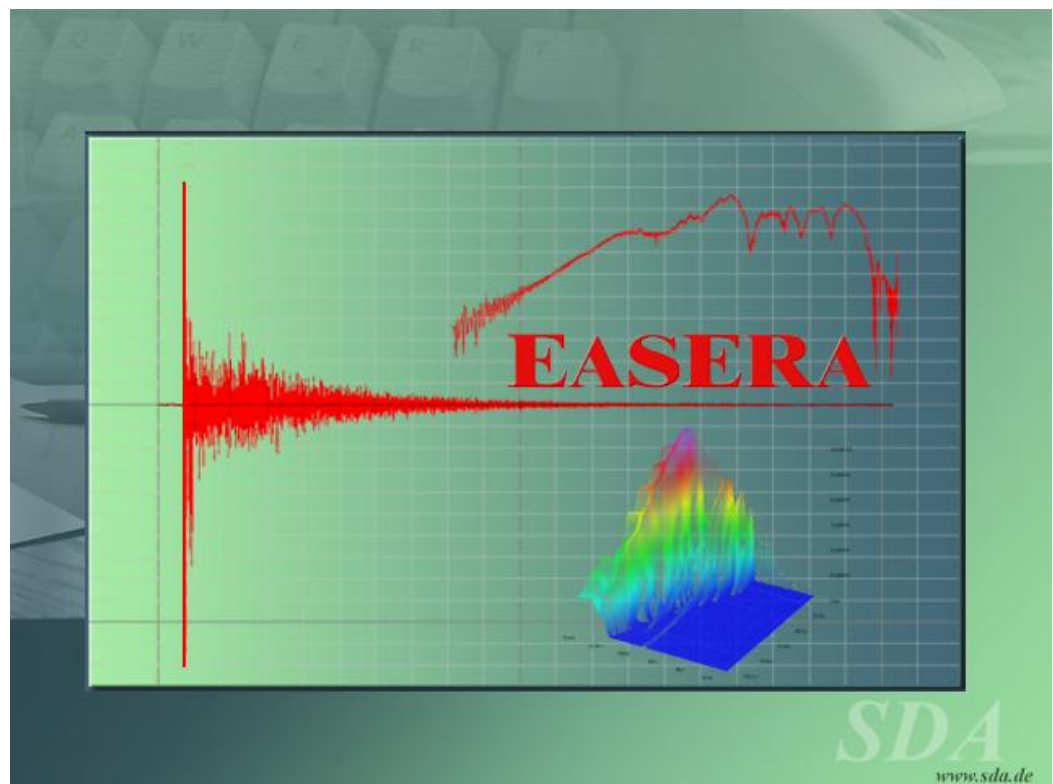


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EASERA 1.2

# EASERA Tutorial



By AFMG Technologies GmbH, Berlin.

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# Overview

## Preface

The purpose of this document is to provide you a working knowledge of the program and its many features and capabilities. The Tutorial Sections that follow will lead you through most of the important EASERA / EASERA Pro features, including measurements and evaluating measured data files.

**NOTE: Please do not skip any part of the tutorial. The time you spend working on the various exercises in the Tutorial will save you hours and hours of time later on. Even after you complete the tutorial you will find it helpful to refer back to it from time-to-time. It contains a wealth of helpful hints and shortcuts. If you don't have time now to do all the Tutorial exercises, at least scan through the Tutorial to become acquainted with all the features of EASERA / EASERA Pro. Then, use the manual as a reference guide when you need guidance in how to accomplish specific tasks.**

If you have further questions, please check the help files for EASERA / EASERA Pro, the appendix document for EASERA / EASERA Pro and refer to the textbooks and papers listed at the end of this tutorial. In addition please visit our dedicated EASERA website <http://easera.afmg.eu> and the AFMG internet forum <http://www.afmg-network.com/> as well as the website of your EASERA distributor:

- Worldwide distribution by AFMG Technologies: [www.afmg.eu](http://www.afmg.eu)
- Educational version through AFMG Foundation gGmbH: <http://www.afmg-foundation.com/>

## EASERA vs. EASERA Pro

EASERA can be purchased in two versions, namely as the standard version *EASERA* and the professional version *EASERA Pro*. Compared to the standard version the professional version contains an extended set of functions that simplifies the evaluation of more complex measurements and allows advanced measuring methods to be used.

Also, EASERA / EASERA Pro can be extended by several modules, namely the *TDS* module, which adds Time Delay Spectrometry measuring and processing capabilities, the *MLS* module, which enables Maximum Length Sequence measurements, and the *Polars* module, which adds functions for automated measurements. (Note that EASERA Pro also includes the *MLS* module.)

**Please note, the following Tutorial has been written with both the EASERA and the EASERA Pro user in mind. However, in several places functions are introduced which are only available in EASERA Pro. Even if you do not own EASERA Pro we encourage you to read these sections for a better understanding. If the feature set of EASERA does not seem to meet your needs, please consider licensing EASERA Pro instead.**

The following list gives an overview of all functions that EASERA Pro or the additional program modules add to the standard EASERA version (This list is based on software version 1.2. It is subject to change.).

## ***Measuring Functions***

EASERA Pro adds:

- Sample rates higher than 48kHz
- Measurement configurations using more than 2 channels
- Creating and loading customized excitation signals
- Editing sequences for automated processing of measurements

TDS Module adds:

- TDS sweep signals and measurements

EASERA Pro or MLS Module add:

- MLS excitation signals, MLS measurements and Hadamard transform

Polar Module adds:

- Automated measurements for sequential or balloon measurements
- Remote control of supported external devices, such as robots and turntables

## ***Real Time Analysis (Live)***

EASERA Pro adds:

- Tolerance mode to detect sound events and save them to a file automatically
- Sample rates higher than 48kHz

## ***Post-Processing (View & Calc)***

EASERA Pro adds:

- Editing functions, such as
  - o Remove DC, Set To Value
  - o Add Value, Subtract Value, Multiply By Value, Divide By Value, Scale To
  - o Square, Square Root, Inverse, Power
  - o Custom Smoothing, Expand/Shrink, Change Length
  - o Divide, Subtract, Multiply Measurement Files
  - o Remove Air Absorption, Change Sample Rate
- Distortion Analysis
  - o Harmonics
  - o Relative Spectra (K2 to K7)

- THD

TDS Module adds:

- Software-based TDS post-processing and frequency response analysis

Note:

The following Editing functions are available in the standard version of EASERA:

- Filtering
- Windowing
- Averaging Measurement Files
- Adding Measurement Files
- Cyclic Move
- Undo / Redo
- Copy / Paste



# **Lesson 1: Evaluating a Measurement**

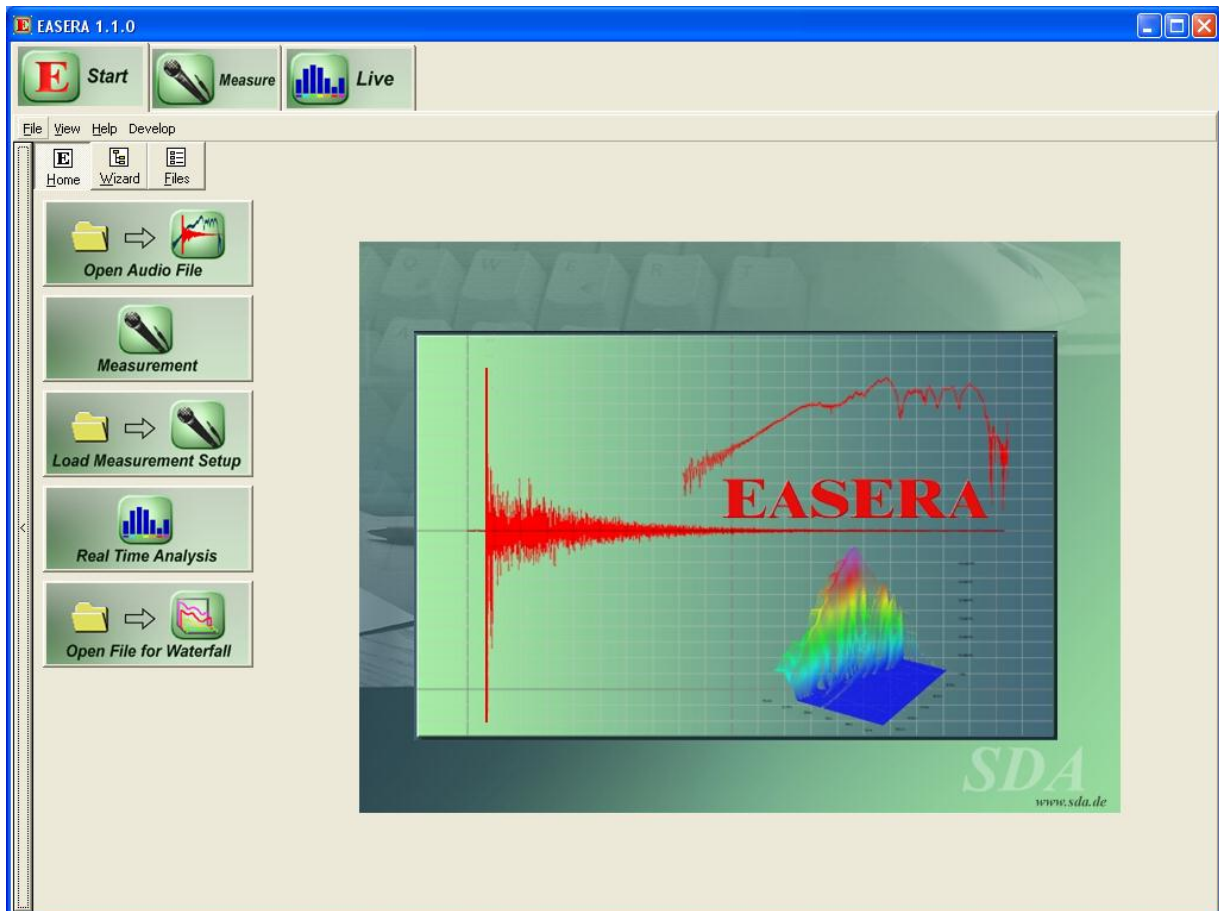
In our first lesson we are going to open an existing file to perform several evaluations with it.

### Starting EASERA

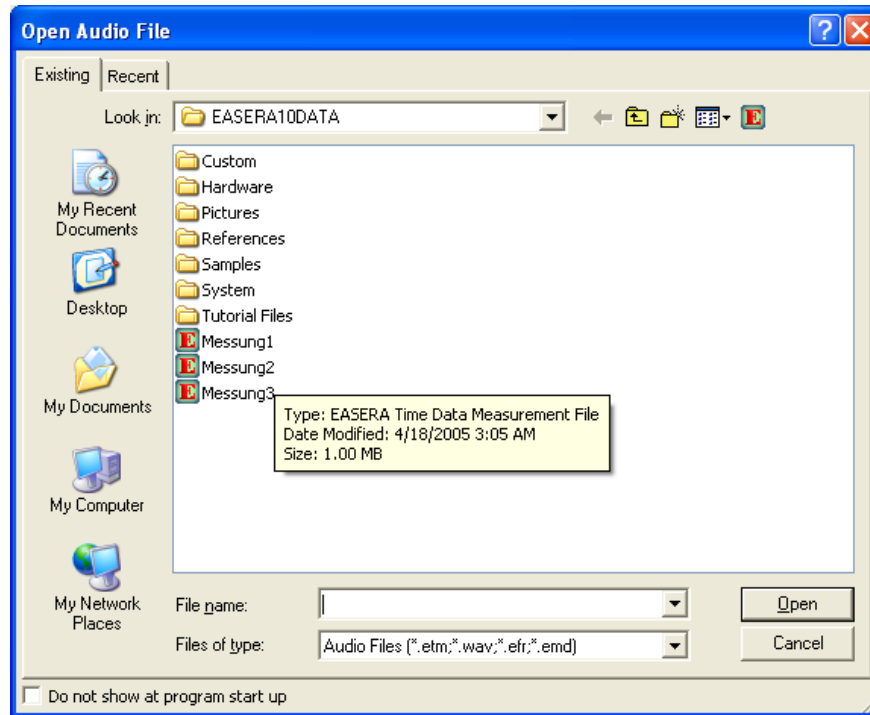
After Installation of EASERA you should have the following symbol on your desktop:



Start EASERA by double clicking on this symbol (alternatively by selecting *AllPrograms\AFMG\EASERA\EASERA* from the Windows Start button). You then will see the EASERA *Start* page.



On the left side of this starting screen you find five big buttons, the upper one of which is labeled *Open Audio File*. By clicking on this button with the left mouse button you will see the dialog box for opening a file.



This dialog comprises two pages:

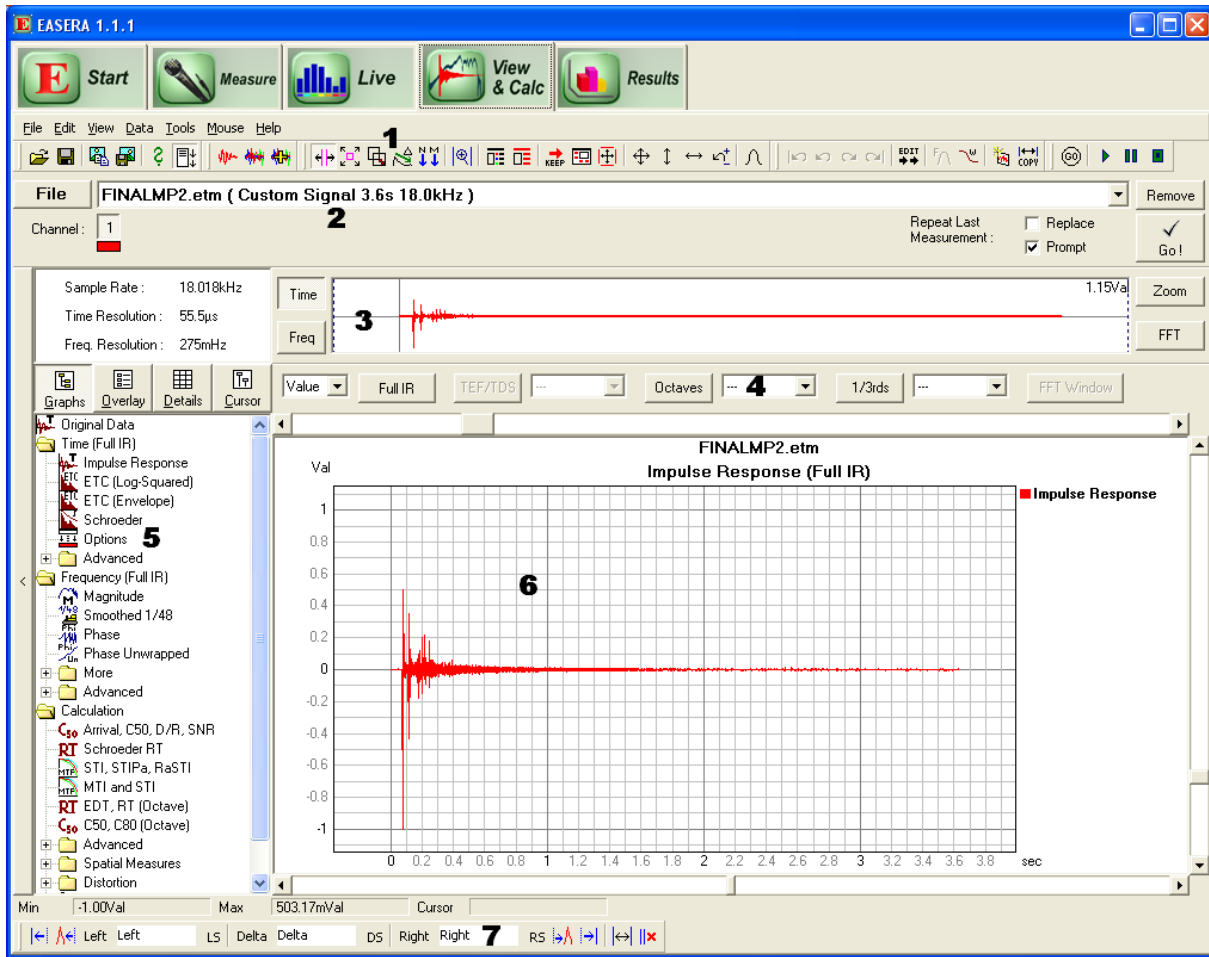
- Under *Existing* you find a tree structure of your hard disk, similar to that of the explorer in Windows.
- A list of the files you have recently used can be seen under *Recent*.

Choose the file FINALMP2.etm from the *Tutorial Files* directory under *EASERA10DATA* and then click on *Open*.

The file FINALMP2.etm is an impulse response which was formerly measured in a foyer of the Berlin Fair by means of the DOS measuring system MLSSA. This explains the sample rate of only 18 kHz, as we will see later.

## The View & Calc page

After opening the file, EASERA will automatically change to the *View & Calc* page. On this page you find many options which make it seem perhaps a bit overwhelming at first. It is, however, structured in a very logical way so that you will soon be able to manage, as you can see in the following picture:



- (1) At the top you have – as is the case with most Windows programs – a menu bar and a button bar. Immediately above the menu bar are large buttons that allow you to switch between the various pages in EASERA.
- (2) Below these you have the files area where you may choose between the existing files, add new files or initiate a new measurement.
- (3) Next is the visualizing area where the chosen file is shown as either a time or a frequency diagram.
- (4) In the filter area it is possible to choose octave or third-octave filters as well as a windowing of the data.
- (5&6) The largest space has been reserved for the evaluation area: On the left-hand side (5) in the navigator page one can choose under *Graphs* which diagram (6) is to be shown on the right-hand side.
- (7) The bottom lines show some information and provide some marker functions.

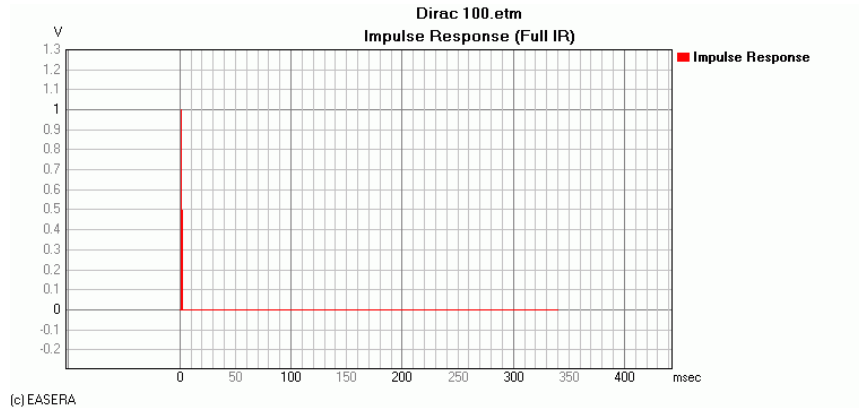
We are now going to have a look at some of these areas.

## The impulse response

The first diagram shown is the impulse response. An impulse response is the system behavior resulting from a minimally short excitation impulse (Dirac). The behavior of a linear system is fully described by such an impulse response and all the other evaluations can be derived from it.

Real systems have no ideally linear system behavior: Thermal noise, random pick-ups of disturbing sources, distortions and other influences make their real behavior deviate from an ideal system behavior. These deviations are not described by the impulse response.

A minimally short excitation impulse would look as follows:



In EASERA there is no impulse of this kind used for excitation, but the impulse response is derived from more suitable excitation signals which we will look at later on.

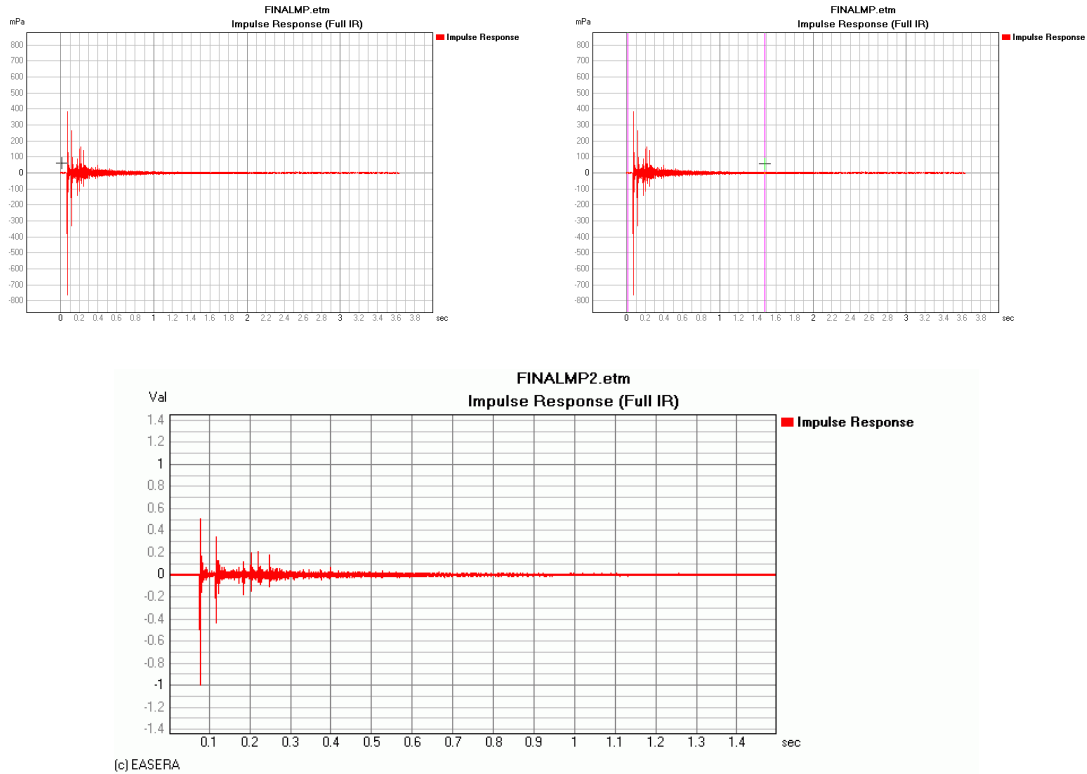
If we look at the impulse response IR of FINALMP2.etm, we see that the first impulse of the IR deviates downwards (with the MLSSA measurement the standard polarization of the measuring chain was not observed) and that it "decays" with some room reflections and a certain reverberation (as is typical for a room acoustical measurement).

