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1 Introduction to the WinMLS Room Acoustics Module

Room acoustics parameters can be calculated from measured impulse responses using the room acoustics post processing capability of WinMLS. The parameters are calculated according to the definitions, requirements and recommendations of the ISO 3382 standard¹. Additional parameters like speech transmission indices are included as well. The results can be presented graphically, in a text window, or saved to a text file, with contents according to response type and calculation settings chosen by the user.

2 Practical information, tips and warnings

- When measuring room impulse responses using WinMLS, it is recommended that a room acoustics setup is loaded. Such a setup can be loaded and saved from the **Setups Toolbar** (can be turned on/off from the **View** menu), or from **Measurement->Save/Load Measurement Setup**. When the setup is loaded, the **GO** button on the standard toolbar can be used to start a measurement, or **File->Insert** can be chosen to insert an already existing response from a file.
- Calculation options and parameter settings for the room acoustics calculations can be modified from the **Room Acoustics** menu. These settings are saved with the post-processing setup file found on the **Setups Toolbar**.
- The room acoustics calculations have been developed with accuracy as the main goal. Optimization of calculation speed has not been allowed to compromise accuracy. The calculations have been tested against international benchmarks with excellent results².
- The room acoustical parameter results are given with a considerably higher accuracy than the typical measurement repeatability when repeating measurements in a given position in a normal room.
- To avoid time consuming calculations, do not permit excessive impulse response lengths and sampling rates.
- For the noise detection and truncation compensation algorithms to work optimally, the impulse response should be sufficiently long to have a late part with a stationary noise floor, also after filtering in the analysis bands.
- Observe normal caution required for high quality MLS measurements, like avoiding time wrapping, distortion and long averaging times in slowly time-varying systems.

¹ ISO 3382: 1997, *Acoustics - Measurement of the reverberation time of rooms with reference to other acoustical parameters*.

² Post-processing tests have been carried out on impulse responses described in a report by J.S. Bradley, "An international comparison of room acoustics measurement systems," National Research Council of Canada (1996).

- It can be worthwhile spending some time finding the right setup and level settings when measuring in a new room, to find the optimum trade-off between background noise (acoustical and electronic) and spurious components due to distortion. Loud is not always best in terms of resulting signal-to-noise ratio, due to loudspeaker distortion.
- Always control the measurement quality while there is still time to repeat the measurement. Check normalized, logarithmic impulse responses to check whether the response is sufficiently raised above the noise floor and distortion components. Check room acoustical parameters for sensibility. Check decay ranges. Check filtered Schroeder plots to pinpoint anomalies.
- The calculations will in general be meaningless if performed on non-impulsive sound records, even if results may be shown.
- The calculations might be inaccurate when performed on impulse responses with very short decay times, for instance electronic systems (amplifiers, filters, etc.). The decay curve formation is based on the assumption that decay times are on the order of at least a tenth of a second, like it will be in any room but an anechoic chamber.
- Examples of measured room impulse responses and corresponding parameters in text-file format are given in the WinMLS measurement examples folders.
- Further descriptions of sensible measurement procedures can be found in ISO 3382.

3 Specification of input impulse responses

The room acoustics calculations operate on a primary and an optional secondary impulse response. Most parameters are based on a single response, usually measured with an omnidirectional microphone. These parameters are always calculated from the primary response. Some parameters, like lateral fractions and interaural crosscorrelations, are based on two responses. In these cases a secondary response must be provided as well, measured with an omnidirectional or figure-of-8 microphone. The primary and secondary impulse responses must always have the same length and sampling frequency.

Specification of input responses to the room acoustics calculations is done by making the responses active in plot windows. The primary response is always taken as the active measurement in the upper plot, and the secondary response is taken as the active measurement in the lower plot. If only a single plot is displayed, its active measurement becomes the primary response. The secondary response is still the active measurement in the lower plot, and is specified in the **Measurement Selection Toolbar** (shown in the figure below) or in **Plot->Select Plot Type(s) and Active Measurement**. In any case, it is a good idea to check one of these to make sure that the correct impulse responses are specified before room acoustics calculation is initiated.



After a measurement the resulting response becomes the active measurement in the current plot window. It is therefore automatically chosen for room acoustics calculations. Measurement channel two can be automatically sent to the lower plot by checking the appropriate box in the bottom part of **Measurement->Measurement Tasks**. If it is checked and a multi-channel measurement is performed, channel 1 will be plotted in the upper plot and channel 2 will be plotted in the lower plot. Make sure two plots are viewed, with only one plot channel two will not be displayed.

If several measurements are inserted using **File->Insert...**, the first measurement will be displayed in the upper plot, the second will be displayed in the lower plot, the third will be displayed in the upper plot, and the fourth in the lower, and so on. Note that measurements may be selected in any order by clicking on the measurement files in the **File->Insert...** dialog.

4 Making calculations

The room acoustics calculations can be initiated in three ways:

1. From the main menu, item **Room Acoustics->Calculate**
2. Choosing **Room Acoustics Parameters** as plot type, either from **Plot->Select Plot Type(s) and Active Measurement** or the **Plot Toolbar**.
3. Checking **Display selected room acoustics parameters** at the bottom part of **Measurement->Measurement Tasks**.

In case 1 calculations are carried out when choosing **Calculate**, and results for the selected parameters are displayed in a text grid window and/or saved a text file according to the settings in **Calculation Options**.

In case 2 calculations are carried out each time plots are refreshed, and the result for the single selected parameter is plotted (the settings dialog box is displayed by double clicking in the plot). In both cases input responses are taken as active measurements as described above, and calculations are made according to the settings under the items **Calculation Options** and **Parameter Settings** in the **Room Acoustics** menu.

In case 3 room acoustics parameters are being computed and displayed to a grid when measurement files are inserted using **File->Insert**. If **Room Ac. Parameters** is selected as plot type, the selected parameter will be displayed in the plot as well according to case 2. Note that this will reduce the speed, since the computation will be performed twice.

