]. \tag{1}$$

where $p(t)$ is the measured impulse response.[4]

[1] In this article all the references are to the ISO 3382:1997 standard; there is a ISO 3382-1:2008 revision that is actually at approval stage. We've seen a draft edition in which there are no differences in parameters definitions between this and the previous version.

[2] The understanding of architectural acoustic measurements implies the knowledge of impulse measures theory and techniques; possible references are [2] and [3].

[3] This can be mandatory when environment noise is too loud.

[4] $p(t)$, when not differently specified, is always a pressure impulse response measured with an omnidirectional

The *Definition index*[5] is less used than *Clarity*; in the balance it includes not only the late energy, but the total energy of the signal:

$$D_{50} = \frac{\displaystyle\int_0^{50\mathrm{ms}} p^2(t)\,\mathrm{d}t}{\displaystyle\int_0^{\infty} p^2(t)\,\mathrm{d}t}. \tag{2}$$

At last the *Center Time* is the time of the center of gravity of the squared impulse response; it can be measured in seconds, and avoids discrete division of the impulse response into early and late periods:

$$T_s = \frac{\displaystyle\int_0^{\infty} t \cdot p^2(t)\,\mathrm{d}t}{\displaystyle\int_0^{\infty} p^2(t)\,\mathrm{d}t} \quad [\mathrm{ms}]. \tag{3}$$

2.3 Sound strength

In an open space, it's obvoiusly impossible to naturally 'amplify' an acoustic signal, but in a closed room this can be possibile and a measure of *how much* the environment increases (or decreases) the perceived loudness of a sound is given by the Sound Strength parameter. It is defined as the logarithmic ratio of the squared and integrated sound pressure of the measured impulse response to that of the response measured at a distance of 10 m from the same omnidirectional sound source in a free field:

$$G = 10\log\frac{\displaystyle\int_0^{\infty} p^2(t)\,\mathrm{d}t}{\displaystyle\int_0^{\infty} p_{10}^2(t)\,\mathrm{d}t} \quad [\mathrm{dB}], \tag{4}$$

where $p_{10}(t)$ is the instantaneous sound pressure of the impulse responset measured at a distance of 10 m in a free field.

2.4 Early and late lateral energy measures

The acoustical comfort is greatly increased if the sound experience can surround the listener: for this reason a set of *spatial* parameters that give a measure of the sound source virtual width, or the *enveloping* effect is defined. These purposes need spatial informations: with a recording system composed by an omnidirectional and a figure-of-eigth pattern microphones[6] it is possible to get some of these information and calculate, i.e., *Lateral Fraction* parameter, defined as:

$$LF = \frac{\displaystyle\int_{5\mathrm{ms}}^{80\mathrm{ms}} p_L^2(t)\,\mathrm{d}t}{\displaystyle\int_0^{80\mathrm{ms}} p^2(t)\,\mathrm{d}t}. \tag{5}$$

Lateral Fraction expresses the ratio between the lateral sound energy ($p_L(t)$[7]) and the total energy that comes to the listener ($p(t)$[8]), giving a measure of the *Apparent Source Width.* It can be used also the *Lateral Fraction Cosine*, that is another approximation of the lateral energy fraction:

$$LFC = \frac{\displaystyle\int_{5\mathrm{ms}}^{80\mathrm{ms}} |p_L(t) \cdot p(t)|\,\mathrm{d}t}{\displaystyle\int_0^{80\mathrm{ms}} p^2(t)\,\mathrm{d}t}, \tag{6}$$

it comes from the observation that, in fact, the figure-of-eight pattern is a cosine pattern, and the resulting contribution to lateral energy for an individual reflection varies with the square of the cosine of the angle of incidence of the reflection relative to the axis of maximum sensitivity of the microphone.
Surround effect can be measured with the *Late Lateral Sound Energy* (LG), defined as:

$$LG_{80,\infty} = 10\log\frac{\displaystyle\int_{80\mathrm{ms}}^{\infty} p_L^2(t)\,\mathrm{d}t}{\displaystyle\int_0^{\infty} p_{10}^2(t)\,\mathrm{d}t} \quad [\mathrm{dB}], \tag{7}$$

where $p_{10}^2(t)$ is the same measure seen in 2.3. This parameter is related to the perceived listener envelopment or spaciousness in the auditorium.