## Zooming in

The interesting part of the impulse response takes place during the first 1.5s and we are now going to zoom into this period. To do this we make sure that we are in the Zoom X mode:



In the diagram we select the range from 0s to 1.5s by dragging the mouse accordingly. (We move the mouse to the X position of 0, click with the left mouse button and keep it pressed down, drag the mouse to the X position of 1.5s and release the mouse button.)

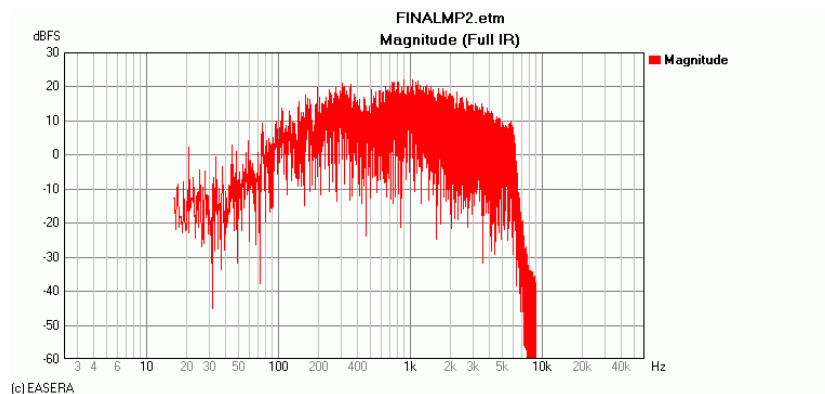


We now see clearly the main impulse, a pronounced first reflection after about 40ms, four succeeding distinct reflections and a decaying reverberation.

One could now zoom further into the diagram by selecting corresponding ranges with the mouse. For reverting to the full impulse-response view, just double click in the diagram. (Alternatively choose the menu item *View/Full* or the corresponding button in the button bar.)

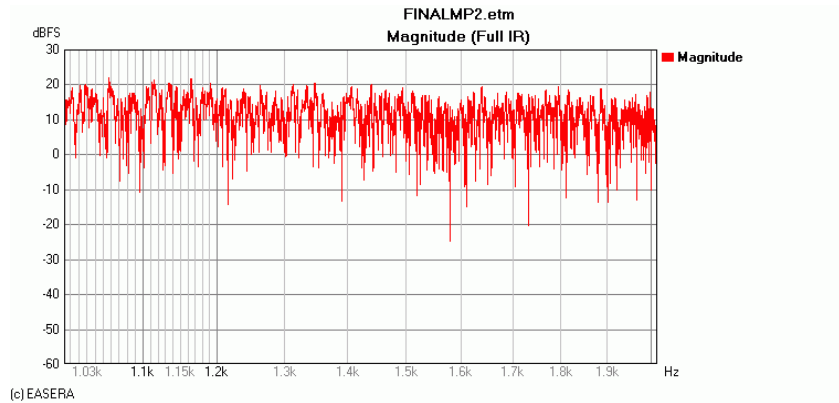
## Frequency response

We now want to have a look at the frequency response of this room at the microphone position. To do this we click in the *Graphs* navigator page on the *Magnitude* item.



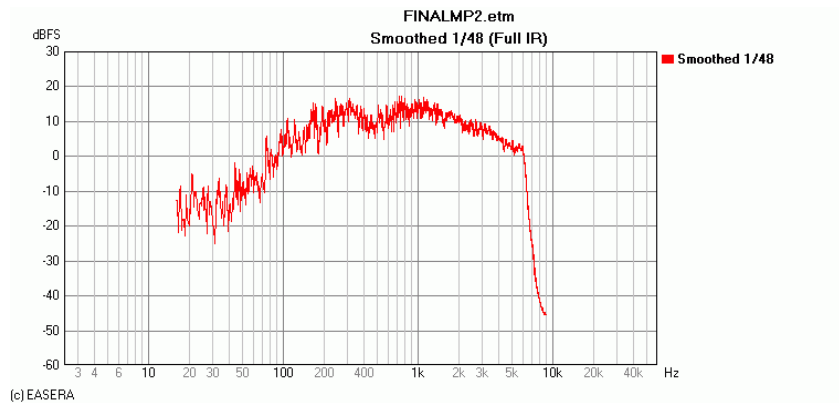
First one notices that between 6 and 9 kHz the frequency response drops by approx. 70 dB. In an overview window on the left side of the visualizing area *Sample Rate* of 18 kHz is given – so above 9 kHz the measurement does not show any data.

Moreover it is noticeable that the curve becomes rather "scattered" with rising frequency. The reason becomes evident when we zoom into the range between 1 kHz and 2 kHz (by selecting the corresponding range with the left mouse button):

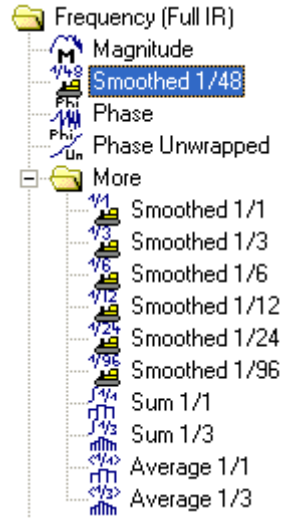


We have here a "comb-filter-like" frequency response, i.e. a series of very narrow dips in the frequency response, which are caused by the interference of the many room reflections. By double clicking in the diagram we return to the complete frequency response diagram.

It is certainly difficult to gather significant information from such dense displays. This is why frequency responses are usually smoothed. Click in the *Graphs* navigator page on *Smoothed 1/48*, in order to apply a smoothing to 1/48<sup>th</sup> octave.



This already looks more distinct, but could still be too detailed if you wanted to adjust this frequency response by means of an equalizer. The folder *More* offers you a list of further smoothing intervals:

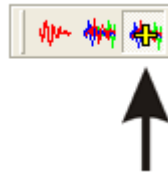


If you work with a third-octave band equalizer, smoothing to 1/3 octave could be reasonable. This smoothing is generally sufficient, since it shows where possible dips or peaks are still contained in the frequency response and would be audible, as previous experience has shown. Smoothing to one musical half-step would correspond to 1/12 octave.

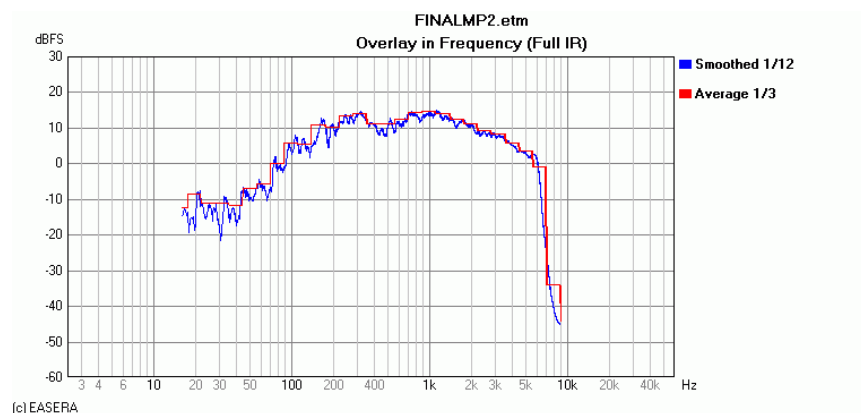
## Overlay

We are now going to create an overlay, i.e. show several curves in one common diagram. To do this we choose first the frequency response with smoothing to one-twelfth octave (*Smoothed 1/12*).

Then we activate the function *Add To Overlay*:



This function is also available in the *View* menu. Now select *Average 1/3* from the *More* folder to add the average level of every third-octave band and obtain the following diagram:

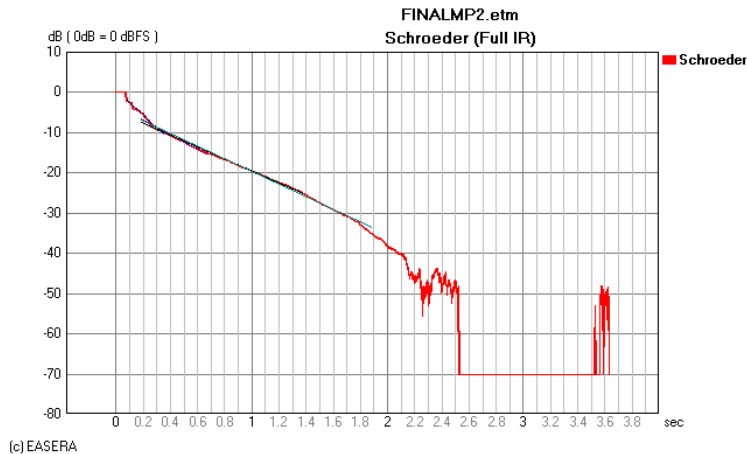




Then deactivate the function *Add To Overlay* by selection the button or menu item again.

## Reverberation time

Next we want to determine the reverberation time and click in the *Graphs* navigator page on *Schroeder RT* (in the *Calculation* folder midway down the list). The diagram now shows the Schroeder plot of the impulse response, i.e. the existing energy integrated from back to front (backward integration):



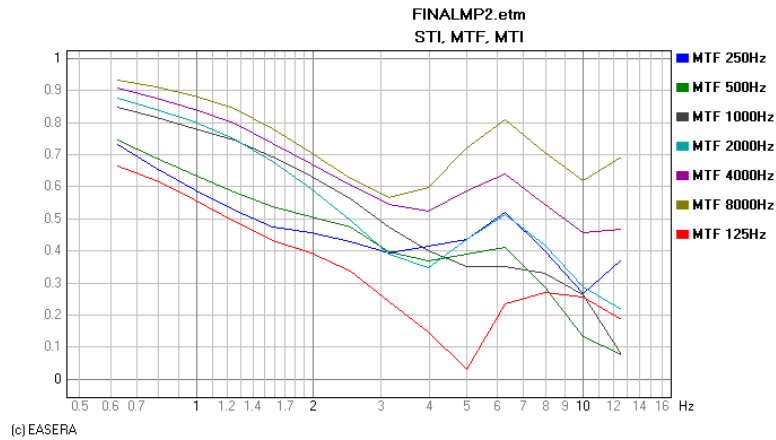
Simultaneously the navigator changes to the *Details* page where the different values for the broadband reverberation time are shown:

	RT	
EDT	1.92	s
T10	3.23	s
T20	4.00	s
T30	3.83	s
Time From	0	s
Time To	0	s
Noise Comp.:	<input type="checkbox"/> ON	

The differences between EDT, T10, T20 and T30 as well as any other peculiarities of the reverberation time (noise compensation, octave-band reverberation times) will be discussed later.

## Speech intelligibility

To determine the speech intelligibility we click in the navigator page *Graphs* on the entry *STI*, *STIPa*, *RaSTI*. The diagram now shows the modulation transfer functions:



On the *Details* navigator page we find the calculated values:

STI	0.529
AICons [%]	9.687
STI (Male)	0.539
STI (Female)	0.554
RaSTI	0.505
Equiv. STIPa (Male)	0.554
Equiv. STIPa (Female)	0.566

The topic *Speech intelligibility* will also still be dealt with in detail, but not right away in the first lesson.

# Lesson 2: Live

Normally EASERA conducts measurements by emitting a stimulating signal, recording the response of the measured object and calculating from it the impulse response, from which all the other quantities will be derived thereafter. We will look at this in detail later.

Sometimes, however, one needs a permanently updated representation of the frequency response, the way one knows it from the old real-time analyzers (RTA). By means of such a representation it is for instance possible to check the signal chain before a measurement or examine the energy contents and the spectral components of ambient noise.

This real-time frequency analysis will be carried out on the *Live* page.

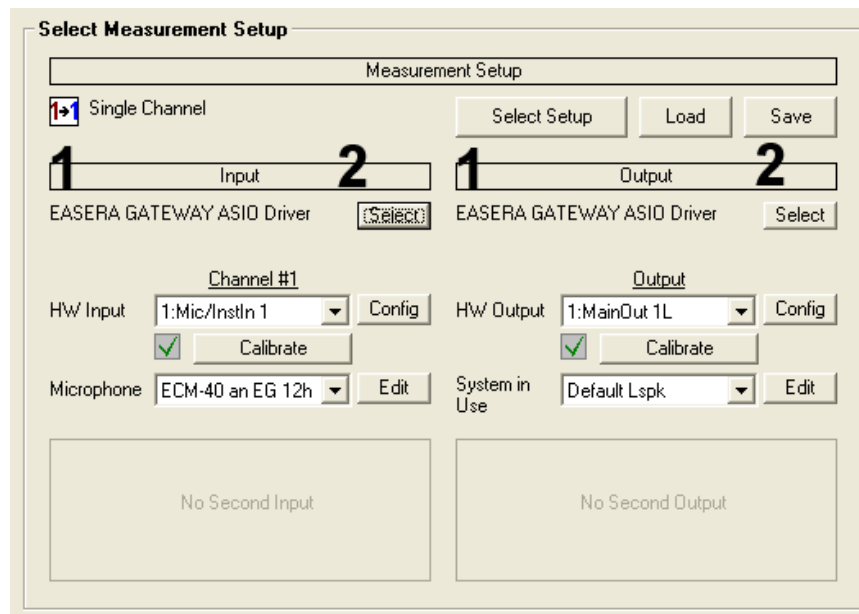
## Selecting the soundcard

Before changing to the *Live* page we want to first make sure that the correct soundcard has been selected and, if necessary, change to another one.

Pressing the *Measure* button at the upper edge of the window will change the view to the *Measure* page.

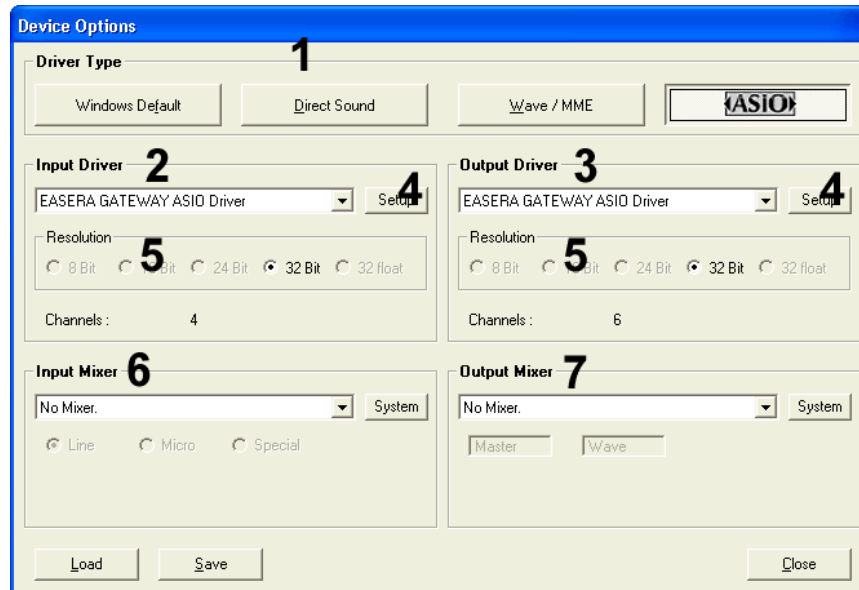


There we choose the *Select Measurement Setup* section:



In this section you find many option settings which shall not yet be dealt with. Directly under the captions *Input* and *Output* we find the presently selected driver (1), in this example the *EASERA GATEWAY ASIO Driver*. The driver of the soundcard reveals the type of soundcard used; in this case it is the *EASERA GATEWAY*.

In case the wrong soundcard was selected, click on the *Select* (2) button. This will open the *Device Options* dialog box shown below:



At the top of the dialog box you see four buttons (1) for the different driver types supported by EASERA:

- If the soundcard that is used provides an ASIO driver, you should use it by selecting the *ASIO* button, since it enables you to use more than two input channels.
- If there is no ASIO driver at your disposal, you should select the *Direct Sound* button.
- If this is not available either (or does not work stably) then select the *Wave/MMW* button.
- If EASERA does not work right away with any of these settings, then opt for *Windows Default*. EASERA then works with the Windows API functions and the soundcard included in the Windows system control.

Depending on the driver type chosen by you, the corresponding drivers that are installed will be at your disposal in the selection lists under *Input Driver* (2) and *Output Driver* (3).

With a mouse click on the *Setup* (4) button you open the corresponding dialog box of the soundcard driver for making any necessary adjustment, e.g. of the sample rate.

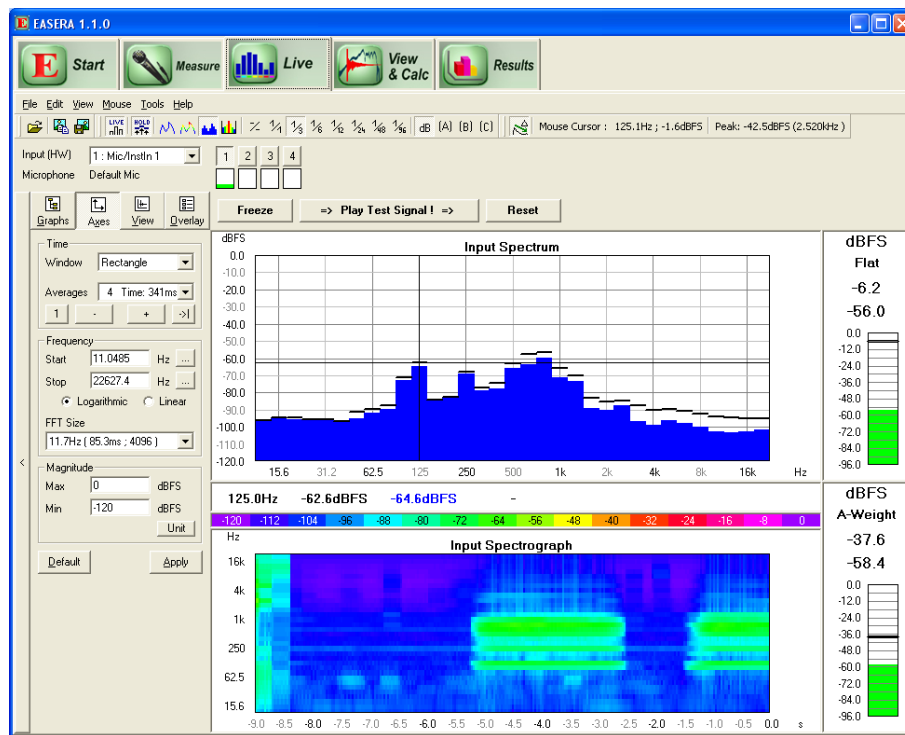
Under *Resolution* (5) you may adjust the bit resolution, provided the driver offers several options.

Under *Input Mixer* (6) and *Output Mixer* (7) you may choose if a mixer is to be used, and in the affirmative case, which mixers are to be used. We will look at this more closely later.

Click on the *Close* button to close the dialog and accept the settings.

## The Live page

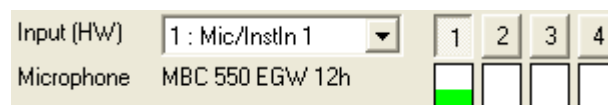
Let us now change to the *Live* page:



Under the menu and the button bar this page is subdivided into three segments for channel selection, diagram selection, and diagram view. Immediately above the menu bar are large buttons that allow you to switch between the various pages in EASERA.

### Channel selection

At the top there is the channel selection area where you may choose the channel to be displayed:

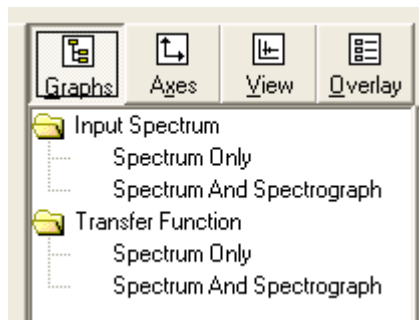


Below the buttons you find small level control indicators allowing you to see instantly the channels to which signals are applied.

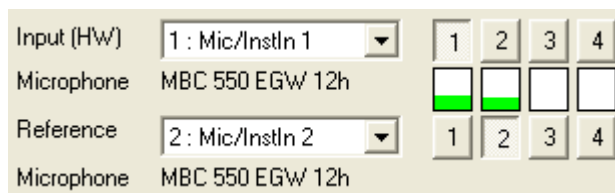
Depending on the hardware chosen there are up to 32 channels available simultaneously (in the EASERA Pro Version), or else 4 channels of the EASERA Gateway or 8 channels of the AUBION X.8. Normally these volume indicators show the color green for channels that have signals applied to them. If the level is getting close to the clipping limit, the color changes to yellow. When the color changes to red, the level has reached the clipping limit and you should reduce the level, otherwise the measurement will be corrupted by the influence of distortion products.