Before parameter calculation, WinMLS automatically detects the start of the impulse response. This creates a reference for all subsequent time windowing and alignment of the response. Some parameters require impulse response energy to be calculated in time windows. All windowing is done before filtering to avoid energy leakage across window borders, even if this may increase the processing burden considerably. Then the impulse response is filtered in the specified frequency bands, the noise floor is optionally analyzed and compensated for, and a decay curve is calculated by backwards integration. The algorithms for the decay curve formation and parameter calculation are further described in the following subsections.

Errors leading to calculation abortion will prevent results from being presented and produce error messages explaining the situation.

5 Presenting results

Room acoustics parameters can be presented in a text grid, saved to a text file, or plotted directly in a graphics window. The check boxes **Display in text grid** and **Send to text file** specifies how to present and/or save results when the calculation is initiated through **Room Acoustics->Calculate**. All chosen parameters are displayed in the grid or saved to the file.

If the text grid box is checked, a result window will open after the calculation is finished. The data in the grid can be copied to the Windows clipboard for export to spreadsheets or word processing software.

An example of output data is shown below, the contents are:

1. The name of the primary measurement, or impulse response file
2. Octave band center frequencies in Hz
3. SNR in dB, for primary and (optional) secondary input response
4. Parameters as function of frequency
5. Parameters independent of frequency
6. Estimated source to receiver distance in m

	63	125	250	500	1000	2000	4000	8000	16000
F(Hz)	63	125	250	500	1000	2000	4000	8000	16000
SNR(dB)	20.7	43.7	47.1	53.4	49.3	49.9	47.4	45.4	44.0
EDT(s)	1.39	1.20	1.32	1.55	1.79	1.42	1.09	0.67	0.51
T30(s)	0.46	2.37	2.12	2.29	2.38	2.12	1.74	1.13	0.81
Tu(s)	0.46	2.38	2.30	2.29	2.34	2.07	1.69	0.87	0.61
Tc(ms)	119.4	108.7	127.0	126.3	134.4	107.3	94.2	55.4	41.2
C(dB)	0.6	0.1	-6.6	-2.0	-3.3	-0.7	-0.3	3.1	4.6
D(%)	436.0	61.2	15.9	40.4	33.7	45.7	48.9	68.1	79.5
STI	0.57								
RASTI	0.55								
r(m)	21.8								

(This example incidentally shows a typical anomaly, the combination of a low frequency octave band filter, a short time window, windowing before filtering and a high noise floor creates transient effects that can produce odd looking results, like the 63Hz D(%) value. Calculations are according to recommendations, but the practitioner must use his common sense in interpretation of the data.)

Blank fields indicate calculation errors, which are most often caused by noisy impulse responses and thereby insufficient decay range, or that time ranges are insufficient for a particular parameter. In these cases only particular parameters will be missing.

In some cases all parameters can be missing for a particular frequency band. The most common reason is when truncation of the impulse response at or above the noise floor is chosen in **Calculation Options->Integration options**, and the truncation point cannot be determined due to too much noise or an odd shape of the response after filtering in this band. See section 6.3 for more information about this situation.

If output to a text file box is chosen, results will be saved to a text file with a similar format as the grid. Saving directly to text file can only be done if the room impulse response itself is saved in a file prior to calculation. The text file ends up having the same name as the impulse response file, with the extension **.txt**.

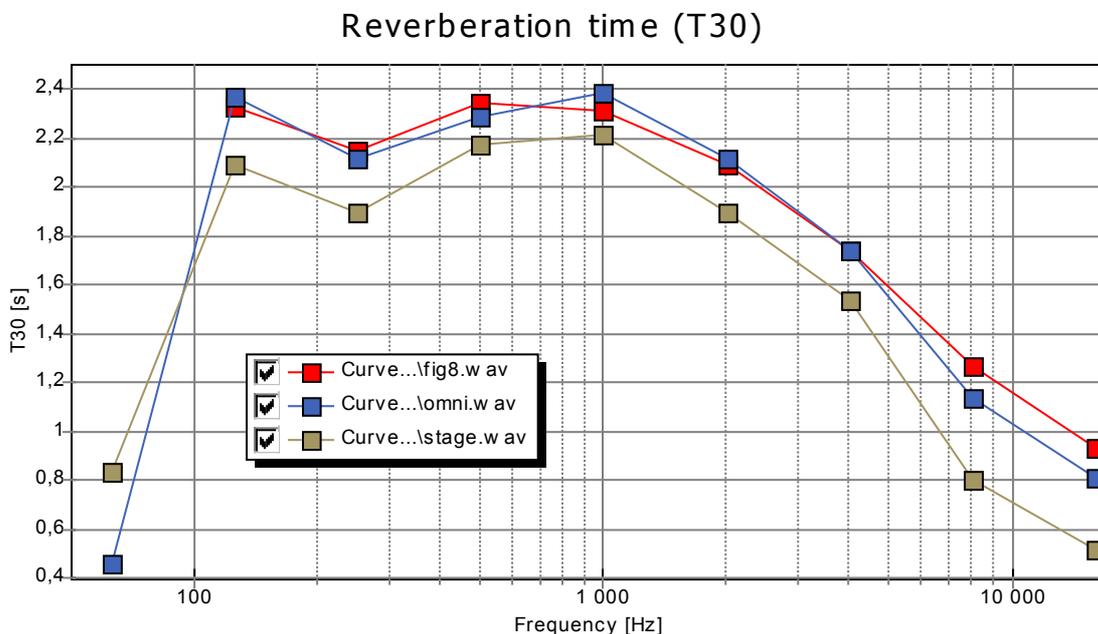
For calculations using a secondary impulse response as well, the name of the resulting text file will be composed using both filenames, for instance:

Primary: C:\WinMLS\Debug\Measurements\Studio1.wmb

Secondary: C:\WinMLS\Debug\Measurements\Studio2.wmb

Result: C:\WinMLS\Debug\Measurements\Studio1Studio2.txt

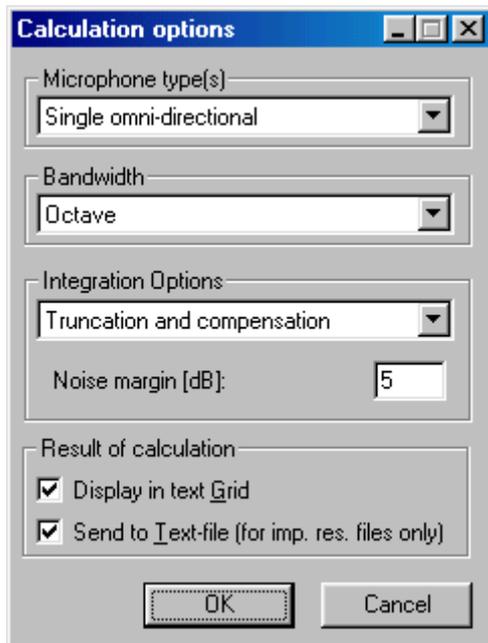
Results can also be plotted directly in a plot window. Only a single parameter at a time can be displayed as function of frequency, but it is possible to plot the same parameter calculated from several impulse responses for easy comparison. Choose the **Room Ac. Parameters** plot type (can be done from the **Plot Toolbar** or **Plot->Select Plot Type(s) and Active Meas.**). The parameter to be plotted is chosen by double-clicking in the plot or from **Plot->Plot Type Settings->Room Acoustics**. An example of plotting reverberation time in octave bands for three different responses is shown below.



6 Calculation Options

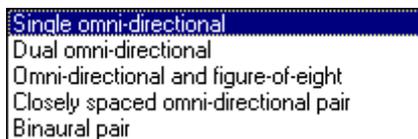
Room Acoustics->Calculation Options will display the dialog box controlling the fundamental setup of the room acoustics calculations. All settings are saved in the post-

processing setup file. The dialog can also be reached via **Plot->Plot Type Settings->Room Acoustics**.



6.1 Microphone type(s)

Choose the processing mode by indicating the constellation of microphone(s) used during the measurements. The options are:



1. *Single omni-directional*
Single microphone, most common situation.
2. *Dual omni-directional*
Two independent microphones. This option is provided for independent measurements but where simultaneous calculation can be an advantage, when measuring using the 2-channel capability of WinMLS. Two separate calculations will be carried out, and results will be presented in separate windows or files.
3. *Omni-directional and figure-of-8*
Combination of an omni-directional (primary) and a figure-of-8 (secondary) microphone. This option is used for calculation of lateral parameters, see section 7.6. The secondary response will only be used for these parameters, other parameters are calculated from the primary response.
4. *Closely spaced omni-directional pair*
A pair of impulse responses from either two identical microphones measured simultaneously, or from a single microphone measured in two locations. In the present

version of WinMLS the processing is exactly like that carried out for option 5. This option is included for future extension, for instance calculation of lateral parameters without using a figure-of-8 microphone.

5. *Binaural pair*

A pair of impulse responses from either two identical microphones measured simultaneously, or from a single microphone measured in two locations. This option is used for calculation of binaural parameters, see section 7.7. The microphones are normally placed in or at the ear canal of a dummy head or a real person. The secondary response will only be used for these parameters, other parameters are calculated from the primary response.

The different microphone types make possible different parameter calculations. For a description of the various parameters, see section 7.

6.2 Bandwidth

After being loaded to the room acoustics module, the impulse responses are filtered in bands according to the settings in this box. The options are:



1. *Octave*

Octave band filters, responses corresponding to the requirements in IEC 1260³. Lowest octave where calculation is attempted is 63Hz, highest octave depends on the sampling frequency of the measured impulse response.

2. *Low-Mid-High*

Double-octave band filters, responses corresponding to the requirements in IEC 1260. Centre frequencies of 176.8Hz, 707.1Hz and 2828.4Hz provides three double octave filters corresponding to a low-mid-high combination of octave bands: 125Hz and 250Hz, 500Hz and 1kHz, 2kHz and 4kHz.

4. *Broadband*

No filtering is performed.

5. *1/3 Octave (Reverberation time only)*

Third-octave band filters, responses corresponding to the requirements in IEC1260. Lowest 1/3-octave where calculation is attempted is 40Hz, highest band depends on the sampling frequency of the measured impulse response. For this bandwidth only reverberation time is generally of interest, and therefore it is the only parameter calculated. The dialog **Room Acoustics->Parameter Settings** will be overridden and not available for changes. To minimize the influence of the filter response on the reverberation time calculation, the impulse responses are filtered backwards. This greatly improves the accuracy for short decay times at low frequencies.

³ IEC 1260: 1995, *Electroacoustics – Octave-band and fractional-octave-band filters*.

6.3 Integration options

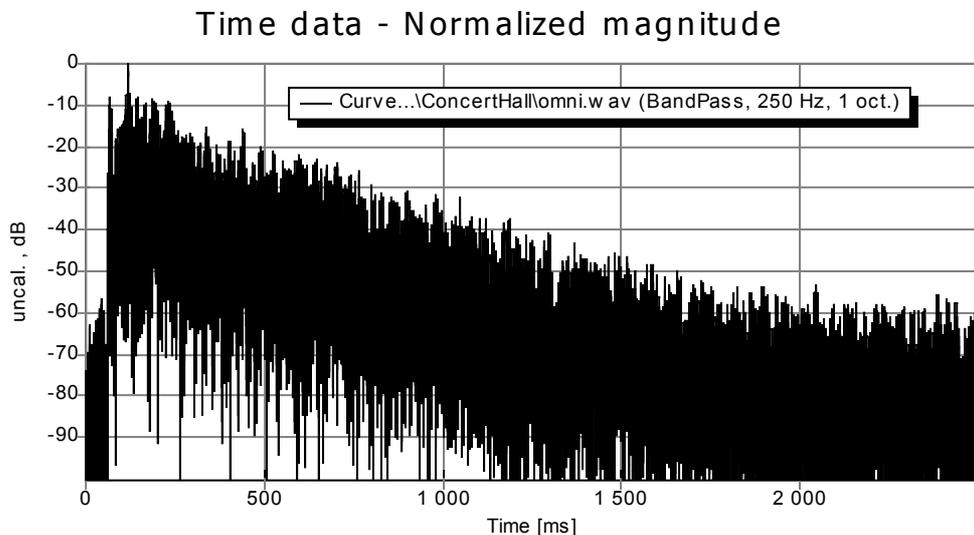
After filtering, backwards integration of the impulse response to form a decay curve, or Schroeder curve, is done as further preparation to the parameter calculations. If $h(t)$ is the filtered impulse response, the decay curve is:

$$d(t) = 10 \log \left(\int_t^{\infty} h^2(\tau) d\tau \right) \text{ dB} \quad (1)$$