2.5 Interaural cross-correlation

Human spatial perception is due to the natural *stereo* human audio system, particularly to the difference between the signals that arrives

microphone.

[5]Adimensional parameter, it can be normalized to '1', or a percentile.

[6]W and Y channels of a microphone, such as a Soundfield®, can do the same.

[7]This is the signal caught by the figure-of-eight pattern microphone: obviusly the norm specifies that it has to be pointed in a way that minimize the direct waves response and maximizes the lateral waves one.

[8]This is the signal caught by omnidirectional microphone.

to the two ears. No differences between left and right sounds means no spatial information, and then a listener won't be able to locate a sound source in a scene that will seem completely flat to him. With a binaural microphone it is possible to record exactly what arrives at two ears, and a cross-correlation operation between these two signals will reveal the *spatial degree* of the information: this is the definition of the *Interaural Cross-Correlation* function (IACF), that can be expressed in the following mathematical form:

$$IACF_{t_1/t_2}(\tau) = \frac{\int_{t_1}^{t_2} p_l(t) \cdot p_r(t+\tau)\,\mathrm{d}t}{\sqrt{\int_{t_1}^{t_2} p_l^2(t)\,\mathrm{d}t \cdot \int_{t_1}^{t_2} p_r^2(t)\,\mathrm{d}t}}. \tag{8}$$

Studies have shown that *Interaural Crosscorrelation Coefficients* (IACC) correlate well with the subjective quality *spatial impression* in a concert hall, like the lateral energy parameters set:

$$IACC_{t_1/t_2} = \max(|IACF_{t_1/t_2}(\tau)|) \quad \text{for} \quad -1\,\mathrm{ms} < \tau < +1\,\mathrm{ms}. \tag{9}$$

2.6 Stage parameters

During a musical performance not only the listeners' but also the musicians' perception is important. Musicians need to hear themselves and each other in a way that allows them to perform well. On-stage acoustics are described by 'stage parameters'. The ISO 3382 standard indicates two useful indexes, defined by the following equations:

$$\mathrm{ST}_{\mathrm{Early}} = \frac{\int_{20\mathrm{ms}}^{100\mathrm{ms}} p^2(t)\,\mathrm{d}t}{\int_{0}^{0\mathrm{ms}} p^2(t)\,\mathrm{d}t} \quad [\mathrm{dB}] \tag{10}$$

$$\mathrm{ST}_{\mathrm{Late}} = \frac{\int_{100\mathrm{ms}}^{1000\mathrm{ms}} p^2(t)\,\mathrm{d}t}{\int_{0}^{0\mathrm{ms}} p^2(t)\,\mathrm{d}t} \quad [\mathrm{dB}]. \tag{11}$$

The $\mathrm{ST}_{\mathrm{Early}}$ is related to the musicians' ensemble conditions, that is how they hear themselves, the $\mathrm{ST}_{\mathrm{Late}}$, instead, can give indications about the response of the hall as heard by the musicians.

3 The Aurora's Acoustical Parameter plug-in

Aurora[9] is a suite of plug-ins for Adobe Audition®developed and manteined by professor Angelo Farina. This software includes:

- an ISO 3382 Acoustical Parameters calculator
- complete toolset for impulse response measurement with MLS technique
- complete toolset for impulse response measurement with *ExponentialSineSweep* technique
- various convolution tools, including warped FIR
- inverse filters generator, for equalization purposes and creation of cross-talk cancelling filters for binaural reproduction over loudspeakers
- Speech Transmission Index (STI) calculator
- ITUP56 Time History Analyzer
- Synchronous average and IR select, for separating linear and notlinear components of the impulse response
- Acoustic Quality Test (AQT) plugin for the analysis of the sound field in small enclosure,

that is a complete toolset for acoustic measurements, whose description can be very long and cover many aspects of acoustic science; in this context the focus is pointed on the first module, the Acoustical Parameters calculator.[10] Simplifying, given an impulse response, which have to be loaded in the Adobe Audition®enivironment, it calculates the ISO 3382 acoustical parameters.
The input inpulse response can be mono or stereo; in the first case the parameters calculated are:

- Signal Level [dB]
- Noise Level [dB]
- Strenght (G) [dB]
- Clarity 50 ms (C_{50}) [dB]
- Definition 50 ms (D_{50}) [-]

[9] Website: http://www.aurora-plugins.com.
[10] Presented in [4]

- Clarity 80 ms (C_{80}) [dB]
- Centre Time (T_s) [ms]
- Early Decay Time (EDT) [s]
- Reverberation Time User (T_{user}) [s]
- Reverberation Time (T_{20}) [s]
- Reverberation Time (T_{30}) [s]
- Stage parameters [dB] (*optionally*),

in the second case, the user has to specify first what type of microphones pair has been used for the measurements, specifying one of the following choices:

- 2 omnidirectional microphones
- W-Y channels of a Soundfield microphone
- Pressure-Velocity (PU) probe
- Intensimetric (PP) probe
- Binaural microphone,

then, new parameters subset will be calculated depending on the previous selection. In example, if the signal was recorded using a Soundfield®microphone, it makes sense to calculate the *Lateral Fraction* (LF), the *Lateral Fraction Cosine* (LFC) and the *Late Lateral Sound Level* (LG), because we have spatial informations in the stored two channels impulse response.[11] Similarly, a binaural microphones pair is the probe that the standard specifies for *Interaural Cross-correlation Coefficients* (IACC) measure, so if this is the case, the calculation will give back the IACC value, the IACC peak width and its delay, with possible choice of IACC *early*, *late*, or *full*.[12]
All the results are given in the ten octave bands, a linear average and an A-weighted average of the previous.

3.1 Software implementation

The software written by Angelo Farina is based on the Schröder integral theory: using this theory is possible to calculate the reverberation time by simply recording an impulse response, then the decay can be obtained from a backward integration. This is exactly what the software does, it loads the impulse response from the Adobe Audition®environment, filters the signal with the appropriate octave band filter[13], calculates the Schröder integral and then calculates all the parameters. This procedure is repeated for each octave band.
Results are displayed in a window (fig. 1) where, besides all parameters values, is also plotted the energetic impulse response[14] and the decay.

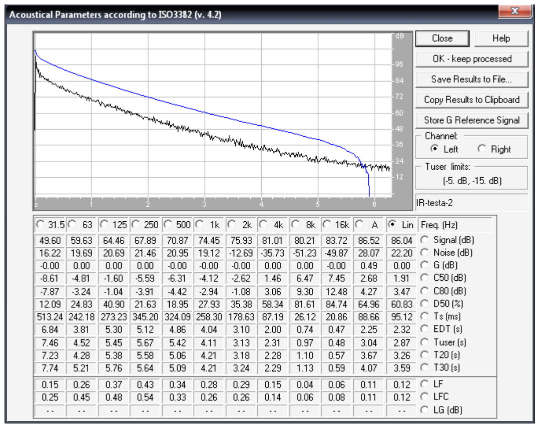

Figure 1: Results window. In black the echogram, in blue the Schröder integral (decay).

It is possible to export the value table in a file, or to the clipboard.

4 The Audacity port

When one thinks about a Mac or a Linux tool for acoustic measurements, it appears there are some intrinsic obstacles to the development of portable applications. The two operating systems *appear* similar, but are really different in many aspects. So writing a multiplatform application that supports Aurora plug-in would have been a long and complex task. A good solution could be the modification of an existing application, but it cannot be commercial, due to its closed source nature. Looking at the open-source world we found many solutions to our project, and between them we have chosen Audacity because

- it is a multiplatform application
- it is good quality audio editor, with expanding possibilities
- it is open-source.