## Diagram selection

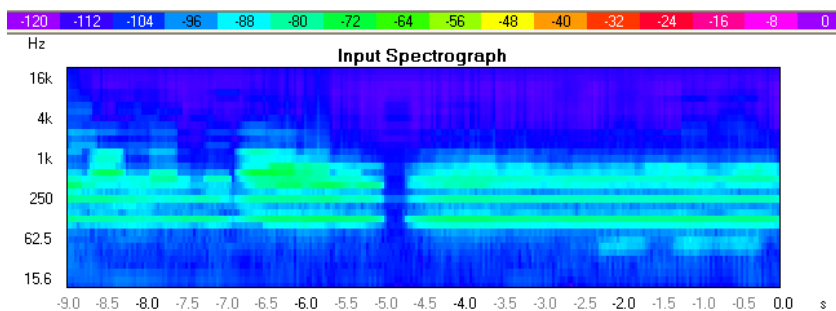
On the *Graphs* navigator page you may choose the diagram you wish to have shown. The available options are *Input Spectrum* and *Transfer Function*, each in the variants *Spectrum Only* and *Spectrum And Spectrograph*.



With *Input Spectrum* the chosen channel is subjected to a frequency analysis. With *Transfer Function* the ratio between one channel and the other is formed and this quotient (on the logarithmic dB scale as a difference) is subjected to a frequency analysis. To do this there is also a reference channel that needs to be selected:

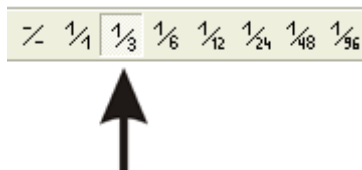


The frequency spectrum, i.e. the level vs. frequency, is always shown. An optional display is a spectrogram in which the behavior of the spectrum vs. time is shown. The level is indicated by an appropriate coloring using the legend:



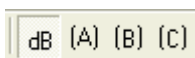
## Bandwidth

If you want to adjust a third-octave-band equalizer, you will most likely wish to work with a bandwidth of  $1/3$  octave. But EASERA also allows other bandwidths:



## Frequency weighting

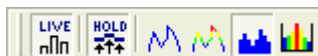
The signal measured can be shown in linear representation (using the *dB* setting) or with frequency weighting A, B or C according to DIN IEC 651.



The chosen frequency weighting is also used for the weighted level indication on the right-hand side.

## Presentation

Use the following buttons to specify how EASERA should display the frequency spectrum:



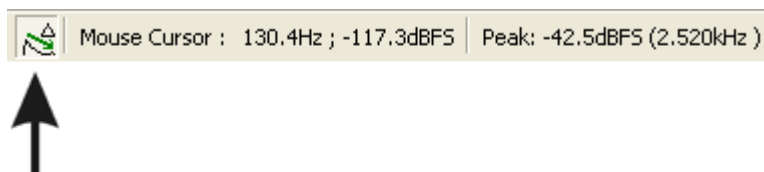
The first button (*Live*) is used to switch the real-time analysis on and off, and the second one (*Hold*) does the same for the peak-hold display. The peak-hold display shows the maximum peak level instead of the average value of the present level. This maximum is held for 1 second (see *Options*) allowing to see easily where the maximum peak level is positioned.

Use one of the following four buttons to choose the way in which the real-time analysis is to be displayed:

- Monochrome curve
- Multi-colored curve (uses color legend)
- Monochrome bars
- Multi-colored bars (uses color legend)

## Mouse Cursor

At the end of the button bar the position of the mouse cursor is indicated, converted into the units of the axial scales.



Use the *Peek* button to determine whether the effective level of the mouse position *y* for the frequency band located at the mouse position *x* is indicated. The “Peak” shows the maximum level of all frequency bands.



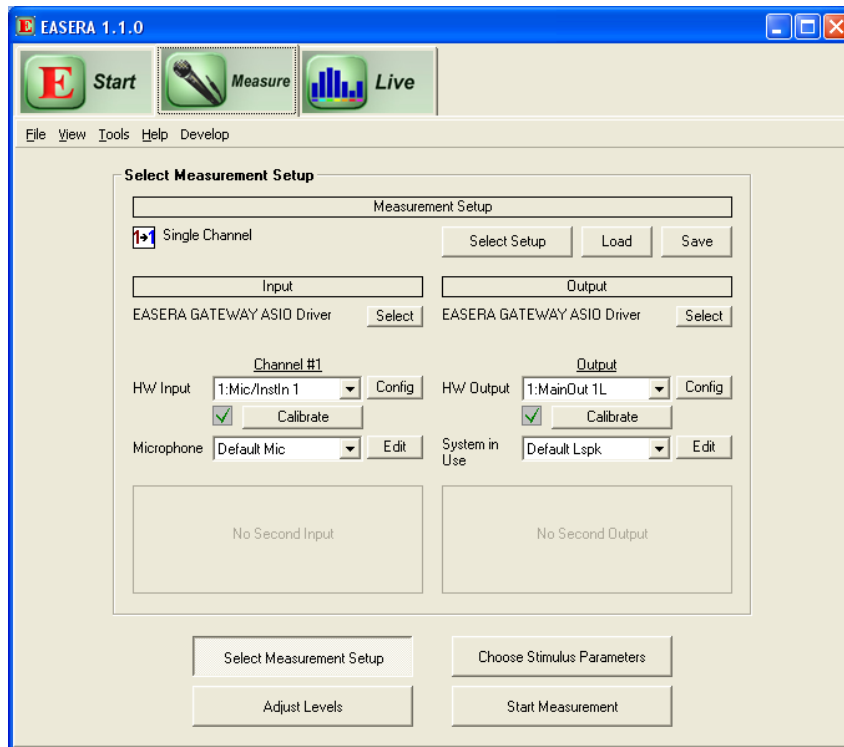
# Lesson 3: Performing a measurement

We are now going to see how a measurement is carried out. In order to do this it is assumed that the sound card has already been correctly selected, see lesson 2.

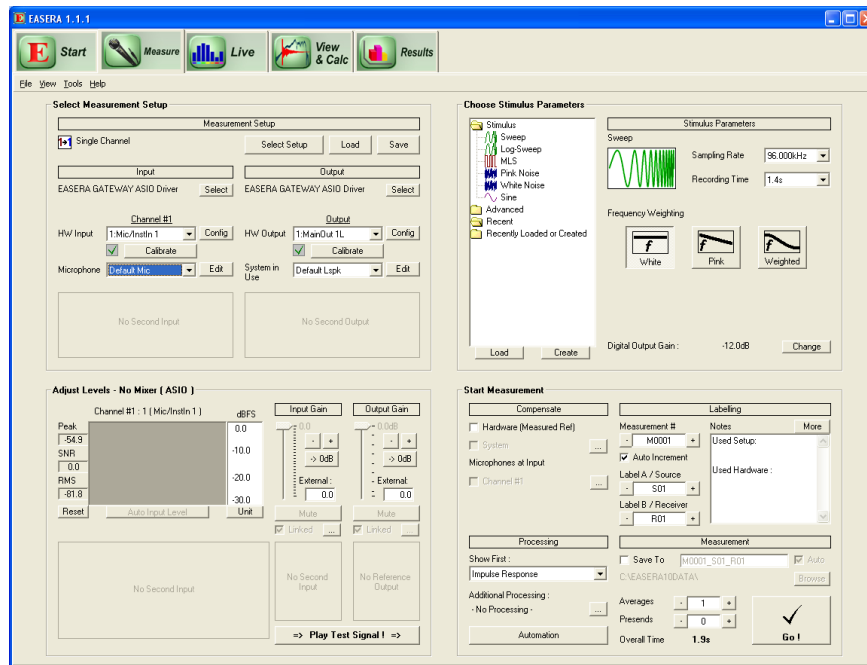
The first measurement is to be a kind of preparation. Connect a microphone to the input of the soundcard and a loudspeaker to the output. Place the microphone in front of the loudspeaker. Then select the *Measure* page.

## Selecting and calibrating a channel

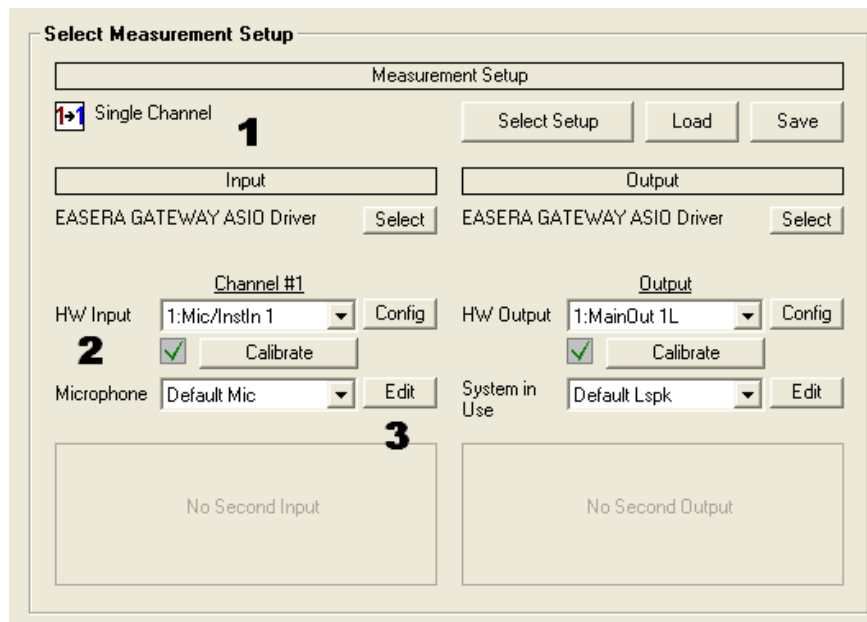
The *Measure* page contains four sections which can be selected using the buttons arranged at the bottom:



If the vertical screen resolution is at least 768 pixels high, then it is possible to show all four sections simultaneously - select *View/Full View* to show the page this way. (To accomplish this with the still very common resolution of 1024x768 pixels it is necessary to select the *Minimize Controls* checkbox under *Options (F9)/Miscellaneous/Appearance*.)



We begin with selecting the channel and go to the *Select Measurement Setup* section where we have already selected the soundcard:

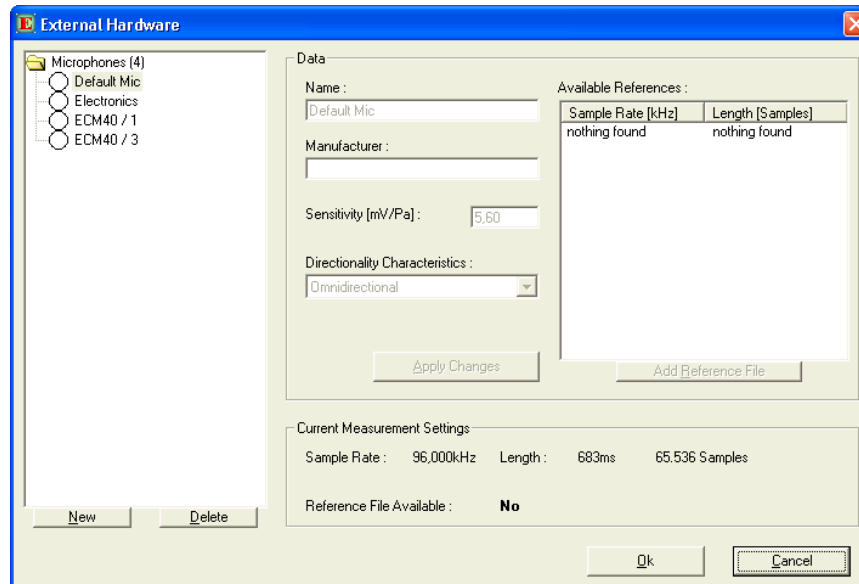


First we choose a *Measurement Setup* by clicking the *Select Setup* button. In this example we want to measure the frequency response using a microphone. Thus the number of input channels is one and we use *Single Channel* (1). EASERA allows using up to 32 input channels, but the converter hardware often only supports a smaller number.

In the *Input* area we can not only choose the AD/DA converter to be used (we have learned this already in lesson 2), but also assign hardware channels and calibrate microphones. You may store the calibration data of diverse microphones in the system so that you need only choose them when required (2).

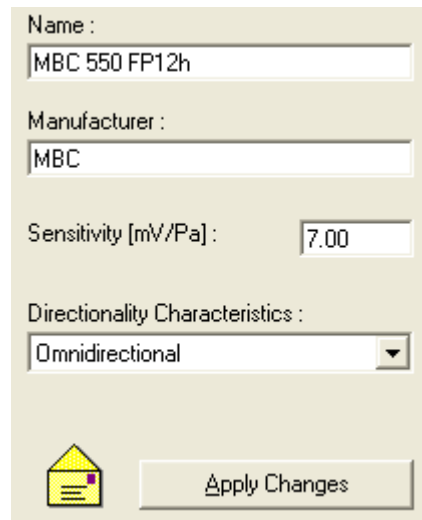
We now want to use an MBC 550 and enter its data into the system. This microphone is connected to channel 1 of the soundcard. Choose under *Channel #1* the channel 1 from the *HW Input* selection list.

Next we will create the new microphone. To do this select the *Edit (3)* button to open the *External Hardware* dialog box:



To create a new entry, click on the *New* button below the list of microphones. First we change the name from *New Mic* to *MBC 550 FP12h*. This means that the microphone is a type MBC 550 and connected to the EASERA GATEWAY with the input level set to the 12:00 h position.

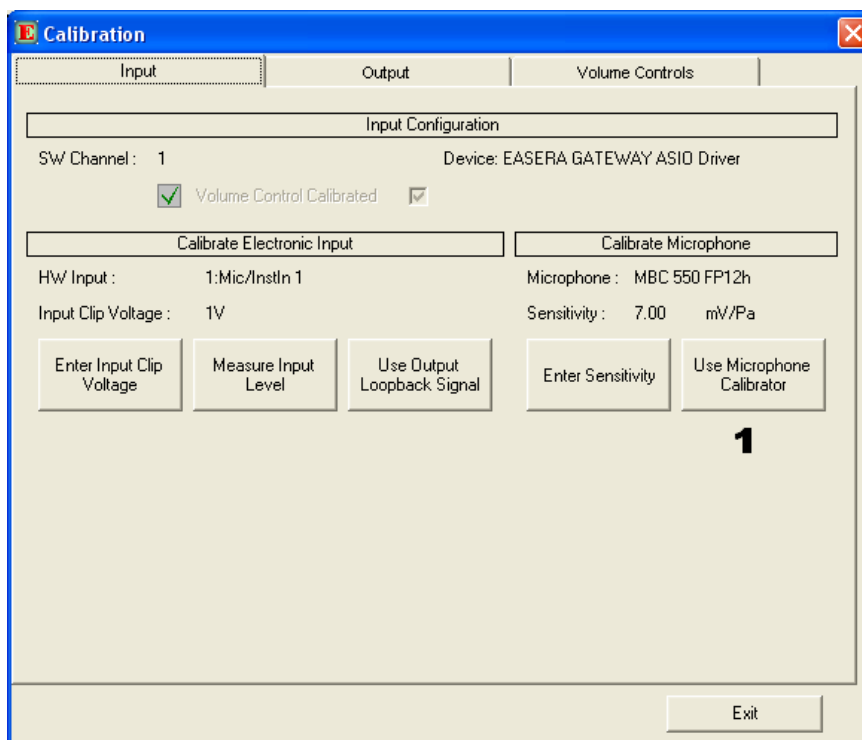
As *Manufacturer* we enter *MBC* and from the calibration diagram we take a sensitivity of 7.0 mV/Pa.



While making changes, an envelope appears to the left of the *Apply Changes* button to remind you that the data have not yet been stored. Click on *Apply Changes* so that the data are saved and the envelope disappears. You will find this symbol at many places in EASERA. Now choose the newly created microphone and close the dialog box with *Ok*.

## Calibrating the microphone

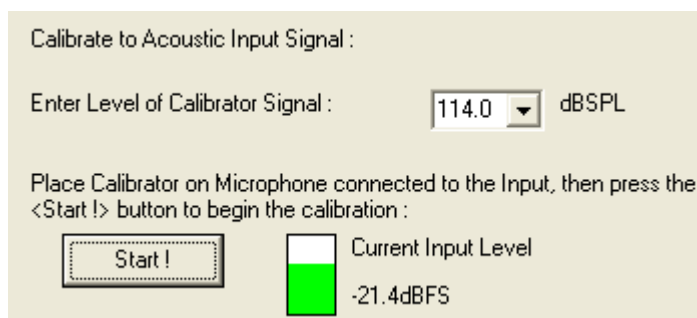
Now the microphone needs to be calibrated. Make sure that the new microphone has been chosen under *Microphone* and click on the *Calibrate* button to open the *Calibration* dialog box.



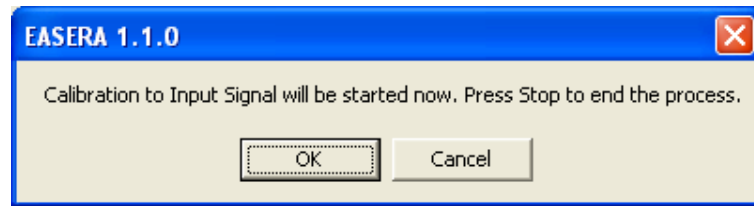
Here we want to calibrate the microphone (*Calibrate Microphone*) and use a sound level calibrator (*Use Microphone Calibrator*) (1). If such a device is not available, it is also possible to enter the sensitivity, if known.

As an expedient you may whistle a tone at a handbreadth distance and with medium loudness into the microphone, which will render about 100 dBSPL +/- 20 dB. (It is true that this is not very accurate, but certainly better than being confronted afterwards with sound pressure levels of 250 dBSPL. Even this would not have any influence on the qualitative behavior of the frequency response nor on room acoustical values like the reverberation time, for instance, but makes any expert flinch at first sight.)

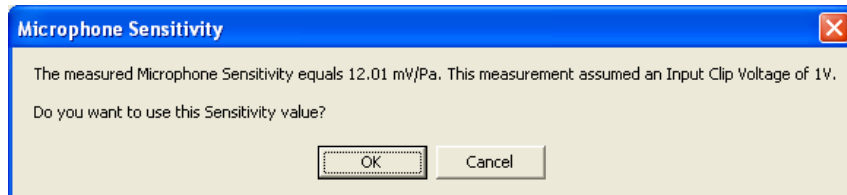
When clicking on the button *Use Microphone Calibrator* you open the following display:



First select or enter the sound pressure level for the calibrator and then click on *Start*. The following message box appears to remind you of the procedure:

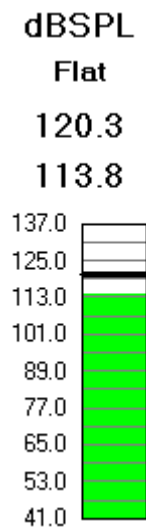


Now switch on the calibrator, connect it to the microphone and click on *Ok*. As soon as the display is stable terminate the measurement by clicking on *Stop*.



Now EASERA displays the result of the measurement. If it seems plausible to you, accept it by clicking *Ok*.

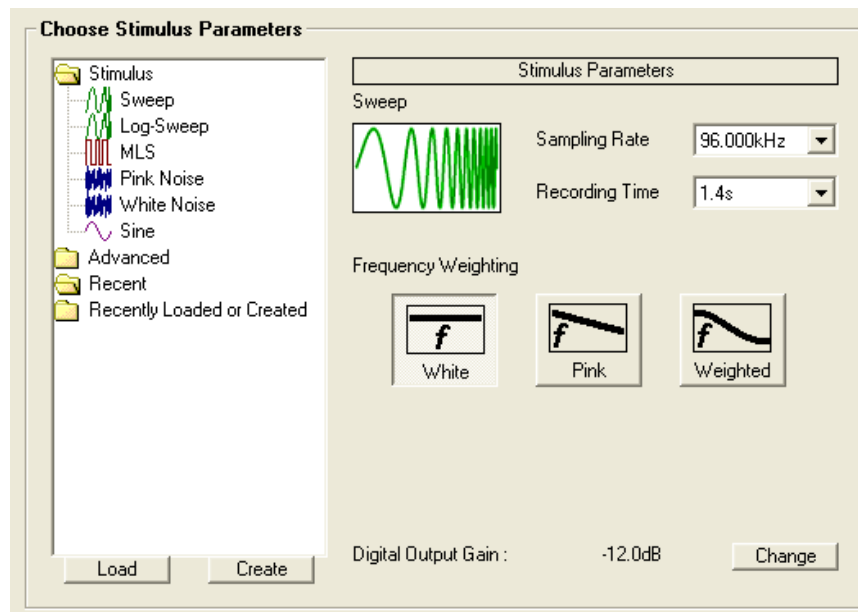
Since we should not blindly trust any measuring system with which we are not completely familiar, we go to the *Live* page to see what kind of levels we obtain here using the calibrator.



We see a peak level of 120.3 dB and a momentary level of 113.8 dB. Deviations of a few tenths of dB are not unusual, if we calibrate under the influence of environmental noise. Thus our microphone calibration is acceptable and the calibration of the output shall be postponed for the time being.