This provides a monotonically falling decay. The curve is usually normalized to start at 0dB. In practice the impulse response record is finite, and the integration becomes finite as well:

$$d(t) = 10 \log \left(\int_t^{t_N} h^2(\tau) d\tau \right) \text{ dB} \quad (2)$$

Also, a measured response tends to contain some noise. Due to the phase randomizing property of the MLS deconvolution, additive noise of any kind in the measurement will give rise to a stationary noise floor, see the example below. This is advantageous as a combined noise-decay analysis of the filtered impulse response record can reveal the location of the cross-point between the response and the stationary noise floor, the response can be truncated at or above this point to lessen the influence of the noise on the results, and finally the truncated energy can be estimated and taken into consideration.



If compensation for truncated energy is incorporated, and the estimated correction is denoted C , the decay curve is:

$$d(t) = 10 \log \left(\int_t^{t_N} h^2(\tau) d\tau + C \right) \text{ dB} \quad (3)$$

The integration options in WinMLS are:

Truncation and compensation
Truncation
Whole response

1. *Truncation and compensation*

Integration of the impulse response, after truncation at or above the crosspoint between the response and the stationary noise floor, compensated for the truncated energy.

2. *Truncation*

Integration of the impulse response, truncated at or above the crosspoint between the response and the stationary noise floor.

3. *Whole response*

Integration using the whole impulse response, no noise analysis carried out.

The noise analysis, truncation and compensation are done after filtering, separately for each frequency band. Option 1 is the most accurate, and therefore recommended for most room impulse response measurements. However, it requires a measurement of reasonably high quality to give good results.

The crosspoint between the decaying response and the stationary noise floor is detected using an iterative algorithm⁴. Subsequently, a truncation point is determined using a safety margin above the crosspoint. With the **Noise margin** parameter the user can specify in dB how far above the noise level at the estimated crosspoint the truncation point is chosen and integration shall start. Typical values are 5-10dB, the recommended value is 5dB. Higher margins should be used only with responses with extraordinary high signal-to-noise ratios. The late slope of the truncated response is calculated by a line fit, and a compensation for the energy lost by truncation is estimated assuming exponential decay to infinity.

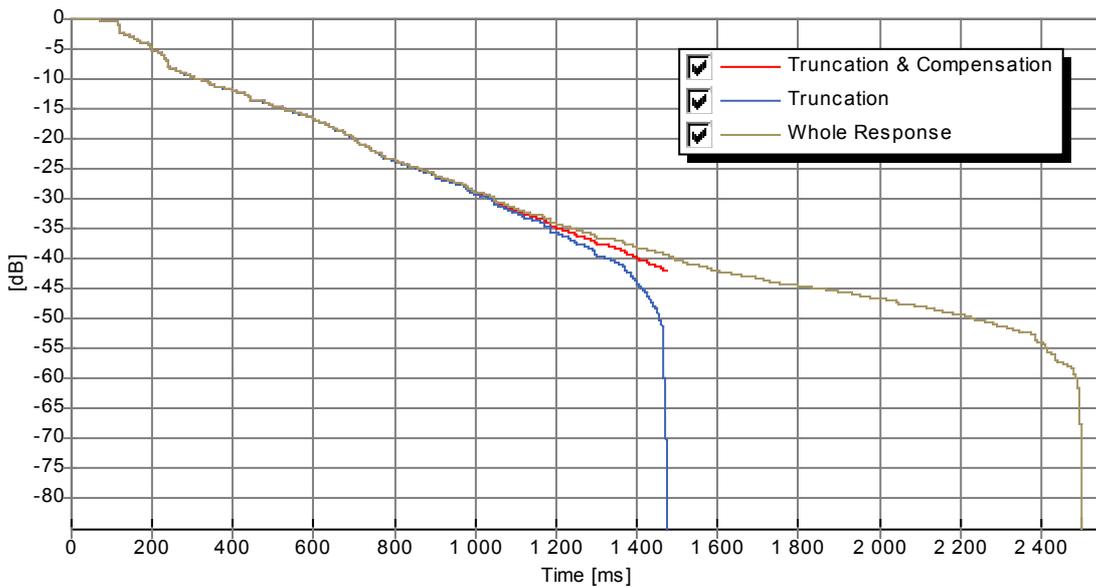
The decay curve forms the basis for calculation of the various decay time parameters, and it can be plotted using the Schroeder plot-type. The plots below give examples of the three options. If truncation is chosen, all room acoustical parameters involving the late part of the response is calculated based on a similarly truncated impulse response. If compensation is chosen and the truncated energy can be successfully estimated, all room acoustical parameters involving the late part of the response are corrected for the truncated energy.

If the truncation point cannot be determined due to too much noise or an odd shape of the response after filtering, no parameters will be calculated for that particular frequency band. If a response produces trouble, turning off truncation and compensation and inspecting the filtered impulse response and Schroeder curve might reveal the source of the problem.

The figure below gives an example of the three options.

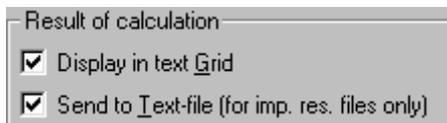
⁴ A. Lundeby, T.E. Vigran, H. Bietz and M. Vorländer, "Uncertainties of Measurements in Room Acoustics," *Acustica*, vol. 81, pp. 344-355 (1995).