[11] These informations come also from PU and PP probes.

[12] For each of these labels correspond differents integration extremes in the equation 8 and, consequently in the 9.

[13] These filters, like 1/3 octave band, are defined in the IEC 61260:1995 standard.

[14] Also called *echogram*, it is the RMS of the pressure impulse response.

Once found the target application, what would been the best implementation of the port? The plug-in way was discarded, because the poor graphic features of the LADSPA plugins standard; there exist other plugin definitions in the open-source world, but they aren't supported by Audacity yet and it seems that the community welcome was a bit cold. It came immediatly clear that the quickest and more powerful way is to embed the Aurora code in the Audacity source.
This operation requested the code conversion from C to C++, so some new objects were created: EffectAcParameters that manages all the calculations, and others responsible of drawing windows, plots, etc. Another big step was the conversion from Microsoft Windows®windows library to wxWidgets, that is the choice for Audacity; to do that, all the widgets related code was rewritten from scratch, with the closest similarity between original Aurora plug-in and the new module as target. The result is visible in figure 2.

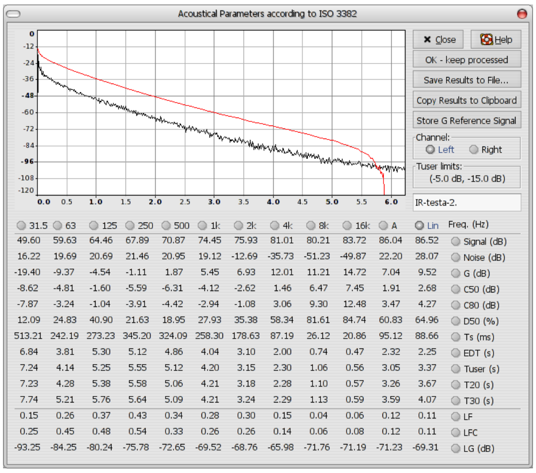

Figure 2: Audacity port of Aurora Acoustic Parameter module (GTK).

The compilation of the code was tested on Linux environment with GTK based window manager (on wich fig. 2 is referred), Mac OSX and Microsoft Windows®, always with success.

5 Acoustical Parameters module comparison

The Acoustical Parameters module was written for scientific use, but it also has to be a reliable tool for anyone that does acoustic measurements. It is then necessary to validate it not only by some heavy "field" use , but also to compare the results from our module to those obtained using the most important commercial toolsets available:

- *Dirac* version 3.0[15] from Brüel & Kjær
- *WinMLS 2004* version 1.07[16] from Morset Sound Development.

The test consisted simply in the calculation of the acoustical paramters for a set of impulse responses recorded with a binaural head in the small concert hall of *La Casa della Musica*, Parma.
The results obtained have been reported in the tables 1, 2, 3, 4 and 5: they are limited to the central bands, because *Dirac 3.0* don't give results over 4 kHz for ISO 3382 parameters. *Dirac* is also clearly intended to be used for measurement sessions, that can be saved, then for elaboration on these sessions in which all measurement setup and parameters, like, i.e., the microphones configuration, etc. are stored. If a sound file cannot belong to a session, the application offers only limted options on it. So it wasn't possible to calculate IACC.
As it can be seen, the values produced by Acoustic Parameters module, *Dirac* and *WinMLS* are very close, in some cases identical.

		Acoust. Param.	*B&K Dirac 3.0*	*WinMLS 2004*
EDT	[s]	4.71	4.72	4.70
T_{20}	[s]	5.57	5.54	5.55
T_{30}	[s]	5.66	5.55	5.64
C_{80}	[dB]	- 3.74	-3.63	-4.2
D_{50}	[-]	0.20	0.21	0.19
$\mathrm{ST_E}$	[dB]	3.14	3.25	-
$\mathrm{ST_L}$	[dB]	8.03	8.37	-
$\mathrm{IACC_E}$	[-]	0.89	-	0.87

Table 1: Parameters confrontation for the 250 Hz band.