## Stimulating signal

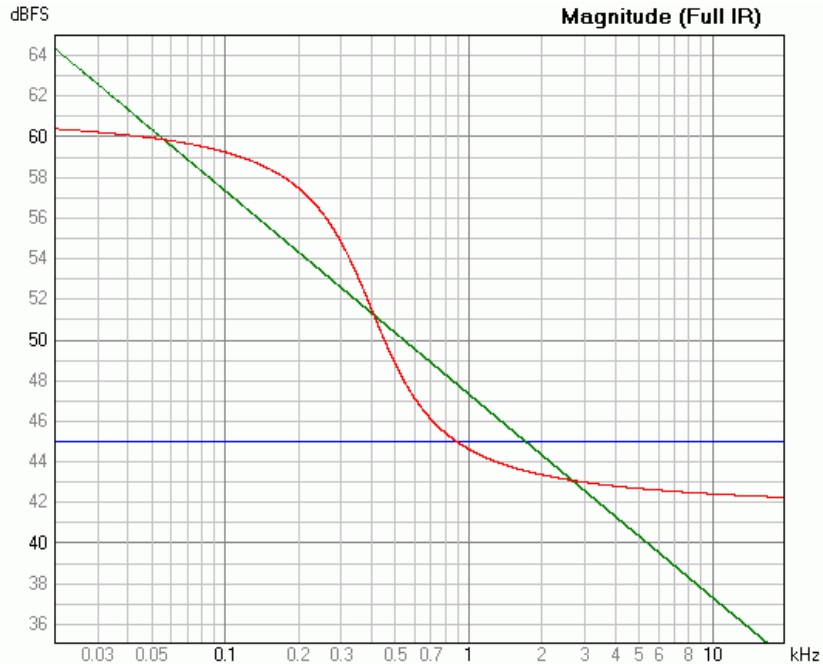
We now want to deal with the stimulating signal using the *Choose Stimulus Parameters* section:



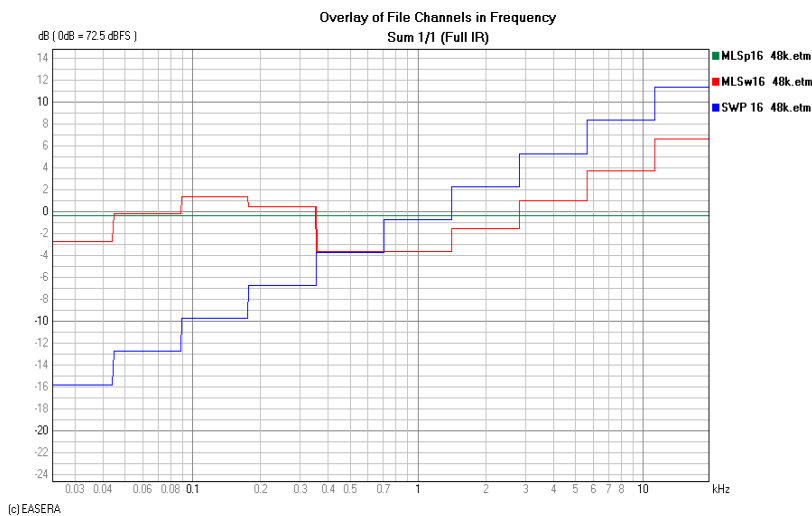
The question of which is the most appropriate stimulating signal gives rise to controversial arguments among experts. At this moment we do not want to discuss the pros and cons of the different stimulating signals, but wish to first see which options are at our disposal:

- Sweep, i.e. a sweeping sine signal
- MLS (Maximum Length Sequence)
- Noise
- Sine (individual pure sounds)

It goes without saying that the individual pure sound is not helpful for recording a broadband impulse response or frequency response. This leaves us with the first three options mentioned which are each available with three frequency weightings: *White*, *Pink* and *Weighted*:



*White* is a linear frequency weighting. The level is the same for all frequencies (see blue curve). This weighting is a bit unfavorable if the stimulating signal is to be reproduced by a multi-way loudspeaker: If one preferred to make the best of the load capacity of the woofer, the tweeter would be overloaded. If, on the other hand, the load capacity of the tweeter were considered to be most important, an unnecessarily poor signal-to-noise ratio in the low-frequency range would result. But all this is rather difficult for the practitioner to deduce from the above plot. In practice we are accustomed to considering the level conditions by frequency bands. This correlation is shown in the following diagram where one sees that white noise (see blue curve again) contains more energy in the upper frequency bands than in the lower ones.



In the linear frequency plot with *Pink* stimulus, the signal decreases towards the higher frequencies at a rate of  $1/f$ , i.e. with a drop of 3 dB per octave. This is represented by the green curve in both diagrams. The drop is easily seen in the linear plot. In the plot with octave bands, however, the  $1/f$  drop and the frequency-proportional summation of the band energies just compensate each other producing a frequency-independent behavior which is of great importance for the audio measuring technique. Thus one obtains with pink-



weighted stimulating signals comparable levels in the bass range as well as in the treble range. The signal-to-noise ratio towards the higher frequencies is reduced.

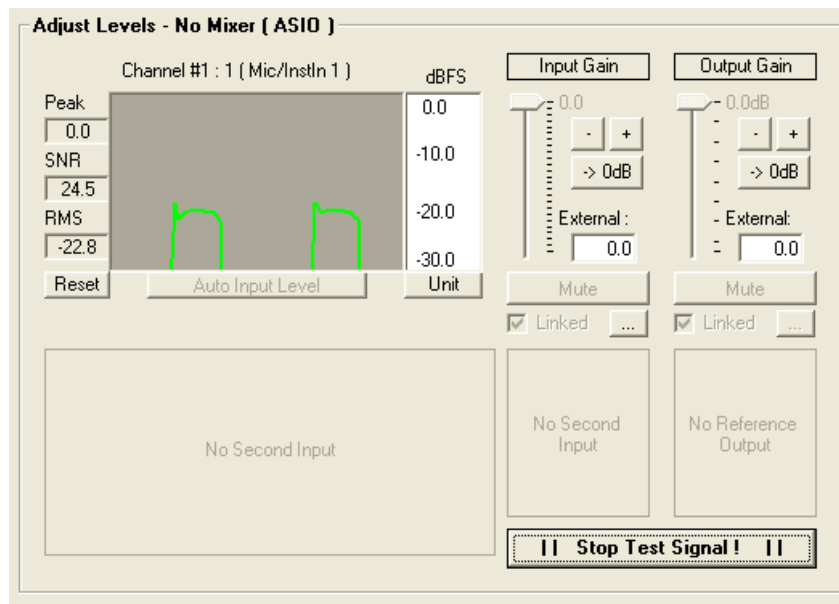
For making good use of the pink behavior and improving the signal-to-noise ratio for the high frequencies simultaneously, there also is the *Weighted* mode, where the difference between bass and treble in the linear plot amounts to only 18 dB between 20 Hz and 20 kHz, in the octave band plot to only 10 dB, and the mostly high-level mid-band range is even lowered a bit (5 dB).

Thus *Weighted* should be the best suited frequency weighting for most acoustical applications. Should it be necessary to quantify distortions, it is necessary to use a *Pink Sweep*.

For the measurement we are now going to set a *Weighted Sweep* as the stimulus.

## Checking the level

We now change to the *Adjust Levels* section. Here you can start the emission of the signal by clicking on *Play Test Signal*. Prior to this the output level controls should, however, be adjusted to *Zero* so that a signal that is too powerful cannot damage the loudspeakers and/or the ears. If there are no Windows mixers provided, as is the case here with the EASERA interface, this has to be done in the provided device mixer or directly at the hardware.



We now see the input level generated by the sweeps, here with the peak approx.  $-20$  dB below the clipping limit (FS, full scale). This means we would still have some reserve. As a rule it is recommended that the signal should not be more than  $-10$  dBFS, a limit where the curve would be shown in yellow (and in red after reaching the clipping limit).

If the level were too high, the *Input Gain* and/or the *Output Gain* could be adjusted downward. The faders arranged on this page control the soundcard mixers directly. But if we do not have any soundcard mixers controllable in the software domain (as is the case here with the EASERA Gateway) the faders cannot be moved so the hardware level controls need to be used.

Under *Choose Stimulus Parameters* one could, however, vary the *Digital Output Gain*. For sound devices like the EASERA Gateway where there are no software faders, it is recommended that the *Digital Output Gain* be set to between  $-14$  dB and  $-20$  dB.

## Calibrating the faders

A short excursion: If at this point we had a soundcard mixer at our disposal and would turn down the input level, the calibration would get lost. Since this would be annoying (especially if it would happen repeatedly), EASERA offers in the course of an overall calibration the option of calibrating also the soundcard mixer volume controls. Then you could also move the faders without losing the calibration.

As long as we have not calibrated the soundcard mixer volume controls, we should keep the fingers off all faders, or we would have to recalibrate afterwards.

## Starting the measurement

Now we change to the *Start Measurement* section:

Here we can adjust several details.

At the top left in the *Compensate* area it is possible to choose reference files that compensate for the frequency responses of the devices used. If, for instance, a room acoustical measurement is carried out with a loudspeaker, one could find out the frequency response of this loudspeaker first and take it into consideration with the following measurements. In the same way one could use the reference file of a microphone.

At the top right in the *Labeling* area it is possible to label the measurement and to enter related notes. If the *Auto Increment* box is checked, each measurement is assigned a consecutive number. For the time being we will not go into further detail.

After finishing the measurement the system automatically switches to the *View & Calc* page. At the bottom left in the *Processing* area, one may determine under *Show First* which diagram is to be shown first. The meaning of *Additional Processing* will be dealt with as soon as we know what is meant by *Processing*. The button *Automation* provides access to functions permitting the recording of balloon data (radiation behavior of a loudspeaker).

Select the *Save To* box in the *Measurement* area to automatically save the measurement under the given file name defined by EASERA.

## Averaging

Measurements are rarely carried out under optimum conditions. Often one has to deal with considerable ambient noise due to street traffic and/or work in the building. In order to achieve results of tolerable exactness under such conditions one falls back on the process of averaging: Under *Averages* one sets the number of measurements to be carried out. EASERA then automatically produces the average value of these measurements. With each doubling of the number of measurements the signal-to-noise ratio is enhanced by 3 dB. We will set *Averages* now to the value four.

With measurements using noise or MLS excitation, the room or system has to be prestimulated. This means that the stimulating signal is emitted at least once without being accompanied by a measurement. The number of such prestimulations is set by *Presends*. Normally a value of one is sufficient.

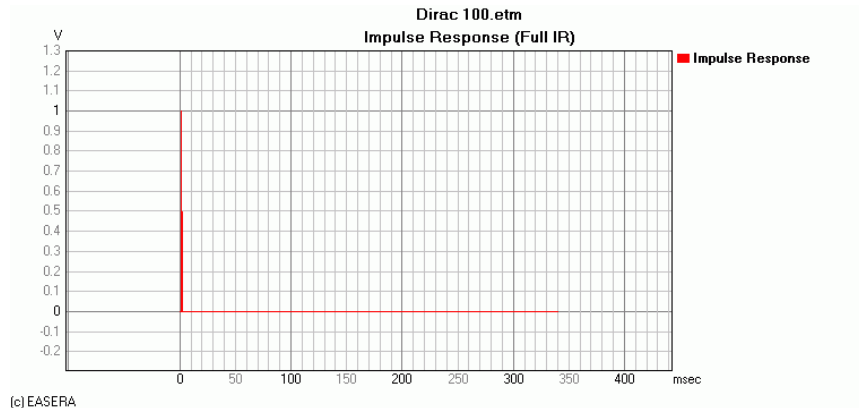
Finally the measurement is started using the *Go* button.

# Lesson 4: What is an impulse response?

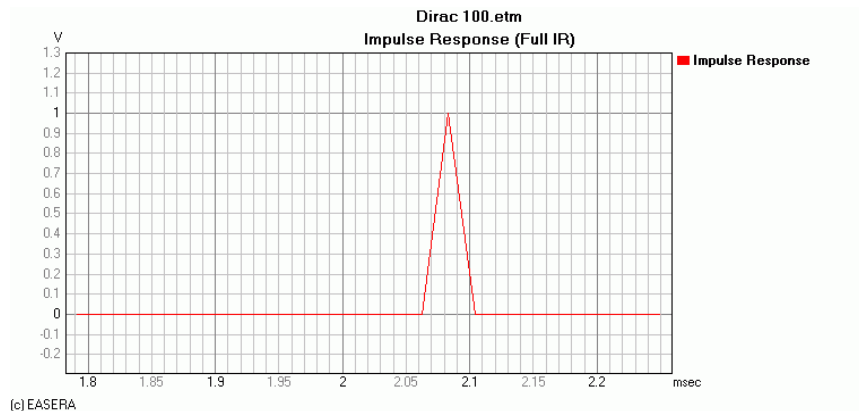
All calculations carried out by EASERA are based on the impulse response which is the response of a system to a Dirac impulse. In this lesson we are going to have a closer look at this procedure.

## What is a Dirac?

In the *Tutorial Files* directory under *EASERA10DATA* we find the file *Dirac 100.etm* which we are going to open in the *View & Calc* page:



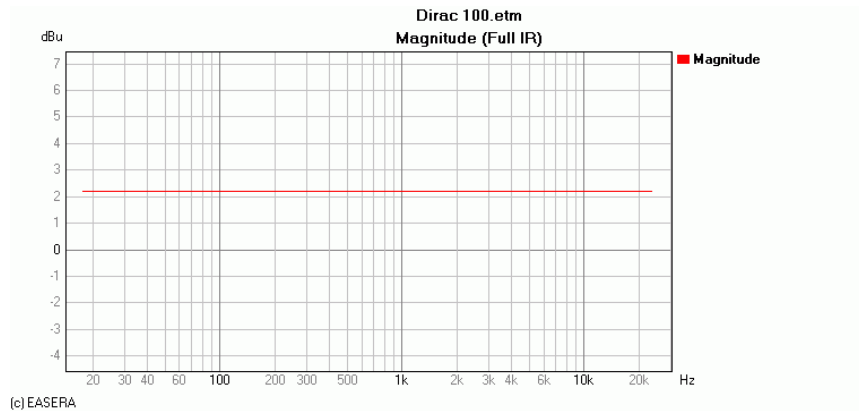
The impulse response we obtain is that of a very narrow impulse.



If we zoom in on the time axis, we note that the impulse has a width of exactly one sample. Thus it is the shortest impulse the system is able to describe. Such an impulse is called a *Dirac pulse* by the mathematicians.

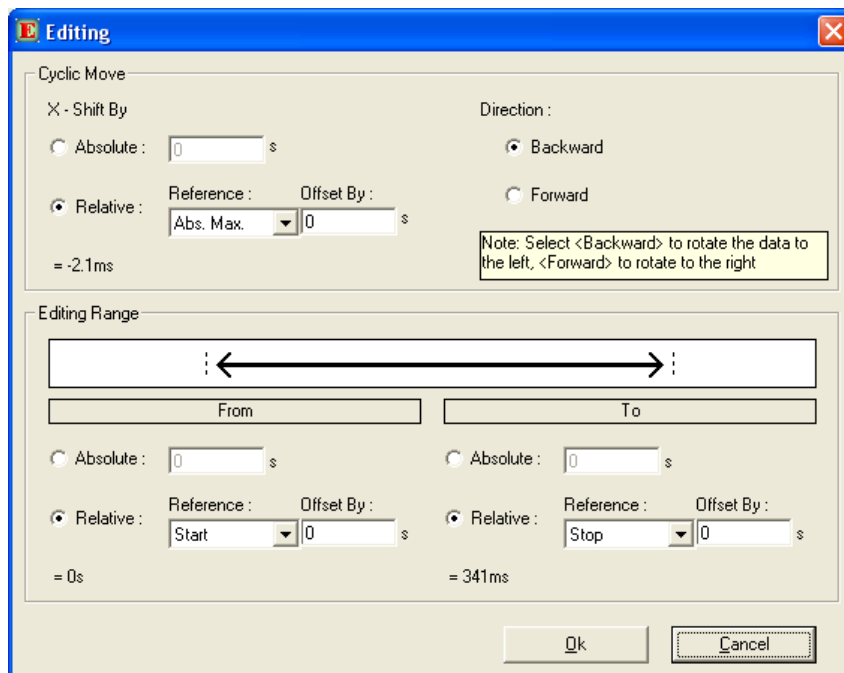
Mathematically this pulse is defined as having infinite energy and zero width, which is not possible in the real world, even digitally. In EASERA this is a signal that consists of a single sample set to one with all other samples set to zero. A Fourier transform of this Dirac pulse results in a response that contains all frequencies from zero to the Nyquist limit (1/2 the sample rate) with equal energy and a known phase.

If we look at the frequency response of this Dirac impulse, we note that it is ideally linear.



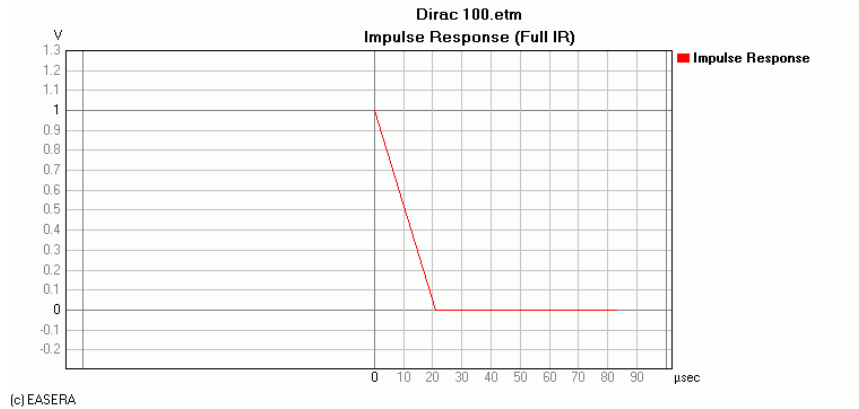
If we had a look at the phase response, this would not be ideally linear. This is due however, to the fact that the impulse peak is at 2.08 ms and not at the beginning, which implies that a corresponding runtime phase occurs additionally.

We now want to shift this impulse to the beginning. To do this we go to the impulse response and choose *Edit/Cyclic Move/Move Abs Max to Zero*. We are then confronted with the following dialog:



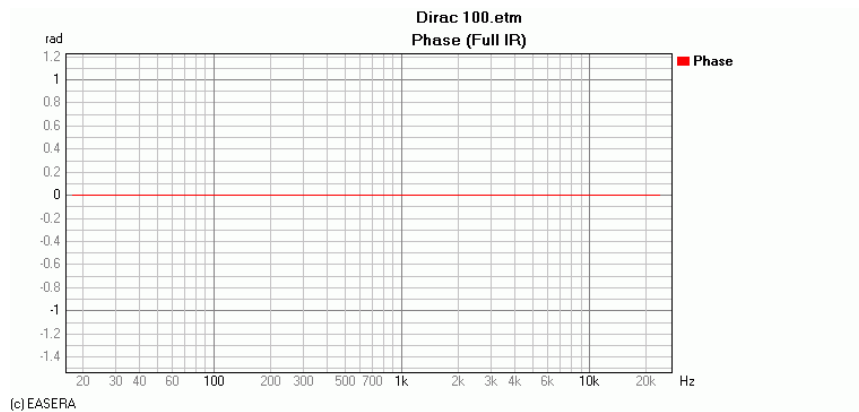
Here one could adjust not only the time span by which the data shall be cyclically shifted, but also in which range of data this shall be done. Since everything has already been correctly adjusted here, we accept the dialog with *Ok*.

Cyclic shifting means shifting the data on the X-axis, whereby all data shifted beyond the beginning or the end of the data range are added to the respective other side.



If we now consider the impulse response, we see that the impulse is at the very beginning.

When looking at the phase response, we see that it is also ideally linear, meaning it is a constant zero.



We are now in a position to tell what kind of signal a Dirac impulse is: It is a signal with ideally linear frequency and phase behavior.

## What is the Dirac impulse good for?

By using a Dirac impulse it is now possible to determine the impulse response of a system. The impulse response is the output signal of a system stimulated by a Dirac impulse.

If an ideally linear system is involved (by approximation: a length of wire), the output signal will also be a Dirac impulse. With all other systems the output signal will show more or less pronounced deviations from the input impulse.

Let us take a short digression into the systems theory: If any system is stimulated by a signal  $u(r)$ , the system response  $y(t)$  in the time domain is the convolution integral of the stimulating signal and the impulse response  $h(t)$ :

$$y(t) = \int_0^{+\infty} u(r) \cdot h(t - r) dr$$

(In the frequency domain, the convolution integral is to be replaced by a product.)

Consequently, a (linear, time-invariant) system is completely described by its impulse response, all other properties can be deduced from it.

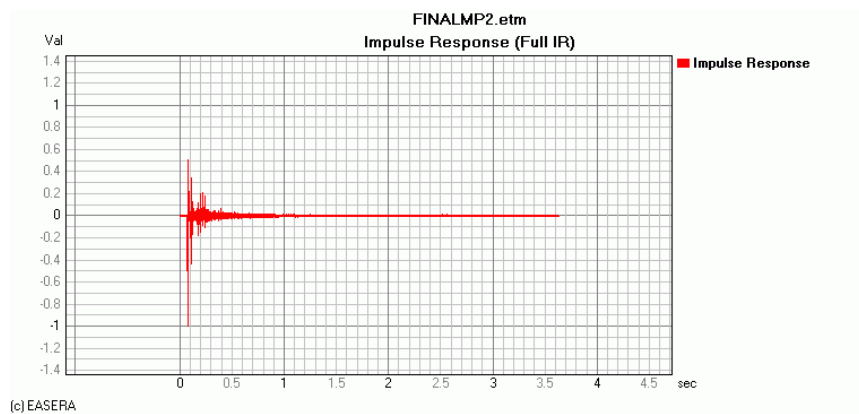
One must say that the Dirac impulse is rather inappropriate for making measurements in practice. Since one cannot use impulses of any desired magnitude without colliding with the amplitude limit, one has to measure with amplitudes far too small but maintaining a sensible signal-to-noise ratio at the same time. Nevertheless it is done to obtain first impressions, e.g. by clapping the hands or bursting a toy balloon (in the early days with an impulse generator as an excitation source for so-called impulse sound tests), where the Dirac impulse is approximately reproduced.