Normalized Schroeder Curve



6.4 Result of calculation

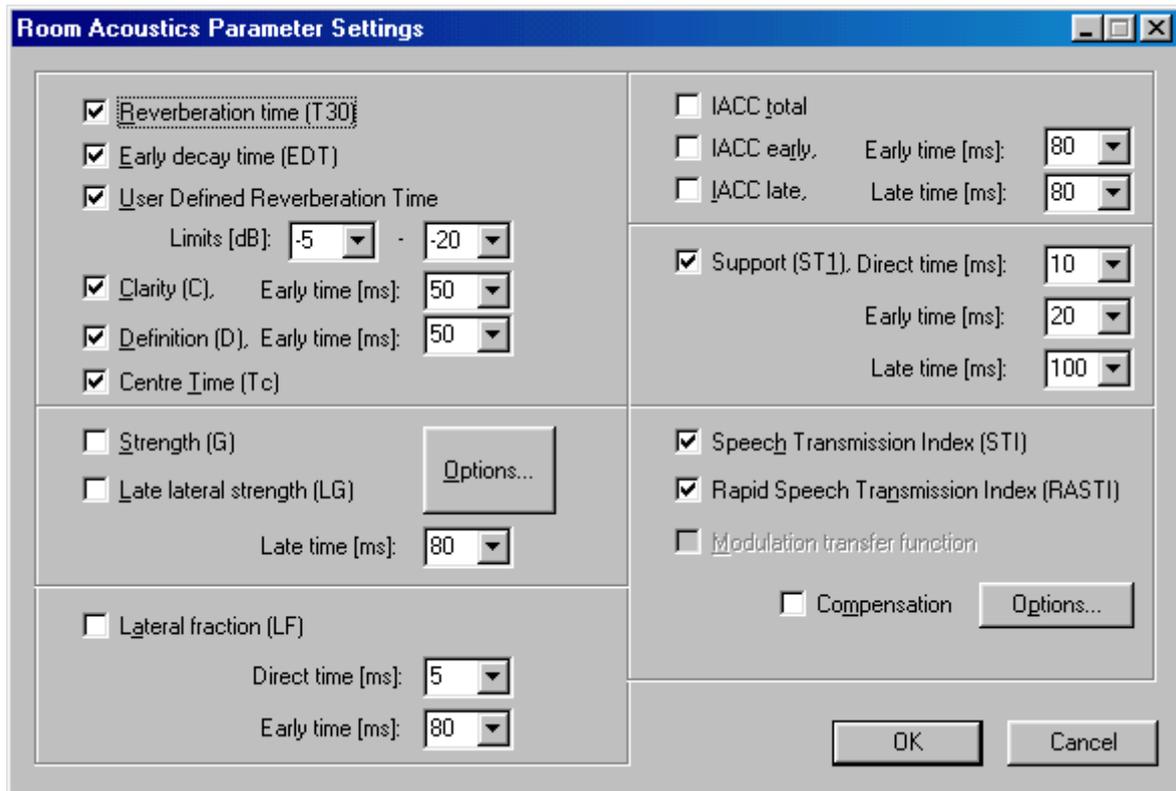
Room acoustics parameters can be presented in a text grid, saved to a text file, or plotted directly in a graphics window. The check boxes **Display in text grid** and **Send to text file** specifies how to present and/or save results when the calculation is initiated through **Room Acoustics->Calculate**. More information can be found in section 5.



7 Parameter settings

Room Acoustics->Parameter Settings will display the dialog controlling which parameters are to be calculated, using check boxes. For some of the parameters the calculation can be controlled by specifying level and time limits, or even more advanced settings. All settings are saved in the post-processing setup file. The dialog will not be available if 1/3-octave bandwidth is chosen in **Room Acoustics->Calculation options->Bandwidth**, since only reverberation time is calculated for the 1/3-octave. The dialog can also be reached via **Plot->Plot Type Settings->Room Acoustics**.

For a more thorough introduction to the room acoustical parameters, see the ISO 3382 standard, or any good textbook on room acoustics.



7.1 Decay range (SNR)

The **signal-to-noise ratio (SNR)** is not included in the **Parameter Settings** dialog, it is however an important parameter, and it is always the first parameter given in the results grid (and can be displayed as marks in the room acoustics plots) and is therefore discussed here.

Utilizing the phase randomizing property of the MLS deconvolution, the actual signal to noise ratio in the impulse response measurement can be estimated from the resulting response itself. This ratio is, however, not of much practical use in room acoustical analysis. Of much more interest is the so-called **decay range**, which directly describes the range of the decay curve which is available for analysis. For an explanation of the decay curve, see section 6.3.

The SNR in dB as displayed by WinMLS is actually the decay range. It is taken as the level of the decay curve at the point where the response is truncated before forming the decay curve. The resulting decay range is therefore, in addition to the physical signal-to-noise ratio in the measurement, dependent on the type and shape of the room impulse response and the chosen noise margin in the response truncation.

The decay range can be used directly to evaluate the quality of the room acoustical parameter calculations, especially the reverberation time T30. It should be noted, and is illustrated clearly in the examples in section 6.3, that if no compensation for energy removed by truncation has been done, the decay range as calculated here is meaningless. It will therefore not be displayed in the results grid unless Truncation and compensation is chosen in **Room Acoustics->Calculation Options->Integration Options**.

If a secondary impulse response is used due to the chosen microphone option, a second row of SNRs will appear in the result grid. This is calculated separately for the secondary response.

7.2 Decay times

The **reverberation time (T30)**, is the 60dB decay time calculated by a line fit to the portion of the decay curve between -5 and -35dB. If the decay range is less than 35dB, the T30 is in WinMLS calculated using the range available. It is therefore important when using the T30, to keep an eye on the SNR to evaluate the quality. To be a valid measurement according to the ISO 3382 standard, the SNR should be displayed as 35dB or larger.

The standard recommends a certain safety margin to the noise floor. It should be noted that the Noise margin parameter taken into account before forming the decay curve and calculating the decay range provides this safety margin. It is user changeable and described in section 6.3.

The **early decay time (EDT)**, is the 60dB decay time calculated by a line fit to the portion of the decay curve between 0 and -10dB.

The **user defined reverberation time (Tu)**, is the 60dB decay time calculated by a line fit to the portion of the decay curve in an interval specified in the level limit boxes, remembering the minus signs in front. It is often used as a T15 (-5 to -20dB) or T20 (-5 to -25dB) parameter, giving a well defined decay time measure for responses with inferior decay range, or providing an indication of double sloping of the decay if it is significantly different from the T30.

All the decay time parameters are given in seconds. If the decay range in a particular frequency band is insufficient for EDT or Tu measurement, no calculation will be carried out.