The other tables are shown on the next page.

6 Conclusions

This port of the Aurora's ISO 3382 parameters calculator is the beginning of the porting of the entire suite. Main targets were fully reached, so now a reliable, multiplatform ISO 3382

[15] This is not the latest one, but it seem that the improvements from 3.0 to 4.1 - the last release - don't affect the calculation algorithms. There is only an issue: it can manage solely 16 bit unsigned integer samples, so we had to downsample our test file that is a 32 bit float type.

[16] This is the last version released of the software.

		Acoust. Param.	*B&K Dirac 3.0*	*WinMLS 2004*
EDT	[s]	4.82	4.76	4.74
T_{20}	[s]	5.02	5.01	5.01
T_{30}	[s]	4.98	4.96	4.97
C_{80}	[dB]	- 4.40	-4.24	-4.8
D_{50}	[-]	0.20	0.20	0.19
ST_E	[dB]	3.26	4.05	-
ST_L	[dB]	8.70	9.63	-
$IACC_E$	[-]	0.34	-	0.31

Table 2: Parameters confrontation for the 500 Hz band.

		Acoust. Param.	*B&K Dirac 3.0*	*WinMLS 2004*
EDT	[s]	4.44	4.41	4.38
T_{20}	[s]	4.10	4.10	4.11
T_{30}	[s]	4.20	4.19	4.22
C_{80}	[dB]	- 2.55	-2.65	-2.8
D_{50}	[-]	0.30	0.30	0.29
ST_E	[dB]	0.58	0.75	-
ST_L	[dB]	5.95	6.32	-
$IACC_E$	[-]	0.38	-	0.40

Table 3: Parameters confrontation for the 1000 Hz band.

		Acoust. Param.	*B&K Dirac 3.0*	*WinMLS 2004*
EDT	[s]	3.05	3.03	2.97
T_{20}	[s]	3.18	3.17	3.17
T_{30}	[s]	3.25	3.25	3.26
C_{80}	[dB]	- 0.29	-0.15	-0.4
D_{50}	[-]	0.40	0.41	0.39
ST_E	[dB]	1.21	1.13	-
ST_L	[dB]	3.12	3.01	-
$IACC_E$	[-]	0.48	-	0.48

Table 4: Parameters confrontation for the 2000 Hz band.

compliant acoustical parameters calculator it's available under GPL license.

The next modules that has to be converted are the impulse responses measurement tools, including the exponential-sine-sweep generator and the convolver, then Audacity will become a quite complete acoustical measure tool.

		Acoust. Param.	*B&K Dirac 3.0*	*WinMLS 2004*
EDT	[s]	2.18	2.18	2.08
T_{20}	[s]	2.22	2.23	2.23
T_{30}	[s]	2.25	2.26	2.27
C_{80}	[dB]	2.93	2.93	2.6
D_{50}	[-]	0.60	0.60	0.59
ST_E	[dB]	-2.60	-2.57	-
ST_L	[dB]	-1.96	-1.91	-
$IACC_E$	[-]	0.61	-	0.61

Table 5: Parameters confrontation for the 4000 Hz band.

References

[1] ISO 3382:1997. *Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters.* International Organization for Standardization, Geneva, August 2001.

[2] A. Farina. Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique. In *108th AES Convention*, Paris, February 2000.

[3] A. Farina. Room Impulse Responses as Temporal and Spatial Filters. In *WESPAC IX 2006*, Seoul, June 2006.

[4] A. Farina and F. Righini. Software implementation of an MLS analyzer, with tools for convolution, auralization and inverse filtering. In *103th AES Convention*, New York, September 1997.