For this reason it is better to measure using another signal (sweep, MLS, noise...) and convert the result back into an impulse response.

## The impulse response in room acoustics

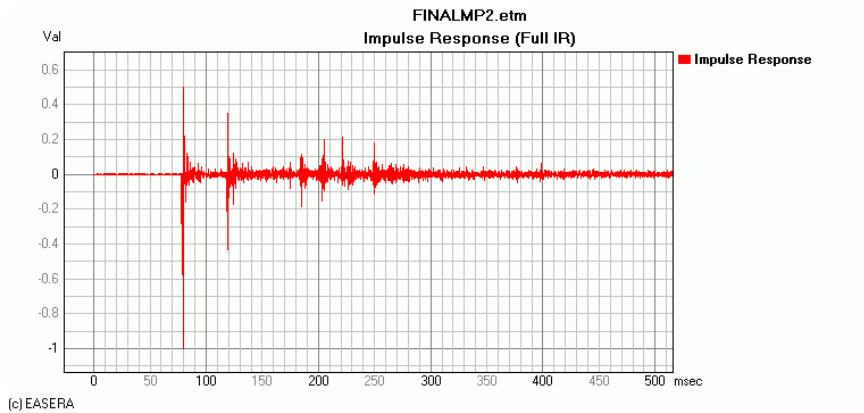
As the behavior of a system is fully described by the impulse response, it is possible to deduct all the other quantities from it (which is in fact done by EASERA). With some experience, one can read at least a part of the system behavior directly from the impulse response.

In room acoustics the impulse response may be used for assessing the reflection behavior of a room.



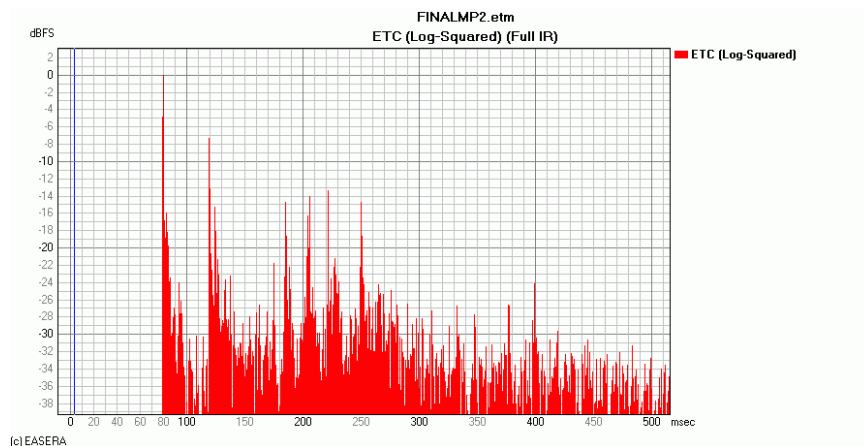
The above diagram shows the impulse response of a room. First one notices that the impulse points downwards. This results from a phase inversion somewhere in the overall system, e.g. due to a phase-mismatched connection of the loudspeaker or an inversion of polarity in the microphone line. It has no bearing on an ensuing evaluation.





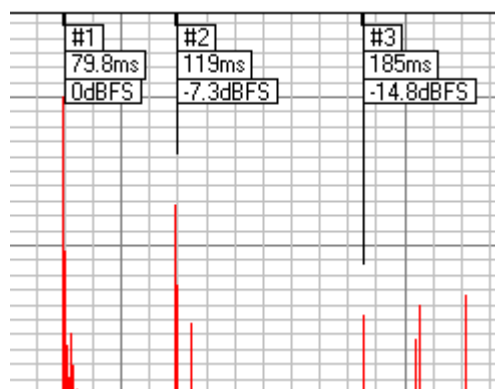
When zooming into this impulse response, we notice a distinct reflection at approx. 40 ms after the direct sound. The smaller impulses following after the direct sound are peaks produced at the loudspeaker and determined by the decay behavior of the same.

For analyzing the individual reflections, the ETC diagram is also very suitable. Select this function (it does not matter if *ETC (Log-Squared)* or *ETC (Envelope)*) and zoom accordingly into the diagram.



One of the advantages of the ETC diagram is the logarithmic dB scaling of the level axis.

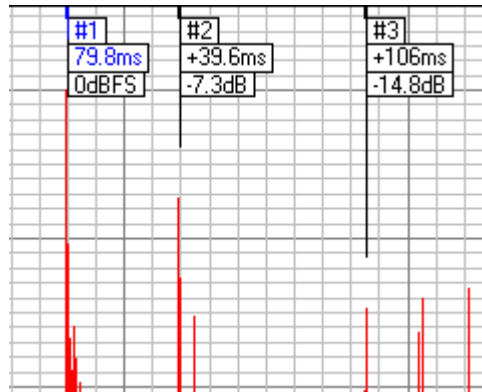
If you choose the function *Mouse/Peek*, you may place cursors into the diagram by clicking with the right mouse key. The cursors then show the time and level values given at the labeled points:



There also exists the possibility of using one of the cursors as the base point (a reference cursor) for a relative display of the other cursors. Usually one chooses the first impulse as a reference for relating all other reflections. In order to establish a reference cursor, click on the navigator page *Cursor* into the column R of the corresponding entry which is then marked by a blue square.

C	A	R	V	Time	Label
		X	X	79.8ms	#1
		X	X	119ms	#2
		X	X	185ms	#3

After selecting the reference cursor, the time and level values of the other cursors refer to this reference point:

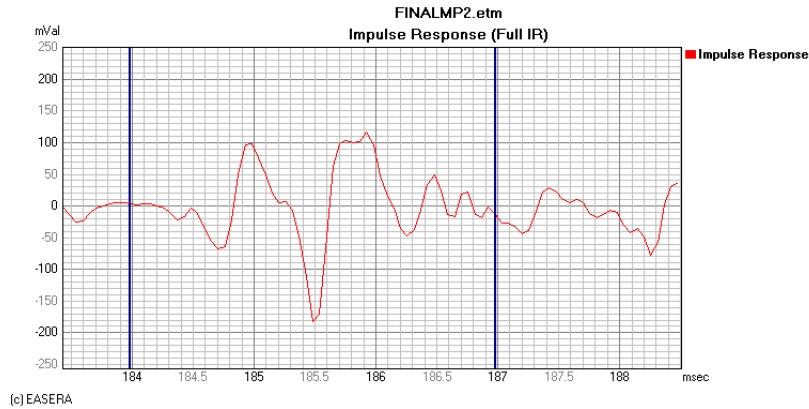


At approx. 106 ms after the direct sound there follows at intervals of approx. 20 ms four smaller, but still clearly pronounced reflections. We would now like to assess whether speech intelligibility could be improved if these reflections were eliminated, e. g. by application of an appropriate damping material. (It goes without saying that by such a measure one would influence the overall reverberation behavior. The exactness of this assessment is thus rather limited.)

For establishing a corresponding comparison we are going to first calculate the speech intelligibility:

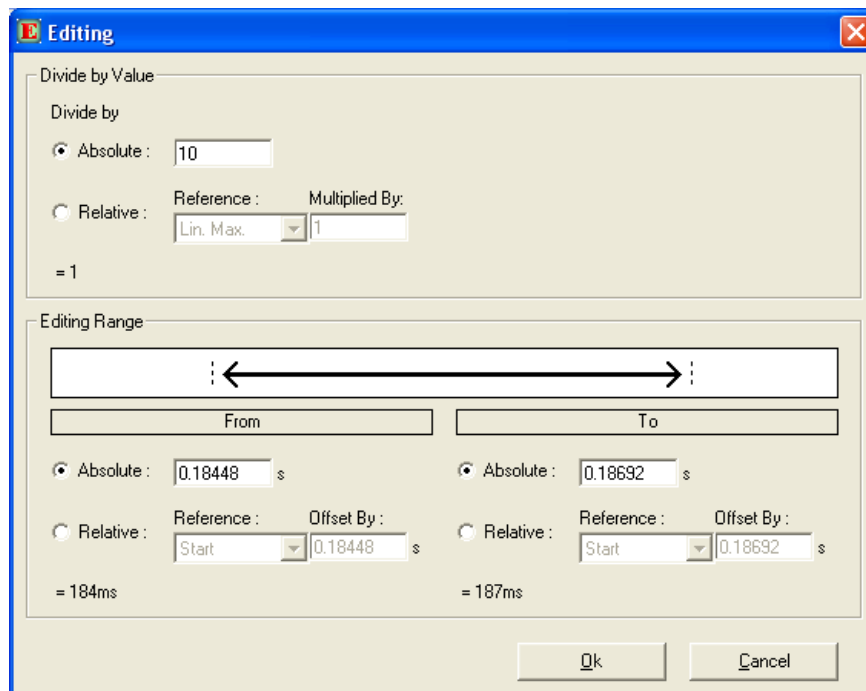
STI	0.529
AICons [%]	9.687
STI (Male)	0.539
STI (Female)	0.554
RaSTI	0.505
Equiv. STIPa (Male)	0.554
Equiv. STIPa (Female)	0.566

We go back to the impulse response and zoom in to a size that we get the first of these four reflections separated:

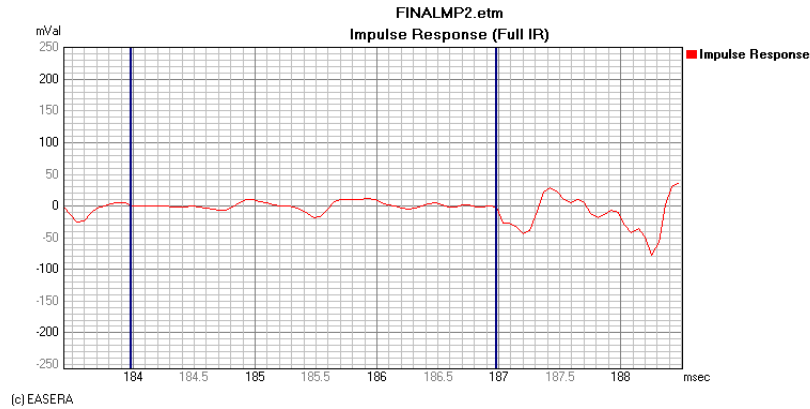


In front of and behind this impulse we place markers (by means of the left and right mouse keys, respectively, while holding the Ctrl key), in such a way that they lie in the zero crossing.

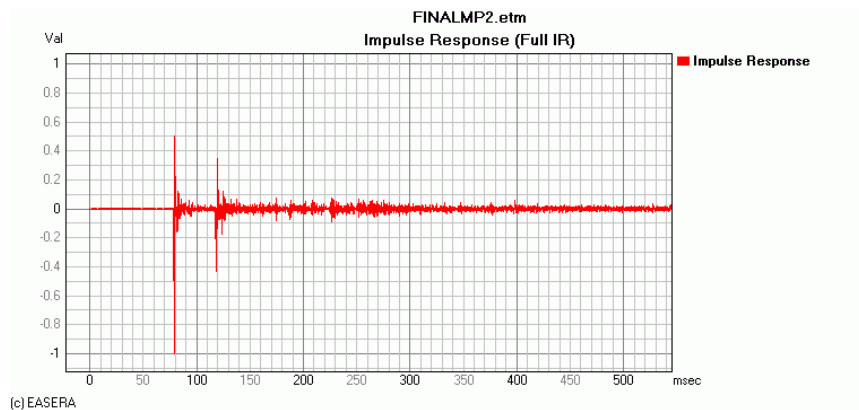
One could now set all values contained between the markers (*Edit/Set To/Zero*) to zero, but this would not correspond to the behavior of the remaining reverberation. Therefore, these values shall only be reduced by 20 dB, i.e. divided by 10. To do this we choose *Edit/Multiply* and *Divide/Divide By Value* and get to the following dialog into which we enter 10 as divisor:



Between the two markers there is now an accordingly reduced level:



Using the same pattern, we also eliminate the remaining three reflections:



Of course, one could obtain an equivalent result in just one step by choosing the editing range accordingly. The effect would be somewhat improved speech intelligibility:

STI	0.531
AICons [%]	9.590
STI (Male)	0.541
STI (Female)	0.556
RaSTI	0.507
Equiv. STIPa (Male)	0.554
Equiv. STIPa (Female)	0.568

We recognize a significant increase of the STI values. (For this example the values shown may not be reproduced accurately, since they depend on the exact time range used for the IR correction. But the tendency is clear.)

However, such a manipulation of the impulse response also always changes the frequency response.

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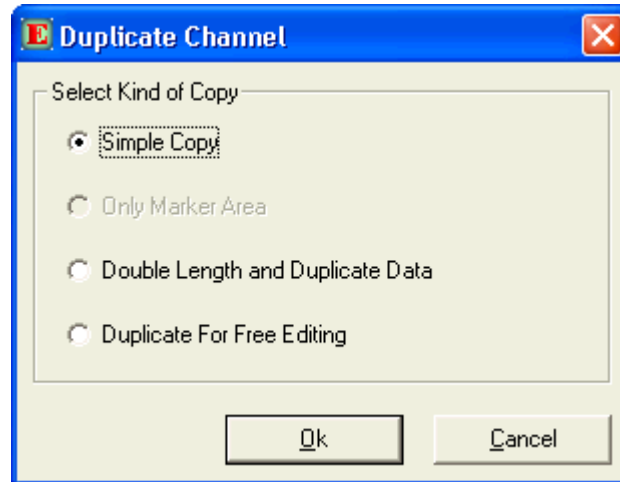
## Impulse response of loudspeakers

The impulse response can be an interesting criterion for the assessment of loudspeakers. Its proper interpretation requires, however, some experience – we are now going to acquire some fundamental concepts.

To do this we need the Dirac impulse once more, but not shifted to the starting point.

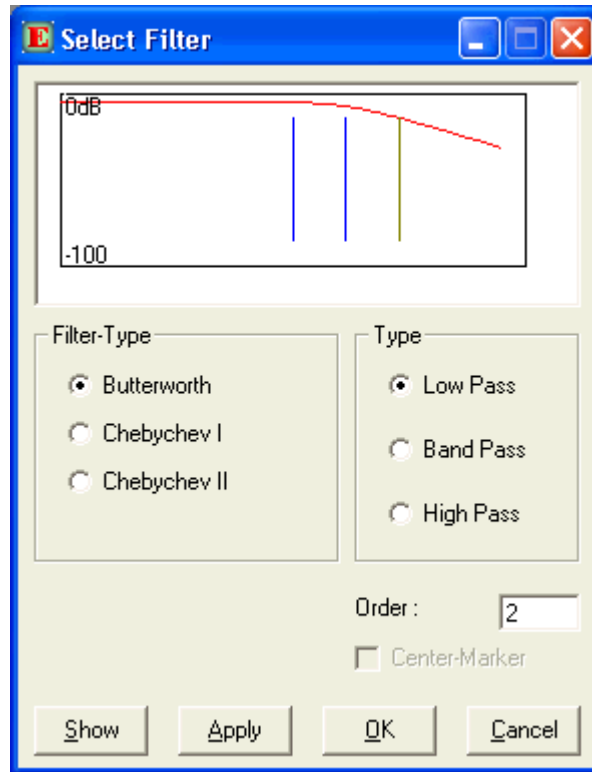
Using a high-pass and a low-pass filter, we split the signal into two sections. With a sampling rate of 48 kHz, the signal is limited to 24 kHz and the mid-point would be 12 kHz.

To enable filters to be set, we must copy the Dirac impulse first (*Edit/Duplicate File* and then *Simple Copy*).



To set the right marker to 12 kHz, we zoom into the range of this frequency and mark using *Ctrl+RMB*, or we enter the value numerically in the marker line at the lower edge of the EASERA window.

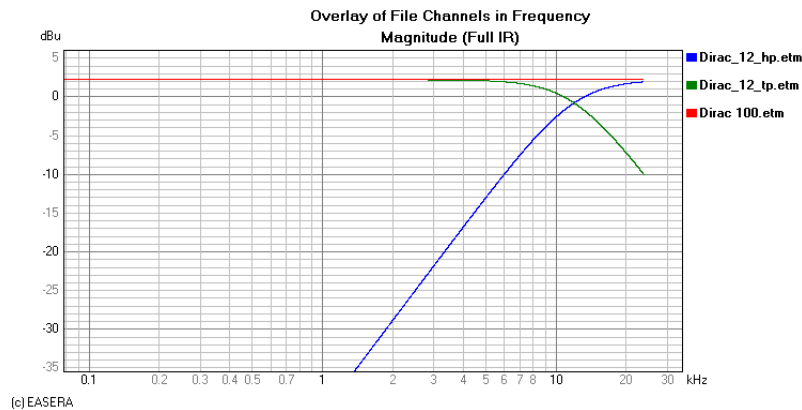
Then we choose *Edit/Filter*:



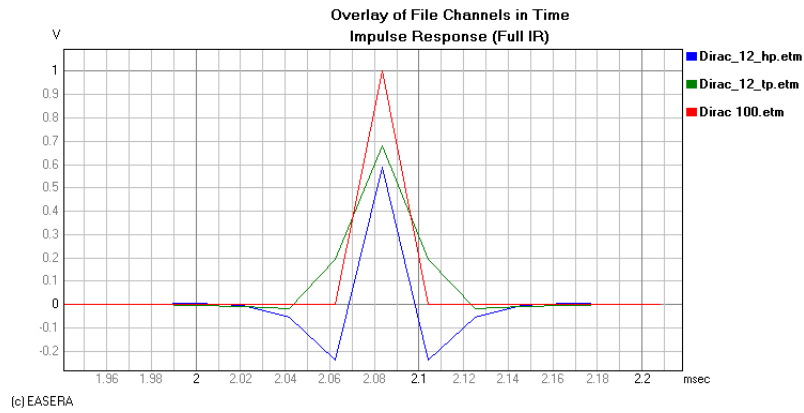
We use a Butterworth low-pass filter of 2<sup>nd</sup> order and confirm with *OK*.

Now we produce a second copy of the Dirac impulse and filter it as well at 12 kHz, but this time using a high-pass filter.

Let us consider the two curves (Low-pass: dirac\_12\_tp, high-pass: dirac\_12\_hp) first in their frequency-response representation:

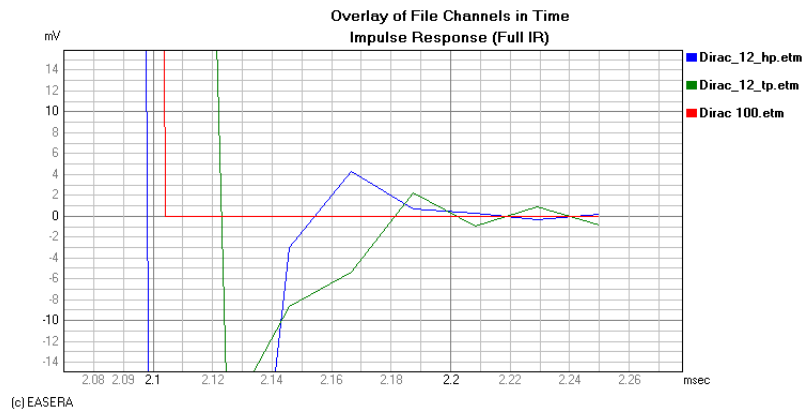


This looks the way it should, so we go on to the impulse response:



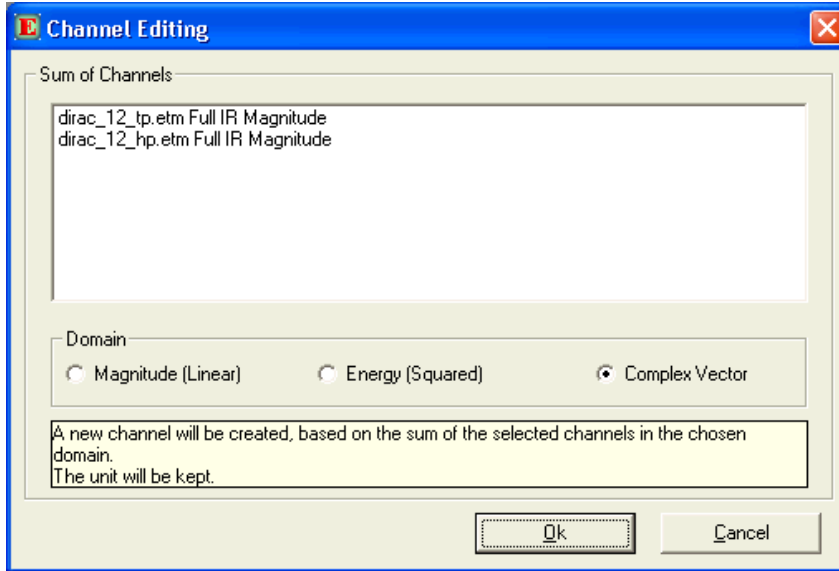
For comparison we still have the original Dirac impulse. The signal with the low-pass filter broadens the signal compared to the Dirac impulse, and the signal with the high-pass filter produces downward overshooting.