7.3 Early energy ratios

The early to late index, or clarity, is

$$C = 10 \log \frac{\int_0^{t_e} h^2(t) dt}{\int_{t_e}^{\infty} h^2(t) dt} \quad \text{dB} \quad (4)$$

where t_e is the early time, typically 80ms for music application and 50ms for speech or stage evaluation.

The early to total index, or definition, is

$$D = 100 \cdot \frac{\int_0^{t_e} h^2(t) dt}{\int_0^{\infty} h^2(t) dt} \quad \% \quad (5)$$

Definition is mostly used for speech signals, and the early time is usually taken as 50ms.

The upper integration limits are taken as the truncation point determined in the decay curve formation, or the end of the impulse response, according to the integration option specified. If compensation for truncated energy is calculated, this is taken into account as well to increase accuracy.

The early limits can be changed in the boxes.

7.4 Centre time

The centre time is the first moment of the impulse response $h(t)$:

$$T_c = \frac{\int_0^{\infty} t \cdot h^2(t) dt}{\int_0^{\infty} h^2(t) dt} \quad \text{s} \quad (6)$$

The upper integration limits are taken as the truncation point determined in the decay curve formation, or the end of the impulse response, according to the integration option specified. If compensation for truncated energy is calculated, this is taken into account as well to increase accuracy.

Due to the time-weighting in calculating the first moment, the centre time is more sensitive to the influence of noise, and will therefore be particularly inaccurately calculated if the SNR is low. To compensate for this, an analytical compensation for the truncation of the first moment is also calculated in the truncation and compensation integration option.

7.5 Strength

The sound strength, or level, is defined as

$$G = 10 \log \frac{\int_0^{\infty} h^2(t) dt}{\int_0^{\infty} h_{10}^2(t) dt} \quad \text{dB} \quad (7)$$

where $h_{10}(t)$ is the free-field response of the (assumed omni-directional) source at a distance of 10m, filtered in the same way as the impulse response itself.

The upper integration limit in the numerator is taken as the truncation point determined in the decay curve formation, or the end of the impulse response, according to the integration option specified. If compensation for truncated energy is calculated, this is taken into account as well to increase accuracy.

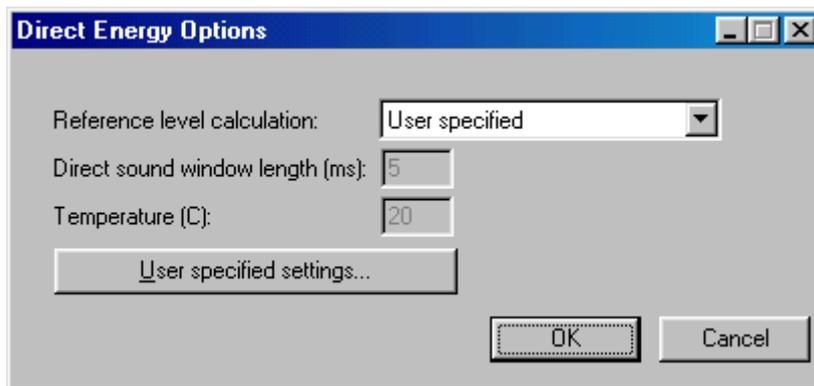
The reference level is:

$$L_{ref} = 10 \log \int_0^{\infty} h_{10}^2(t) dt \quad \text{dB} \quad (8)$$

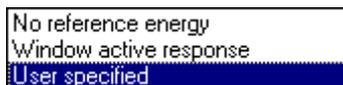
There are various ways of specifying or estimating the reference level, and this is governed by the choices in the direct energy options dialog box. The method is common for strength (G) and late lateral strength (LG).

7.5.1 Direct energy options

Pushing the **Options...** button opens up the direct energy options dialog box, where it is specified how to calculate the reference level.



The main options are:



1. *No reference energy*

L_{ref} is assumed to be 0dB in all frequency bands, which means no correction is made on the level calculated from the primary impulse response. This can be used for instance if the source level is unknown or of no interest, but the level difference at various measurement positions shall be compared. In this case no other options are available in the dialog.

2. *Window active response*

$h_{10}(t)$ is estimated by windowing the primary impulse response, and L_{ref} is calculated subsequently. In this case the window length and temperature is available for change.

3. *User specified*

L_{ref} is given directly by the user, or calculated based on reference measurements specified by the user.

Option 2 and 3 will be explained further in the following subsections. It should be noted that the standard requires a fairly elaborate calibration measurement of the sound source to provide the reference energy at 10m. Only option 3 gives the possibility of obtaining G-values according to the ISO 3382 recommendations.

7.5.2 Window active response

This option is incorporated to provide a simple to use reference level calculation, without the use of separate reference response measurement or time-consuming reference level

measurements. The reference free-field impulse response is estimated from the measured (primary) impulse response as:

$$h_{10}(t) \approx \frac{r}{10} h(t) \cdot w(t) \quad (9)$$

where r is the estimated source to receiver distance and $w(t)$ is a half-Hamming window starting on the arrival of the direct sound in the impulse response. After windowing, the reference response is filtered, and the reference levels are calculated.

The window length can be specified in the dialog box, in ms. A recommended value is 5ms, which provides a useful trade-off between incorporating the impulse response of the loudspeaker while windowing out reflections from the surroundings. In normal concert hall measurements especially the stage floor reflection can be bothersome in this respect, it will in most cases fall inside the 5ms window.

The source-to-receiver distance is estimated based on the detected initial time delay of the direct sound, which requires that the measurement must have been performed with a feedback loop in place. The temperature on the measurement site can also be specified, in degrees Celsius, as this influences the speed of sound and thereby the estimated distance. No great accuracy is required, and in most room measurements 20°C, normal room temperature, can be used. It should also be noted that the initial time delay detected from the impulse response will provide the distance to the face of the loudspeaker, not necessarily to the acoustic center of the source.

Comparison with properly calibrated measurements have demonstrated that this method will normally provide too high G values at low frequencies due to insufficient window length, fairly good G values at mid frequencies (500-1000Hz), and too low G values at high frequencies due to the influence of the immediate surroundings of the transducers. Because of this G cannot be said to be calculated according to the ISO 3382 requirements.

7.5.3 User specified reference level

If user specified reference level is used, yet another dialog box can be opened with the button **User specified settings**. In this dialog L_{ref} can be specified directly or calculated from reference measurements, all assuming calculation in octave bands are used.

Direct Energy corrections

Octave band corrections [dB]

63 Hz	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz	16 kHz
26.1	37.2	42.7	45.3	51.1	55.2	56.8	58.8	-30.1

Octave band corrections [dB]

Source distance - specify in meter (meas. initial time delay is not used): 3

Window type: Whole response (anechoic)

Temperature (C): 20

Perform Computation...

OK Cancel

Direct specification of L_{ref} is done in dB. This is useful in the case where the source and measurement equipment has been pre-measured in an anechoic room or otherwise, the reference level calculated, and the actual room measurement performed with exactly the same equipment and level settings. Alternatively known level changes can be used and corrected for. The calculation procedure described in the following subsection can be used to find reference levels for later use.

7.5.4 Reference level calculation procedure

A reference level calculation procedure is implemented for use with various reference measurement situations. The procedure is initiated by pressing the **Perform computation...** button, which opens up a browser window and the user can specify a set of reference impulse response files. These responses must have the same length and sampling frequency. The responses are windowed according to the choice in **Window type**, and scaled according to the choice in **Source distance**. Thereafter the energy of each is calculated in bands, and the reference level is found by averaging the energy across all the specified responses. The results are displayed in the **Direct energy corrections** dialog box, and should be saved for later use and reference. The calculation might take a while.

For a true omni-directional sound source a reference measurement in a single direction would be sufficient. Most practical sources are 12-faced speakers that start to show lobes in the polar pattern around 1kHz. The ISO 3382 standard therefore specifies that a proper reference level measurement shall be made in a free-field, and data from at least 29 directions shall be energy averaged. WinMLS can handle such a reference measurement, as well as any other less accurate reference that might be chosen.

Alternatives may range from the full set measured in a free-field, via fewer directions, increasingly reflective environments, to actual in-situ single direction responses. The user must choose a reference from what is practically possible, and be aware of the resulting accuracy.

The windowing options are:

Whole response (anechoic)
50ms rectangular (anechoic)
5ms Half-Hamming (in-situ, non-standard)

1. *Whole response (anechoic)*

This option assumes an actual free-field, or anechoic, measurement of high quality. If this is the case, it provides accurate reference levels across a wide frequency range.

2. *50ms rectangular (anechoic)*

This option is similar to option 1 in that it is suitable for free-field measurements. The windowing improves accuracy for most measurements, as in most bands data beyond 50ms will be noise anyway, and the window is sufficiently long to extend to very low frequencies. It is the recommended method for free-field reference measurements.

3. *5ms Half-Hamming (in-situ, non-standard)*

This option is similar to the windowing of the active response as described in section 7.5.2, and can be used for in-situ measurements where a reference impulse response is measured separately. This can typically be done in concert hall measurements where a close range measurement across an empty stage or to a front seat row can provide a better reference than the actual measurement positions.

Source distance options are:

compute from measurement (using initial time delay)
specify in meter (meas. initial time delay is not used);

1. *Compute from measurement (using initial time delay)*

For this option the distance is detected and computed separately from the various reference responses, in the same way as described in section 7.5.2. The temperature can be changed in the dialog as well, and will influence the speed of sound and thereby the distance. It should be noted that the distance to the acoustic centre of the source will be somewhat longer than the distance detected to the nearest loudspeaker face, which might influence close measurements. It is required that the initial time delay is correct, and the measurements must have been made with a feedback loop.

2. *Specify in meters (meas. initial time delay is not used)*

This option should be chosen when the reference measurement distance is known, and the same for all reference responses in the set.

7.6 Lateral parameters

Calculation of lateral parameters requires microphone option omni-directional and figure-of-8, and both primary and secondary impulse response specified. It is further assumed that the omni-directional and the figure-of-8 microphones have the same sensitivities and are properly equalized with respect to each other, in the direction of maximum sensitivity. The two impulse responses should be measured in the same position, if possible simultaneously. The figure-of-8 should have one of the null-directions pointing either towards the source or along the long axis of the room, see ISO 3382. It is also extremely important that they are mutually synchronized, with matching absolute time scales. This can for instance be accomplished measuring with a feedback loop, providing responses with a true initial time delay.

The early lateral energy fraction is

$$\text{LF} = \frac{\int_0^{t_e} h_L^2(t) dt}{\int_0^{t_d} h^2(t) dt} \quad (10)$$

where $h_L(t)$ is the impulse response measured with a figure-of-eight microphone with one of its nulls pointed towards the source. The time limits are t_d , the direct time, and t_e , the early time. LF is normally used for music application, with direct time 5ms and early time 80ms. The limit 5ms is introduced to make sure the direct sound is totally suppressed. The time limits can be changed in the boxes.

The late lateral strength is

$$\text{LG} = 10 \log \frac{\int_0^{t_e} h_L^2(t) dt}{\int_0^{\infty} h_{10}^2(t) dt} \quad \text{dB} \quad (11)$$

This is a fairly new parameter, not mentioned in the standards so far, but it is believed to correlate well with subjective envelopment⁵.

The upper integration limit in the numerator is taken as the truncation point determined in the decay curve formation for the lateral impulse response, or the end of the impulse response, according to the integration option specified. If compensation for truncated energy is calculated, this is taken into account as well to increase accuracy.

The reference energy in the denominator is calculated just as for strength, the same options govern the calculation, and similar comments on accuracy apply, see section 7.5.

7.7 Interaural crosscorrelation

Calculation of the binaural interaural crosscorrelation parameters are normally done from a binaural pair of impulse responses, measured either with a dummy head or microphones in or close to the ears on a real person. It requires microphone option **Binaural pair**. (The microphone option **Closely spaced omni-directional pair** can also be used in the current version of WinMLS, it provides exactly the same processing.) It is not important that the responses are mutually synchronized. The primary response is taken as the left side and the secondary as the right side.