Detail amplification shows more small transient or decay phenomena:

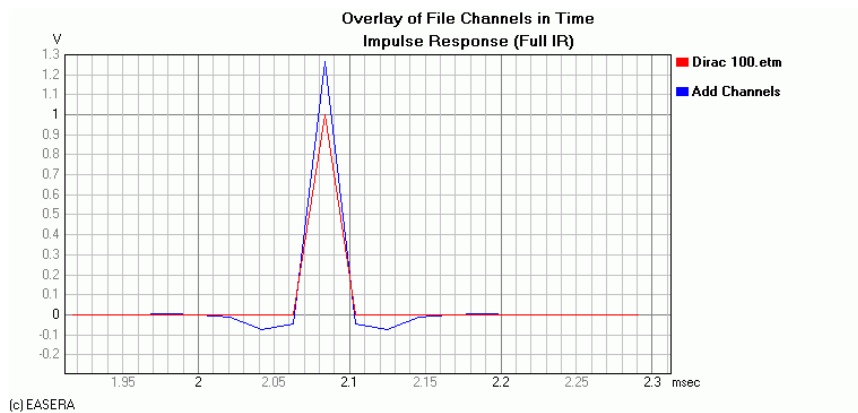


## Adding up signals

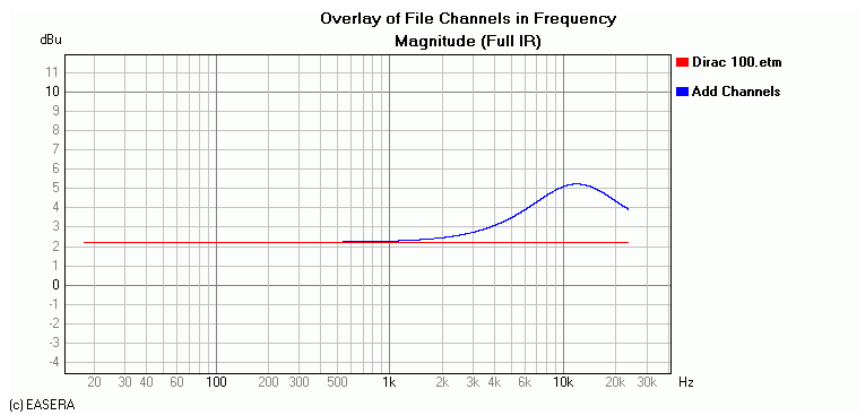
Now we want to add the two filtered impulse responses back together. We establish an overlay (in the frequency domain!) in which these two files are exclusively contained, and then choose *Edit/More/Add Channels*:



We may now decide how we want to add up the signals and opt for *Complex Vector*, i.e. for an addition using consideration of the phase data.



Looking at the impulse response we notice that the sum of the two parts produces a higher amplitude than the original, accompanied by a downward overshooting. This result is plausible when looking at the frequency response:

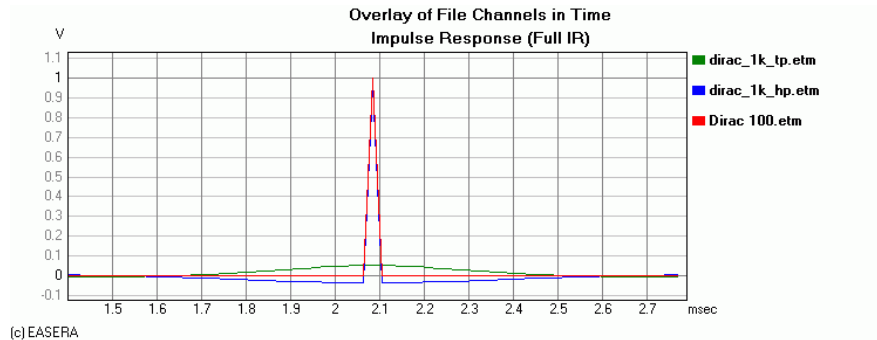




With exception of the Linkwitz-Riley Filter, all conventional filter types produce a 3 dB peak at the separating frequency when being added. This naturally results in increased amplitude and a slight high-pass filter behavior.

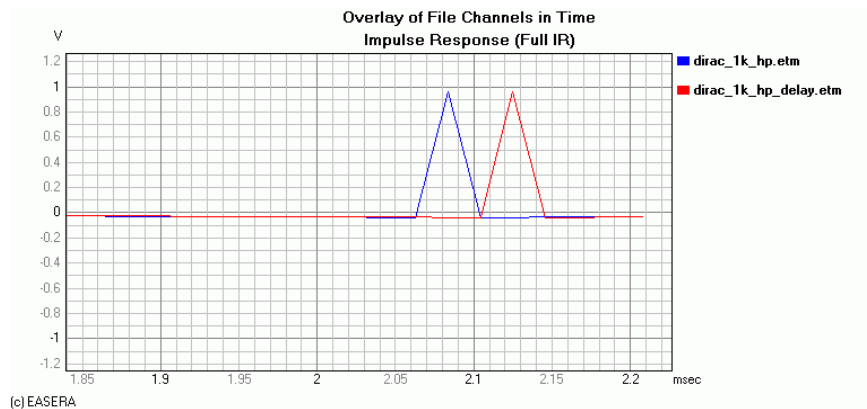
## Crossover frequency at 1 kHz

We are now going to simulate the behavior of a 2-way box for which it would be rather unusual to have the crossover frequency at 12 kHz. For this reason we create two more Dirac copies and crossover these at 1 kHz.

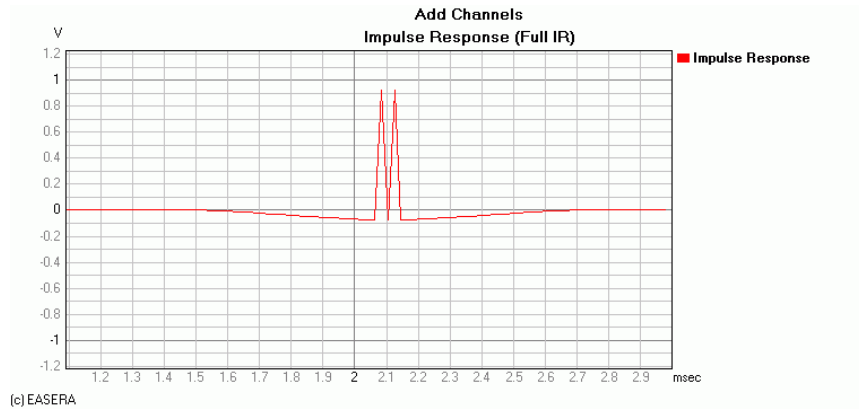


As compared with the crossover at 12 kHz, the low-frequency component now has a much smaller amplitude, since it contains only approx. 4% of the power. If you see pointed jags in an impulse response (from sample to sample), you may be sure that these are high-frequency components.

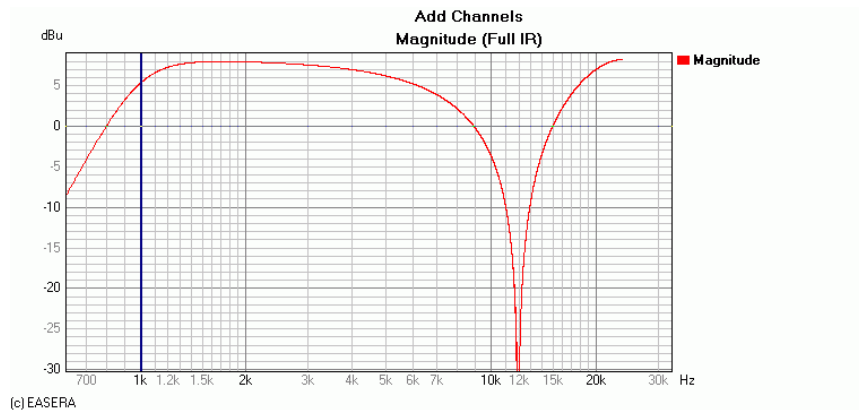
We now want to duplicate the high-frequency component once more and shift the result earlier by 50  $\mu$ s (i.e. 0.00005 s). Seen in an overlay, this looks as follows:



When adding up the two channels, we obtain – not altogether unexpectedly – the following result for the impulse response:



We now examine the appertaining impulse response:

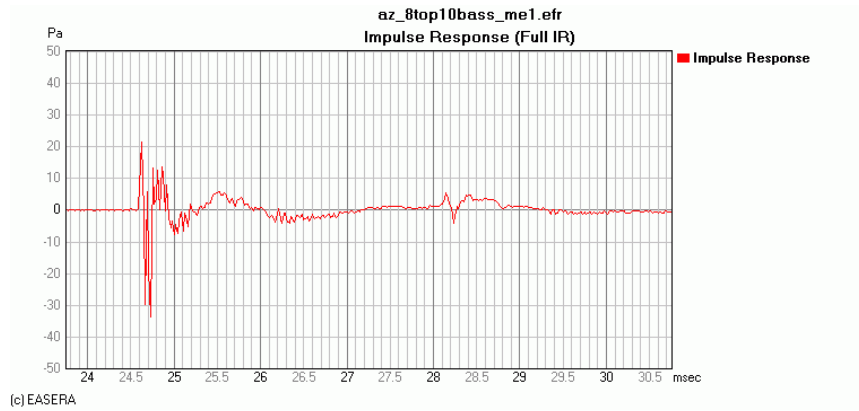


The drop below 1 kHz was to be expected, since we applied a high-pass filter to both signals. It is also not unusual for delayed signals to produce interference. Such interferences always occur when two signals of equal frequency and equal (or at least similar) level are added. With a time shift of  $\Delta t$ , an interference of this kind occurs at a frequency of  $1/(2\Delta t)$ , subsequent ones at  $n/(\Delta t)$  with  $n=1,2,3,\dots$ . With a time shift of  $50 \mu\text{s}$ , the first interference would therefore be at a period duration of  $100 \mu\text{s}$  and thus at a frequency of 10 kHz. The first interference minimum occurs here quite definitively at 12 kHz.

The solution of the problem lies in the sample rate of 48 kHz which results in a time resolution of  $20.8 \mu\text{s}$ . The  $50 \mu\text{s}$  we entered as delay time are rounded to the nearest value and  $(2 \times 20.8 =) 41.6 \mu\text{s}$  is nearer than  $(3 \times 20.8 =) 62.4 \mu\text{s}$  in this case. We obtain the same result by measuring the distance between the two impulse peaks. By converting  $41.8 \mu\text{s}$  to half a wavelength we obtain a frequency of 12 kHz.

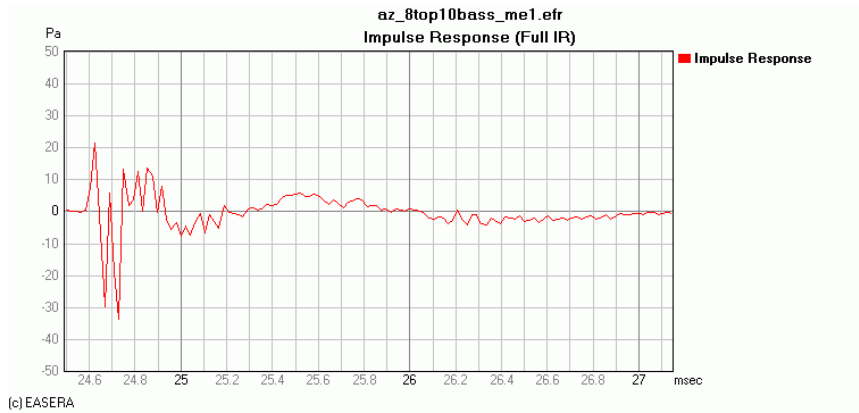
## The impulse response in practice

In practice we do not have such nice impulse responses, but rather something like this:

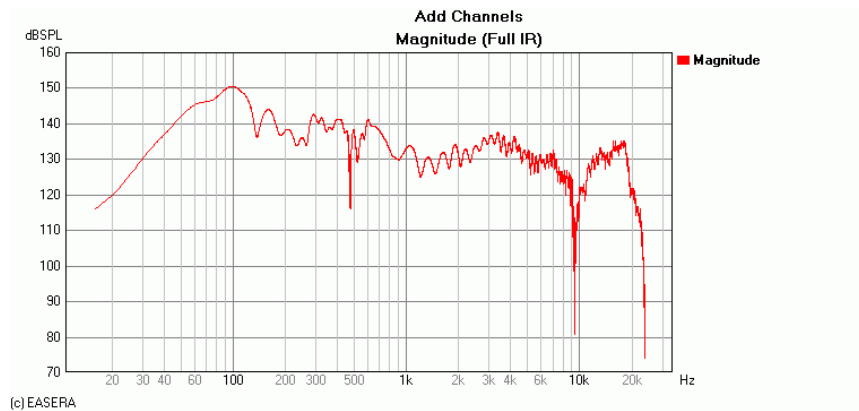


The impulse we see at 28.15 ms is at an interval of 3.52 ms from the direct sound, which would correspond to a distance of approx. 1.20 m. It is the first reflection from the acoustically hard floor.

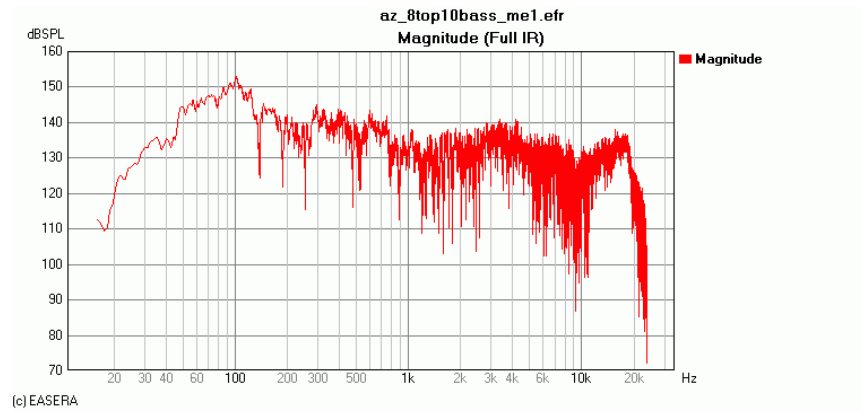
Since we are interested only in the loudspeaker response (in this case two single systems with one 2x8" and one 1" horn as well as two 4x10" bass horns), we zoom in more:



What we see is a negative double peak, presumably caused by the two loudspeakers not being located at an exactly equal distance from the measuring microphone. Having learned this, we are now able to predict in which way this will influence the frequency response. There will be an interference dip somewhat below 10 kHz. A glance at the frequency-response does not disappoint us:



Incidentally: The frequency response is so smooth only because the reflections have been removed by means of two windows. Without windowing and smoothing it would look as follows:



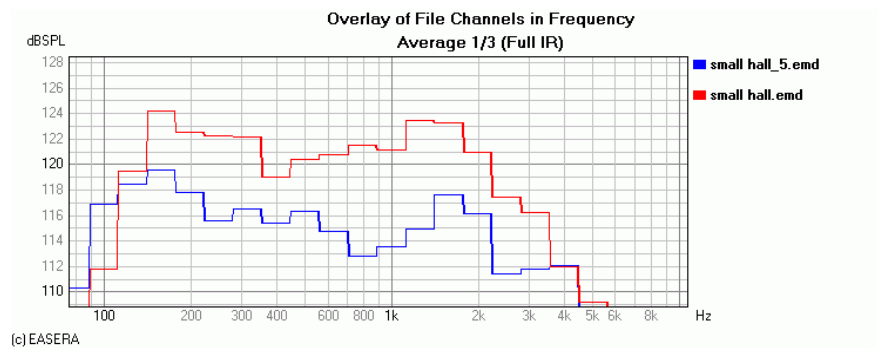
# Lesson 5: Averaging and adding measurements

We now want to see how EASERA combines multiple data records by averaging or addition. For this purpose we will use two measurements performed in a medium-sized hall in Berlin.

The two data sets were both taken at the same measurement position. One of the data sets reflects measurements with an Intellivox Line-loudspeaker (small hall.emd), the other with a loudspeaker serving as a point source that imitates the lecturer (small hall\_5.emd).

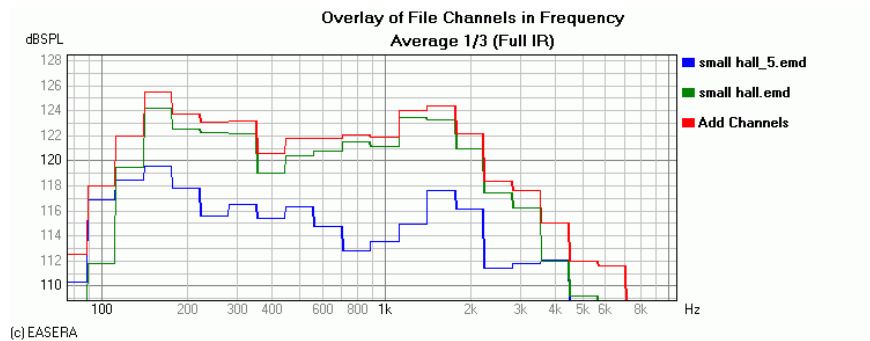
## Adding data

First the two data records will be added. To do this we first display the two files as an overlay in a one-third octave averaged view.



(c) EASERA

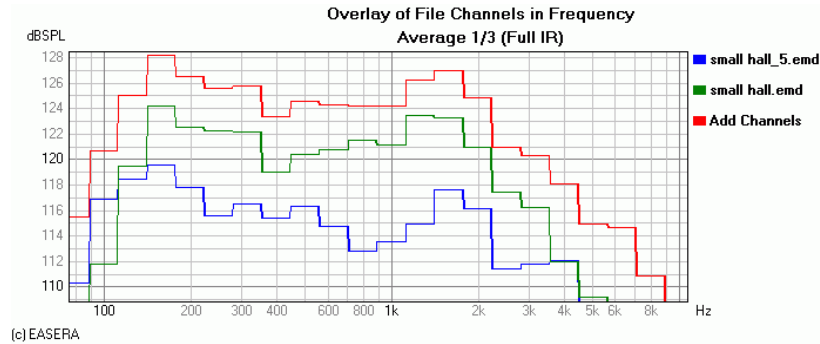
Then we choose *Edit | More | Add Channels*. EASERA offers three options to add the two curves, and we first choose *Energy (Squared)*. Then we zoom in to the resulting curve and add the two summands to the overlay:



(c) EASERA

The two original curves were added as uncorrelated signals would be summed energetically. Where the levels are equal (here at 4 kHz) the overall level is 3dB higher compared to the original values.

Now we add the two curves once more, but this time assuming correlated signals we choose the *Magnitude* option instead of *Energy*:



Now correlated pressures are summed instead of power quantities. Where the levels are equal (here at 4 kHz) the overall level is 6 dB higher compared to the original values.

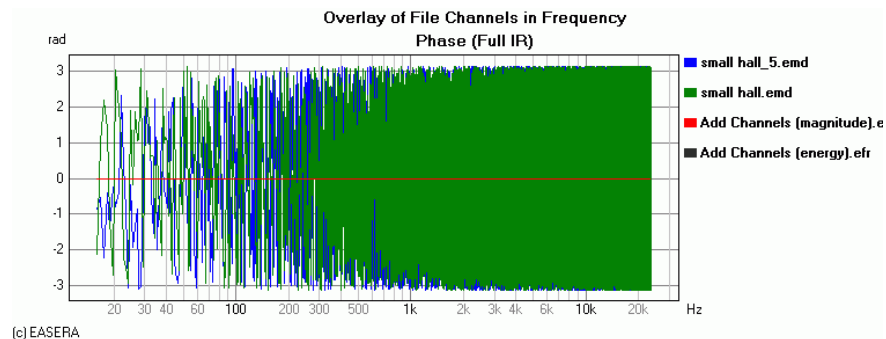
## Frequency resolution and phase behavior

Before having a look at the third option (*Complex Vector*) we first want to investigate the frequency resolution and the phase behavior. If you view the unsmoothed frequency response (*Magnitude*) in one of the two summation curves, you will notice that it follows exactly the one-third octave bands – there is no obvious difference between *Magnitude* and *Average 1/3*.

The reason is that no addition of real data (unsmoothed data) is performed here, but actually an addition of the smoothed data that is currently displayed: If values in one-third octave bands are summed, the unsmoothed addition result will also follow that resolution – a higher resolution can no longer be selected later on.

Usually one prefers to use the unsmoothed data – a smoothed representation can be obtained at any time later. An addition (and later on, also an average) should therefore be performed generally with unsmoothed data (*Magnitude* item on the *Graphs Navigator* page).

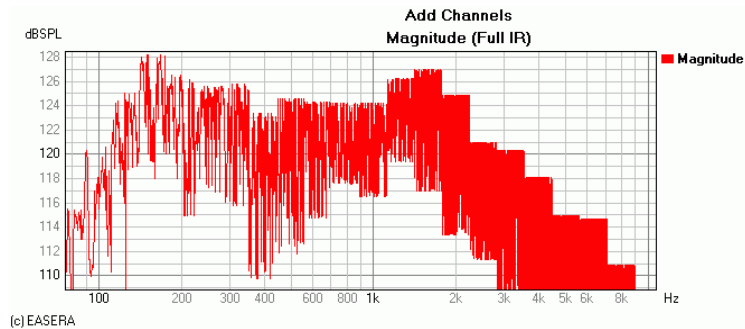
Next we view the phase behavior of the summation curves:



With *Magnitude* as well as with *Energy* each value in the phase result is zero – i.e. the summation implies loss of the phase information. Strictly speaking, the phase information not only gets lost, but is completely ignored in the summation. This means that between the channels involved there cannot be any destructive interference phenomena.