The interaural crosscorrelation function is defined as

⁵ J.S. Bradley and G.A. Soulodre, "Objective measures of listener envelopment," *J. Acoust. Soc. Am.*, vol. 98, pp. 2590-2597 (1995).

$$IACF(\tau) = \frac{\int_{t_1}^{t_2} h_L(t) h_R(t + \tau) dt}{\sqrt{\int_{t_1}^{t_2} h_L^2(t) dt \int_{t_1}^{t_2} h_R^2(t) dt}} \quad (12)$$

where $h_L(t)$ and $h_R(t)$ are the left and right ear impulse responses respectively. The interaural crosscorrelation coefficient is taken as:

$$IACC = \max |IACF(\tau)|, \quad -1\text{ms} < \tau < 1\text{ms} \quad (13)$$

WinMLS calculates three versions of the IACC; total with time limits $t_l=0$ and $t_2=\infty$, early with time limits $t_l=0$ and $t_2=80\text{ms}$, and late with time limits $t_l=80\text{ms}$ and $t_2=\infty$. The time limits t_2 for IACC early (early time) and t_l for IACC late (late time) can be changed in the boxes.

The upper integration limit $t_2=\infty$ for IACC total and late is taken as the truncation point determined in the decay curve formation for the two impulse responses separately, or the end of the impulse responses, according to the integration option specified.

The calculation of IACC is substantially more time consuming than the other parameters, even if it is implemented using a fast frequency domain algorithm. This is especially true for the late and the total versions. If not explicitly needed, these parameters should therefore be excluded from the calculations.

7.8 Stage parameter

Support is defined as⁶:

$$ST1 = 10 \log \frac{\int_{t_d}^{t_l} h^2(t) dt}{\int_0^{t_d} h^2(t) dt} \quad \text{dB} \quad (14)$$

where it is assumed that the response is measured with an omni-directional microphone at a distance of 1m from the source. The time limits are t_d , the direct time, t_e , the early time, and t_l , the late time. ST1 is defined with direct time 10ms, early time 20ms and late time 100ms. The time limits can be changed in the boxes to provide other versions of the parameter.

7.9 Speech transmission indices

The speech transmission index (STI) and the rapid version (RASTI) are single number measures based on both impulse response and signal-to-noise ratio in the room or speech transmission system. The standard IEC-60268-16⁷ describes the measures, which are based on weighted sums of modulation transfer function (MTF) values. The weights currently used in

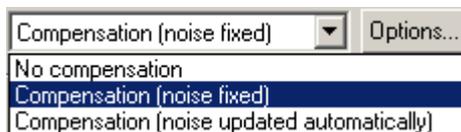
⁶ A. C. Gade, "Investigations on musicians room acoustic conditions in concert halls, II: Field experiments and synthesis of results," *Acustica*, vol. 69, pp. 249-62 (1989).

⁷ IEC 60268-16: 1998, *Sound system equipment – Part 16: Objective rating of speech intelligibility by speech transmission index*.

WinMLS are from the original definitions of STI and RASTI⁸. The calculations are performed on the primary impulse response.

In the calculation of the MTF the impulse response is taken up to the truncation point determined in the decay curve formation, or the end of the impulse response, according to the integration option specified. The MTF is determined by FFT of the filtered and truncated impulse response, and the values at the exact 1/3-octave modulation frequencies specified are found by linear interpolation. This is a fast and very accurate method for all but extraordinary short impulse responses, and for these the modulation reduction for speech transmission is negligible anyway.

If the background noise is low it will not affect the speech intelligibility. However, the background noise should be taken into account if the speech-to-noise ratio is less than 15 dB in the octave bands of interest. This is done in the compensation settings found in **Room Acoustics->Parameter Settings...** and explained below



1. *No compensation*

For this option the background noise is *not* taken into account. The calculations are based solely on modulation reduction caused by the impulse response, thereby assuming that the speech-to-noise ratio is less than 15 dB in the octave bands of interest. This is most common in room acoustics where the effects of reverberation and reflections usually dominate.

2. *Compensation (noise fixed)*

This option takes the background noise and assumed speech spectrum into account.

The **Options...** button opens up a new dialog box where the necessary data can be entered as shown below.

Octave band corrections [dB]							
	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz
Noise:	65	57	47	38	30	25	19
Speech:	67	67	65	58	52	48	40

This includes the background noise and the assumed speech level on the microphone, both expressed as octave band sound pressure levels in dB, for the octave bands that are included in the STI calculation. For RASTI only data from the 500Hz and 2kHz bands are used. The background noise can be measured with WinMLS if the input chain (including the microphone) is calibrated. Alternatively a sound level meter may be used to measure the background noise.

3. *Compensation (noise updated automatically)*

This option is similar to the above, except that the Noise octave band corrections shown in the figure above are computed and updated automatically from the tail of the

⁸ T. Houtgast and H.J.M. Steeneken, "A review of the MTF concept in room acoustics and its use for estimating speech intelligibility in auditoria," *J. Acoust. Soc. Am.*, vol. 77, pp. 1069-1077 (1985).

measured impulse response. In order to do this, the input chain (including the microphone) must be calibrated. The background noise is computed from the last 10000 samples of the tail. Make sure that the measured impulse response has decayed sufficiently.

Please note that this option should not be used on measurements performed using averaging since the averaging will reduce the effect of the background noise.

Therefore use 1 averaging, and an impulse response length larger than 1 second (in a reverberant room, larger than 2 seconds).

There is no provision for displaying the MTF itself in the present version of WinMLS. However, the last calculated MTF is dumped to file mtf.txt in the SetupsPostProcessing subdirectory in the WinMLS program directory, and can be viewed in a text editor.

7.10 Source to receiver distance

The source to receiver distance is estimated based on the initial time delay detected from the primary impulse response, and the speed of sound, as described in section 7.5.2. For the initial time delay, and thereby the source to receiver distance estimation, to be correct, the measurement must be performed with loop-back or a synchronized sound card. Even if this is the case, there will be an extra delay because of the sound card delay. This is set as 40 samples as default (set in **Measurement->Sound Card Settings...**). However, if measurement system correction is used, this delay will be corrected for (set with the  button in the **Measurement Settings Toolbar**).

The source to receiver distance is always output in the result grid or file, and is used in some of the reference level calculation procedures described in section 7.5.