## Complex Vector

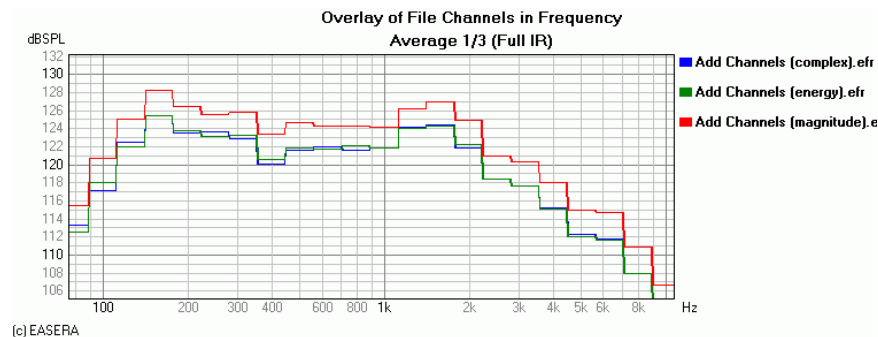
The two curves are now to be added using the *Complex Vector* option where not only is the amplitude information considered, but also the phase information:



The diagram obtained shows first of all large interferences plus the resulting notches in the frequency response. With a vector addition the phase information is considered as well as the magnitude information.

In any case it is questionable if the vector addition of octave and one-third octave spectra is reasonable: The frequency response gets averaged for the respective octave or one-third octave bands, but not the phase response. If the result of such an addition is then shown as a frequency response, there can be no sensible conclusions drawn from it.

As the following diagram shows, the presentation of this curve in the shape of a one-third octave spectrum would result in a typical diagram which, however, is to a large extent congruent with the *Energy* addition:

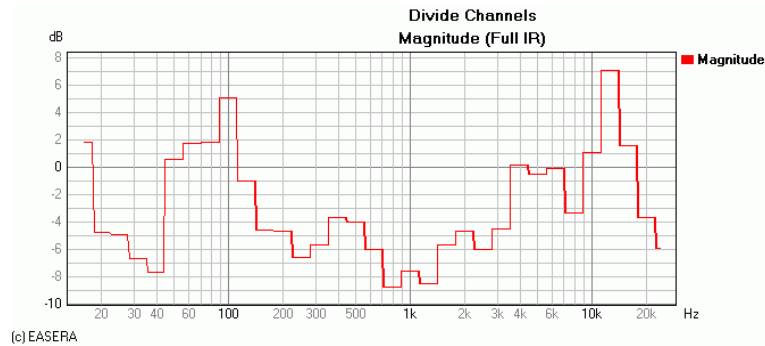


The same result could be expected if the frequency responses would be added using complex values and then shown as one-third octave spectra.

If voltages (mixing console) or sound pressures (two sound sources in one room) are added, this always happens as a vector sum in reality, i.e. considering the phase. This can be simulated in EASERA by adding impulse responses or by adding unsmoothed frequency responses with the option *Complex Vector*. Both procedures lead to the same result.

Besides the addition of curves there are other mathematical operations possible. For the subtraction of data the options *Magnitude*, *Energy* and *Complex Vector* are available, whereas with multiplication and division the calculation is always done with complex vectors. The multiplication corresponds to a convolution, which allows filtering a curve for instance. The division corresponds then to a deconvolution or normalization.





(Levels are generally defined as logarithmic entities so that calculating a difference of levels requires in fact a division.)

## When do we use which option

The newcomer does not always have a clear idea of which option is to be used for the addition (subtraction, averaging...) of data. Here are some guidelines:

- If uncorrelated signals are involved (useful sound and noise spectrum) or – and this will probably be the most frequent application – the averaging of frequency responses over several room positions, the option *Energy* is used.
- If correlated signals are involved, the option *Complex Vector* is used. For instance, with a stereophonic sound reinforcement system first the left and then the right signal path is measured at the same room position. Then it is possible using the option *Complex Vector* to ascertain what the frequency response of the stereophonic system is.
- If two sound levels (signals like sound pressure or voltages) will be added, where the phase behavior is to be neglected in practical application, one may use the option *Magnitude*.

If octave or one-third octave spectra are to be added (subtracted, averaged...), one should use *Energy*, in few cases also *Magnitude*, but not *Complex Vector*.

## Data averaging

To average data a sum is calculated first which is then divided by the number of channels involved. For the options *Magnitude*, *Energy* and *Complex Vector*, the same rules apply as with addition.

## Speech intelligibility

Using speech intelligibility as an example, we now want to see what influence the averaging has on acoustical quantities. For this purpose we use two measurements *small hall3.emd* and *small hall4.emd* which were carried out with the same loudspeakers (Intellivox), but at different measurement positions.

In both cases the measuring positions produce the same speech intelligibility value STI of 0.587. The *Speech Transmission Index* is a value between 0 and 1, where the higher the value the higher the speech intelligibility. If one formed the arithmetic average (or also the geometric average) of the speech intelligibility index values, the result would once again be 0.587.

Now we calculate the three possible averages (*Magnitude*, *Energy* and *Complex Vector*), show the speech intelligibility and obtain the following values:

- Magnitude: 0.754
- Energy: 0.757
- Complex Vector: 0.573

The particularly good values obtained for *Magnitude* and *Energy* are by no means astonishing: The averaging causes the frequency responses to be smoothed (interferences due to differing phases do not occur here), and the smooth phase response throughout improves the result further. Things look different if the option *Complex Vector* is used for averaging. Due to interferences between the measurements involved the frequency response usually becomes more uneven, which decreases the STI somewhat.

As a result we state that one should never try to derive any time-based room-acoustical quantities like e.g. speech intelligibility or reverberation time from these averaged frequency response data.

An average of speech intelligibility values based on averaging impulse responses or frequency responses should therefore never be carried out, neither with EASERA nor with any other measuring tools. One only could and should indicate a representative mean value, derived as the average of several STI numbers, and the regions in which the STI fluctuates.

Speech intelligibility values must always be derived from individual measurements. None of the three options produces even approximately correct results: With *Magnitude* and *Energy* the phase information gets lost in the averaging process, which paradoxically improves the speech intelligibility. When averaging with *Complex Vector*, interferences between the different positions occur which, on the other hand, impairs the values contrary to realistic expectations.

## Other acoustical quantities

What has just been demonstrated by the example of speech intelligibility also holds true analogously for other acoustical quantities: The averaging of impulse responses does not result in a "representative impulse response", the acoustical quantities of which correspond to the average of the acoustical quantities obtained with the individual measurements.

If the impulse responses are at least similar, the result may be an approximation of an average. With non-similar impulse responses, however, there will be no sensible results obtained.

## Frequency response

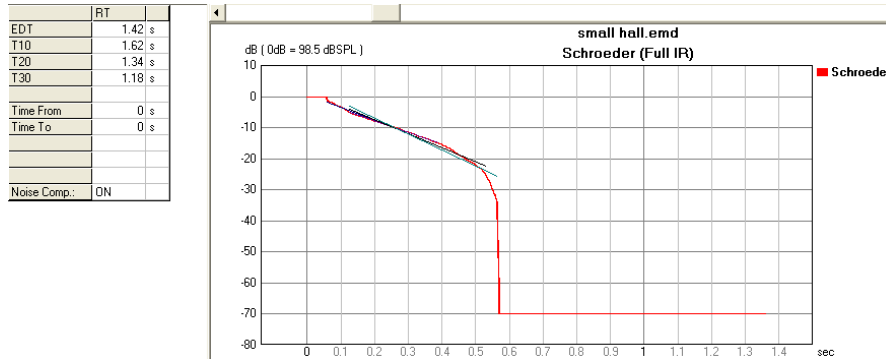
The primary field of application for averaging is the formation of frequency responses over several microphone positions. If a sound reinforcement system is e.g. to be corrected using an equalizer, it has to be optimized not just for one microphone position, but for the whole audience area.

On this audience area one could now distribute several measuring microphones. If the signals of these were summed up using a mixing console (or analogously in EASERA by the option *Complex Vector*), large interferences would result between these signals. Thus it is more appropriate to record the channels one by one (EASERA permits up to 32 input channels) and then to average the frequency responses energetically.

# Lesson 6: Filtering and windowing

Filtering and windowing are essential elements in the preparation of data for display. While filtering takes place in the frequency domain, windowing is done in the time domain. We are now going to have a look at the many possibilities which EASERA offers in this respect.

We will use the supplied file *small hall.emd* and show the *Calculation | Schroeder RT* graph. Now the Schroeder plot is displayed in the diagram area, i.e. the backward integrated energy to which the regression straight lines for the individual reverberation time quantities are attached. The values calculated for the reverberation time are shown on the *Details* navigator page.



The Schroeder integral is derived from the complete impulse response (Full IR), so that we now see the broadband reverberation time.

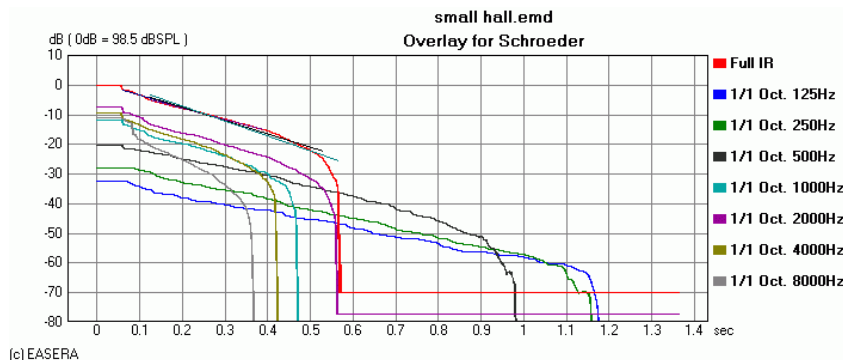
The reverberation time is strongly frequency-dependent usually: Owing to the air damping and the wall-material absorption increasing at higher frequencies, the reverberation time results with low frequencies are often significantly higher than with high frequencies. The broadband reverberation time is therefore practically of limited usefulness.

## Octave and one-third octave spectra

To determine the reverberation times separately for the individual octave bands, we click on the *Octaves* button in the filter section.

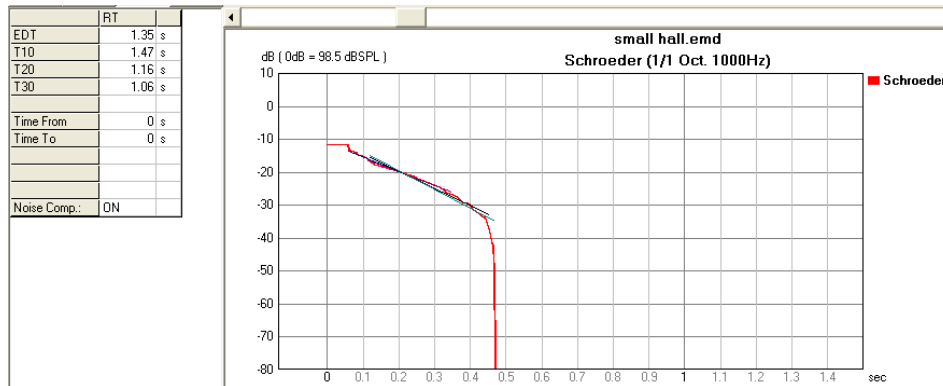


EASERA then decomposes the impulse response (and thus the diagrams also derived from it) into octave bands.

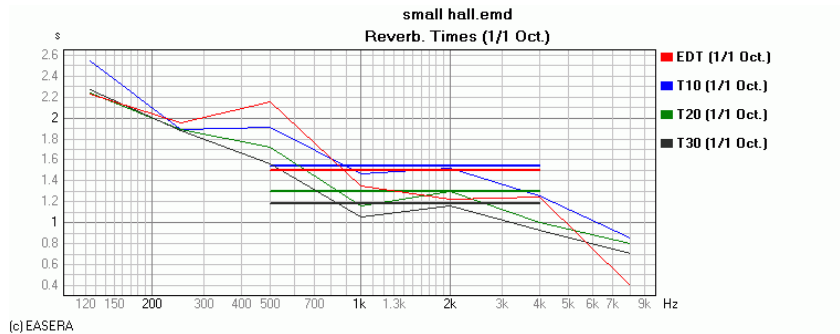


As can be seen here pretty well, the curves are flatter with the low frequencies than with the high frequencies, and the reverberation time is accordingly longer.

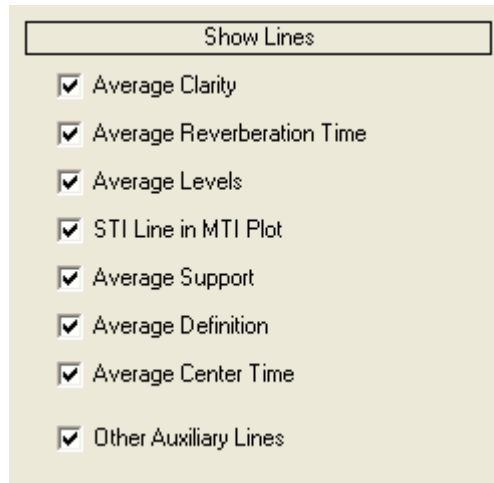
To obtain the numerical values in a specific octave band, the corresponding frequency band is chosen in the selection list directly to the right of the *Octaves* button. In the diagram area only the Schroeder integral for the chosen octave band is displayed now, while the corresponding values are shown on the *Details* navigator page. Please note that the diagram is no longer labeled *Full IR*, but shows the frequency of the chosen octave band.



It would certainly not be very efficient to "collect" the octave-averaged frequency response of the reverberation times one by one via a corresponding number of filtering runs – EASERA offers a combined diagram under *Calculation / EDT, RT (Octave)*:



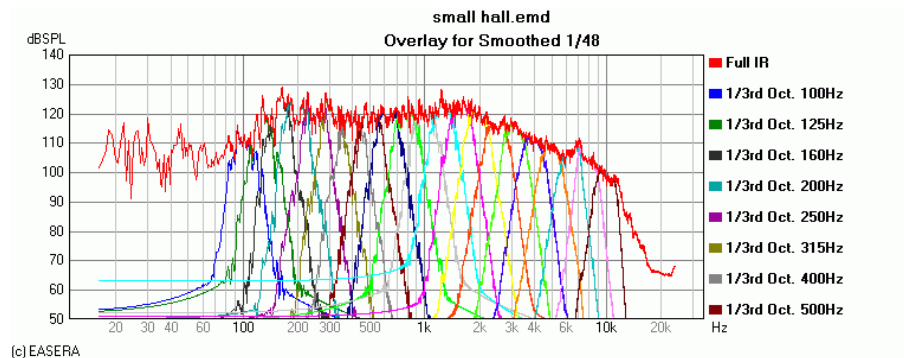
(For a display in one-third octaves the function *Calculation/Advanced / EDT, RT (1/3rd)* would be suitable.) The horizontal lines in the last diagram are an average over the indicated frequency range of 500 to 4000 Hz. This display may be switched off under *Options (F9)*, see *VIEW & CALC / OPTIONS / SHOW LINES*.



## One-third octave spectra

Besides decomposing into octave spectra, EASERA is also capable of decomposing into one-third octave spectra. To do this one clicks on the button *1/3rds* in the filter section. Here it is possible to also view each of the bands separately.

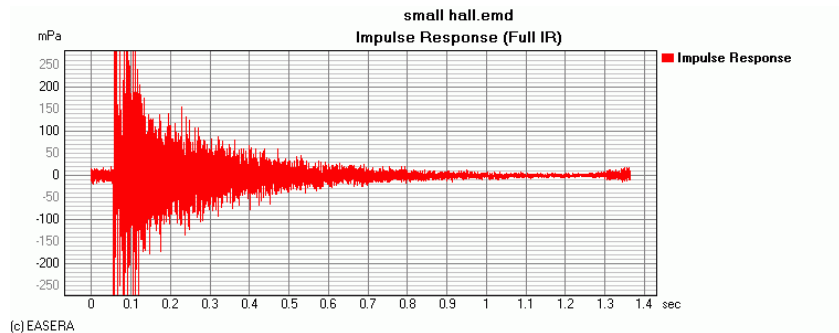
All usual diagrams can be displayed with the octave as well as with the one-third octave filtered spectra. Here we have an example of a frequency response plot with all one-third octave spectra overlaid:



In addition to the octave filters and one-third octave filters, EASERA offers the possibility to set any high-pass, low-pass and band-pass filters you like. This was already described in lesson 4.

## Windowing

As already shown in the last lesson, the distortion components of sweep measurements (White Sweep, Pink Sweep, Weighted Sweep) accumulate time-wise in front of the main impulse. When the main impulse is shifted to the beginning by using cyclical dislocation for the time correction, the distortion components are frequently arranged at the end of the impulse response.

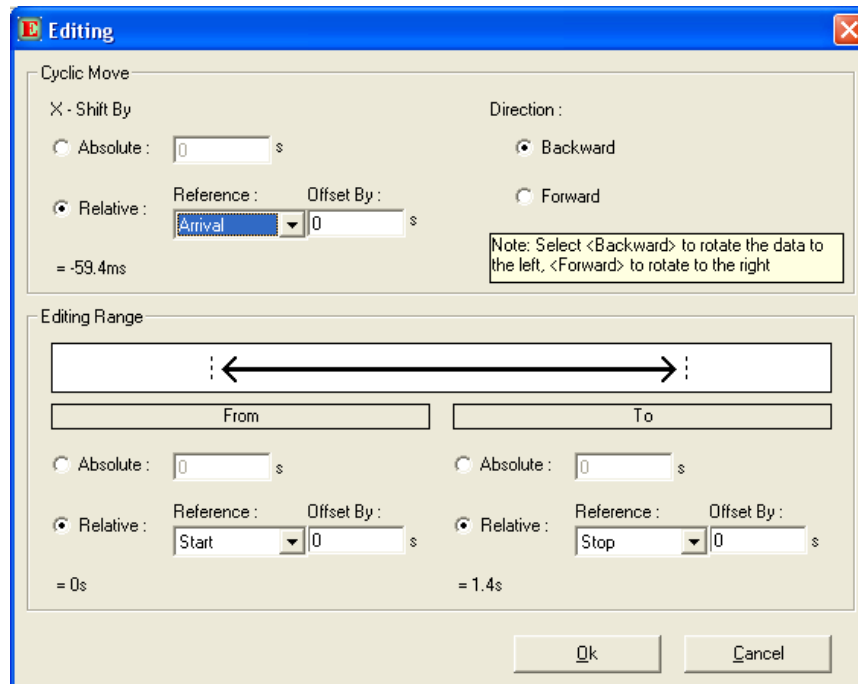


It may well be that distortion components are very interesting for measurements of electrical devices and especially loudspeakers, but if conclusions are to be drawn from the temporal behavior (ETC, reverberation time, ...), they normally only corrupt the results. Therefore we now want to remove these components using a window.

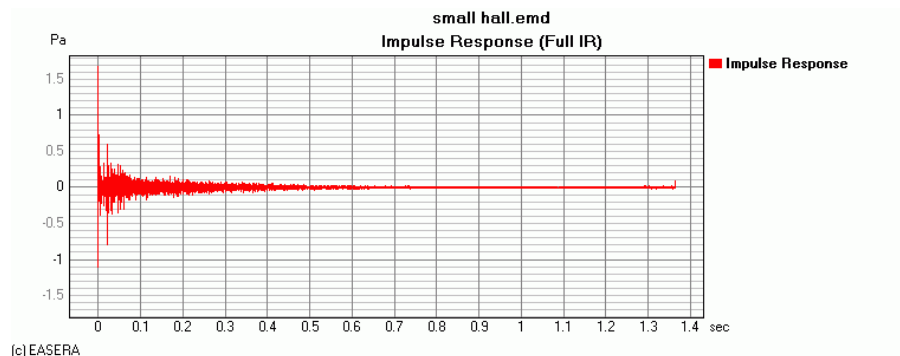
### Time-shifting a curve

On closer examination of the curve we note that the impulse only begins at almost 60 msec. What happens before is already part of the distortion components. As a first step, we will now shift the arrival of the first impulse to 0 msec.

If the impulse is as clearly distinguished from the preceding signal as seen in this example, we can simply select *Edit / Move Arrival to Zero*.



In the opened dialog everything is already appropriately set: We shift to the left, i.e. *backward*, by 0 s relatively to *Arrival*, which was detected here at 59.4 ms (in total we shift by 59.4 ms), and we shift the whole sector, i.e. from 0 s to 1.4 s. Retaining the preset values, the dialog may now be closed with *Ok*.



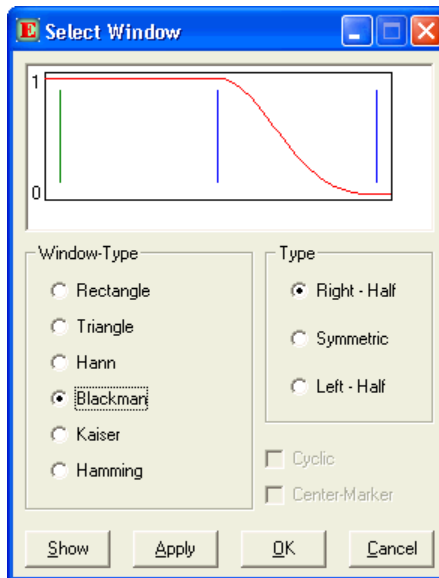
The impulse is now neatly located at 0 s.

## Setting the window

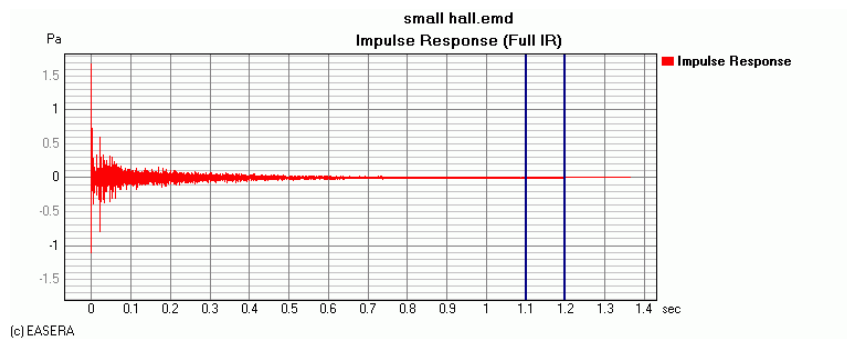
The limits of the window will be located at about 1.1 s and 1.2 s, so that all details after 1.2 s are eliminated. The limits are set using the two markers by choosing either the corresponding mode with *Mouse / Set Marker* or pressing the Ctrl key along with a mouse button. To set the left marker click with the left mouse button in the diagram at position 1.1 s, the right marker is set analogously with the right mouse button at 1.2 s.

Then we choose the *Edit / Window* function.



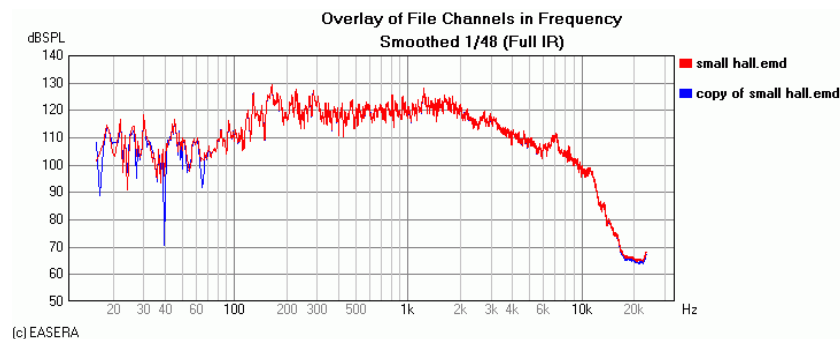


Selection of the *Right-Half* type is essential, since this way one ensures that the signal components to the right of the markers are removed by the window. To select how the signal is to be attenuated within the window limits a variety of shapes are selectable under *Window-Type*. As we are windowing here in a range where the level is per se rather insignificant, the type of window used is of minor importance. (*Blackman* is always a good first choice.) Finally we close the dialog with *OK* and apply the windowing.



As can be clearly seen, there is no signal after 1.2 s.

It has already been mentioned that a change in the impulse response also always affects the frequency response – "windowing out" the distortion components is no exception in this regard. The following diagram shows that the deviations are insignificant – this is no surprise at all, since the energy of the distortion components is also rather low.



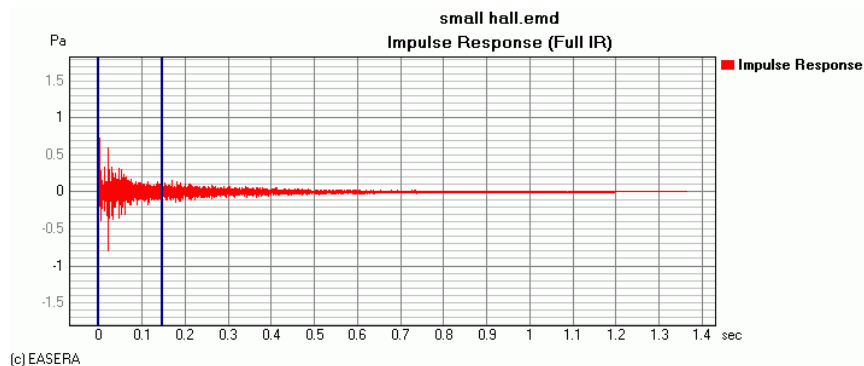
## Filtering

In a similar way we can also filter the data. While windowing is carried out in the time domain, filtering takes place in the frequency domain. Here the limits are also set using markers, and then *Edit / Filter* is selected – we have already practiced this in Lesson 4.

## Windowing the display

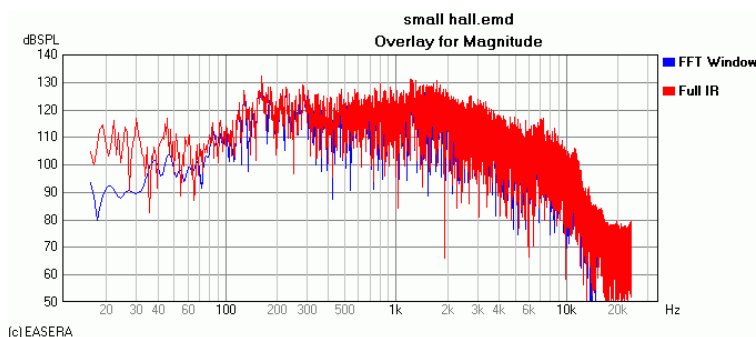
Like all other operations, windowing can also be undone in EASERA using the Undo function, thus enabling a comparison between windowed data and unwindowed data. But if one wanted to contrast both representations by an overlay, it would only be possible by creating a copy.

For this reason it is possible to perform a windowing for the display only. The window limits are also set here using the markers, but since symmetrical windows are always used (all details located outside of the markers are blinded out) we set the left marker to 0 s (most conveniently using the *Left Marker to Left End* button). To make the difference stand out more clearly, the right marker is to be set at approx. 0.15 s:



We then click on the *FFT* button in the visualizing area (on the right side, below *Zoom*). EASERA performs the windowing and changes directly to the unsmoothed frequency response.

Now we choose the *Add to Overlay* function and then click in the filter area on the *Full IR* button.

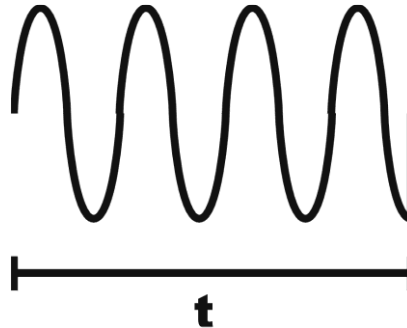


Now an overlay of the original data and the windowed data has been created. The windowed data are now available analogously to an octave or one-third octave band and selected in the filter area using the *FFT Window* button. (With this windowing possibility the *Rectangle* window type is always used. If other window types need to be used (Blackman, Kaiser, Hamming), windowing must be done in the way described before.)

## Window types

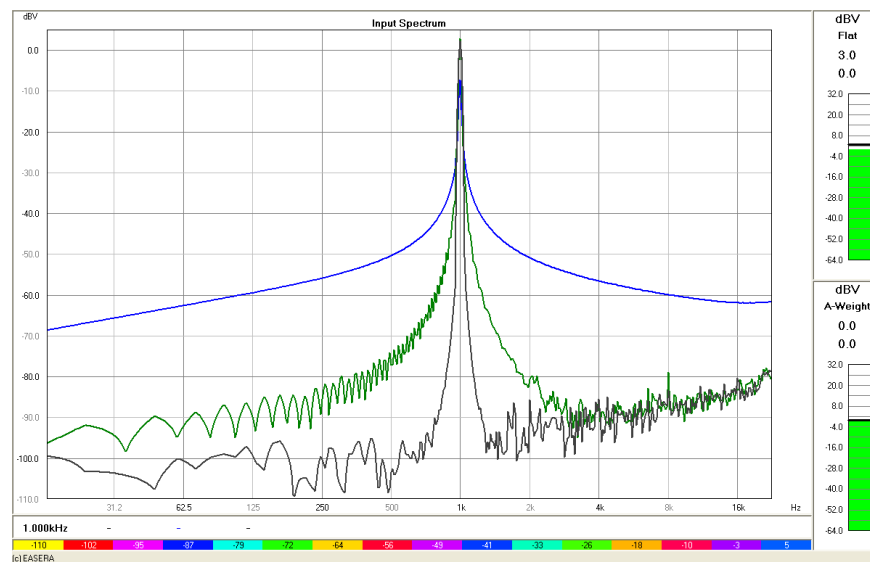
Let us assume we are subjecting a continuous sine tone of 1 kHz to an FFT analysis. Actually there should be just one result, i.e. a peak at the frequency of 1 kHz. This is unfortunately not the case.

An FFT analysis evaluates a certain period based on the sample rate and the number of time samples which are always a power of 2 (i.e. 256, 2048 or 32786). If this evaluating period does not have a whole number ratio with the period duration, the last half wave is partially cut off:



At this point harmonics are then created.

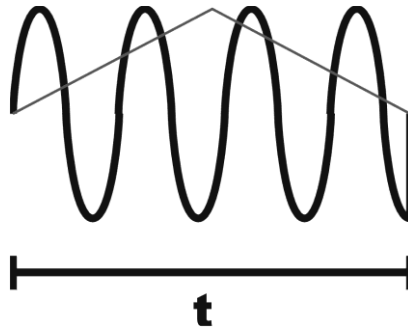
If we look at the 1 kHz signal in the live window, we see without further modifications the following spectrum (rectangular window: blue curve):



At double or half the frequency (2 kHz and 500 Hz, respectively) we are then approx. 50 dB below the useful signal. This value depends, however, largely on how much the signal is truncated at the ends of the evaluation period. Moreover, it would be possible with a pure tone to avoid a truncation by triggering at the beginning of the evaluation period, but not with a more complex signal that is commonly used in practice.

With some evaluations – for instance determination of the distortion components – one would like to know quite precisely the level for each frequency. Influences caused by the lack of windowing are not desirable in this context.

This is why the signal is differently weighted all along the evaluation period. This weighting is called a window. In the simplest case we have a linear increase up to half the evaluation period and then a linear decrease. This type of window is called *Triangle*.



The behavior of the triangular window is shown in the diagram above by the green curve. As compared to the rectangular window it gives an improvement of about 30 dB, although with many side lobes. Some mathematicians have designed windows which corrupt the results even less. As an example, the Blackman window (black curve) was included in the diagram. These window types are derived from trigonometric functions and are, as a rule, more continuous than the triangle.

It is true however, that distortions of a pure tone can be fully avoided only if the wave train fits exactly into the FFT length, i.e. if the transition from the last sample to the first one is continuous. To do this it is necessary for the evaluation time to be an exact multiple of the period length. In this particular case there is also no window required to reduce the boundary effects.

# Lesson 7: Displaying Results

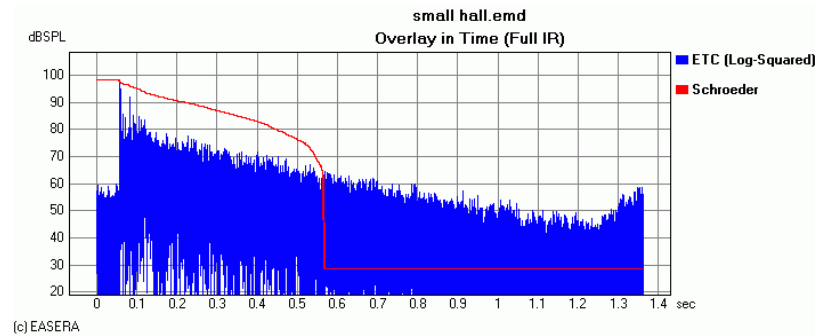
Measurements often need to be included in documentation, and EASERA also offers useful functions for this.

## Creating overlays

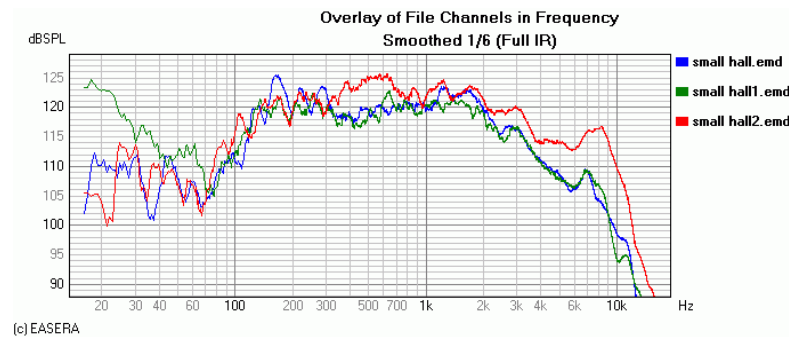
An overlay is the representation of various curves in a single diagram. To create an overlay just activate the *View | Add to Overlay* function and select the curves to be added one by one.

EASERA allows several types of overlays:

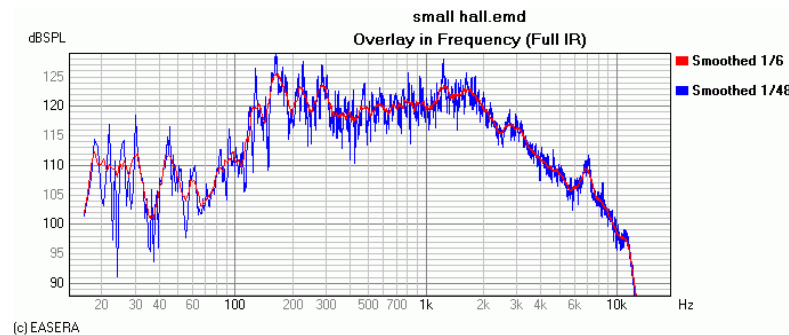
- Different curves in the same diagram, here e.g. ETC and Schroeder-Plot. Please keep in mind that only curves with coinciding axes can be combined in one overlay. A mutual overlay of frequency response and impulse response, for instance, would not be possible.



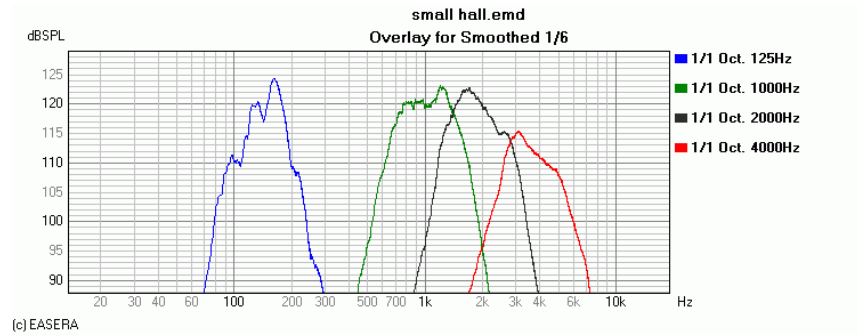
- Different measurements



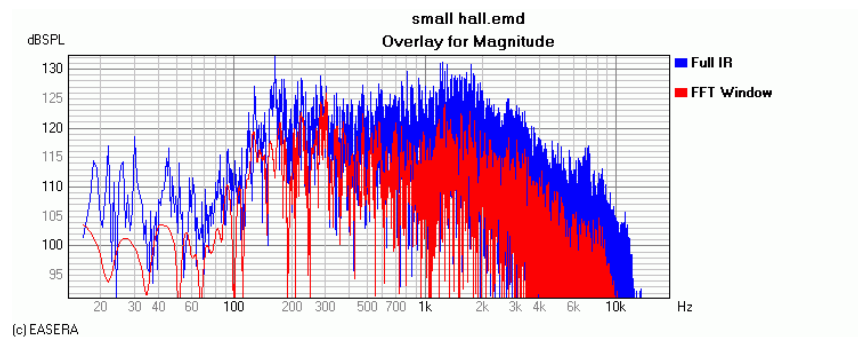
- Different smoothings



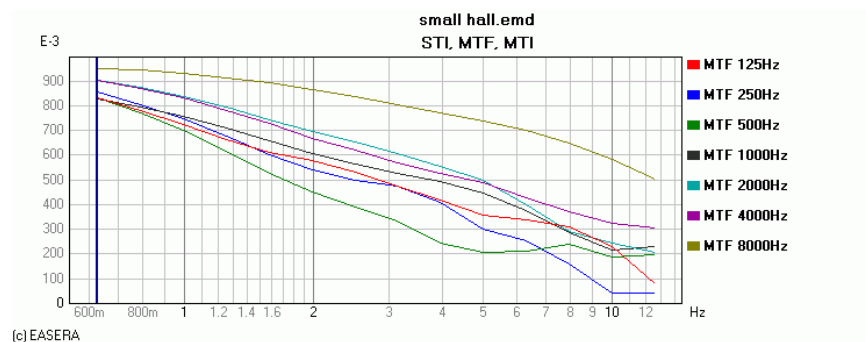
- Different one-third octave or octave bands



- Windowed and unwindowed representation



- For some curves in the *Calculation* area an overlay is automatically generated. In the present example the speech intelligibility is accompanied by the modulation transfer functions in an overlay.



## The Overlay navigator page

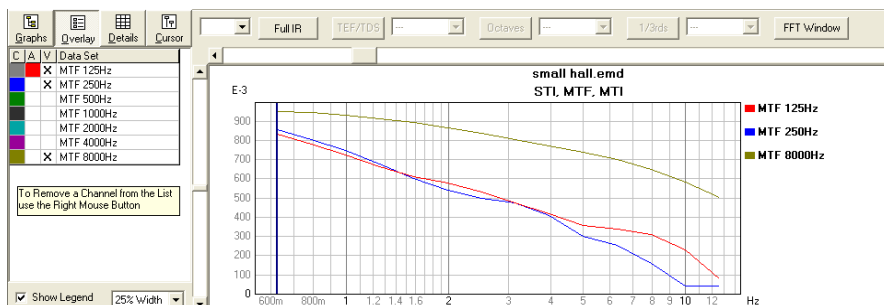
On the *Overlay* navigator page there is a list of all curves included in the overlay. This page offers you the following possibilities:

- You may set the active curve which is always indicated by the color red. All calculations are carried out exclusively with the active curve. (The color of the active curve can be set in the *Options (F9)* window GENERAL | COLORS page in the *Active Channel* section)



To set the active curve click with the left mouse button in column *A* (=Active) on the line to be activated.

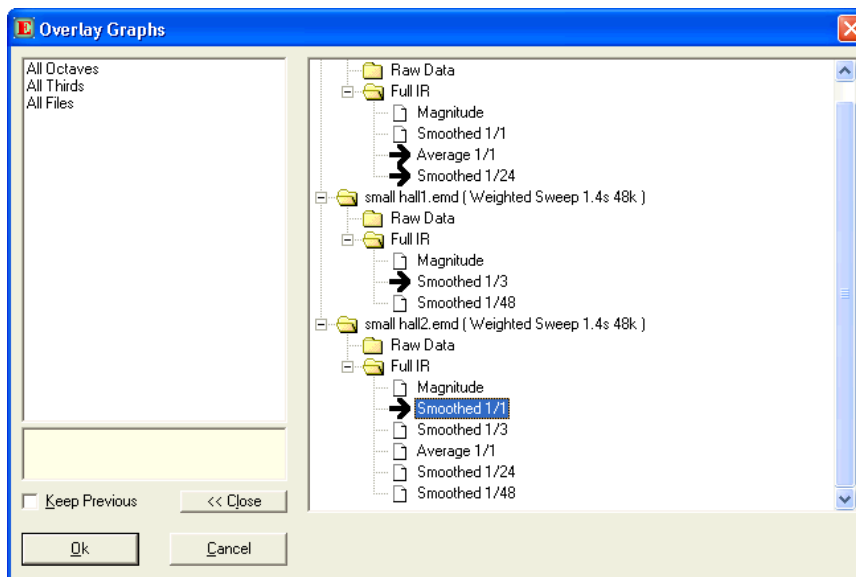
- You may show or hide the curves by clicking with the left mouse button in column *V* (=Visible) on the corresponding line.



- You may remove curves from the overlay by clicking with the right mouse button on the corresponding line.

## Select Overlay

The *View / Select Overlay* function opens a window showing the existing overlays and enables you to select one for display. You may also select from the curves already available in the existing data records those which you would like to combine in an overlay.





This display contains all data records which have been explicitly created since the loading of the file or the measurement.

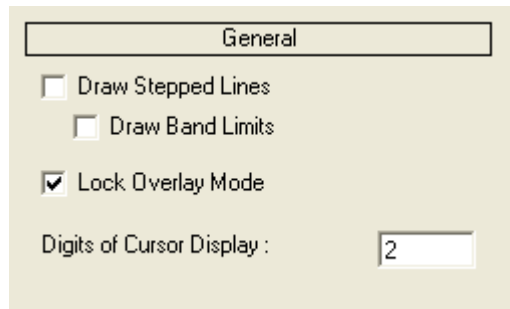
The *Select Overlay* function may also be selected using the following button:



## Add To Overlay

You may determine if the *Add To Overlay* function needs to be activated anew for each curve or if it remains activated as long as it is not explicitly deactivated.

To do this you select the *Lock Overlay Mode* option available in the *Options (F9)* window VIEW & CALC | OPTIONS page in the *General* section. If this option is activated, *Add To Overlay* remains activated until being deactivated by selecting this function again.



The *Add To Overlay* function may also be selected using the following tool button:



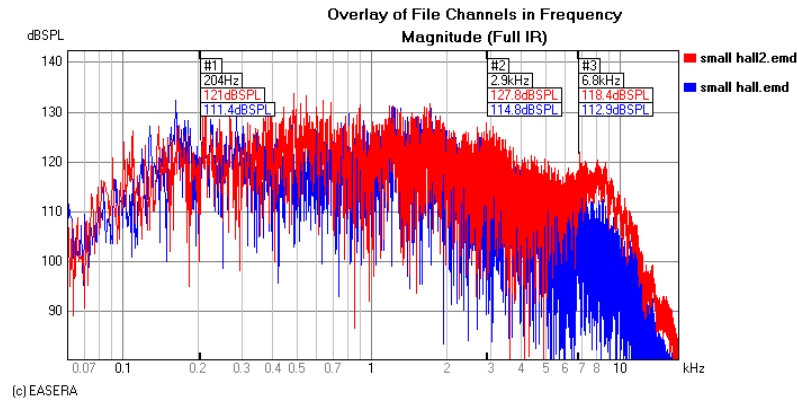
## Display Active Only

If all curves of the Overlay except the active one are to be removed from the display, one may accomplish this by using the *View / Display Active Only* function. This function can also be selected using the following tool button:



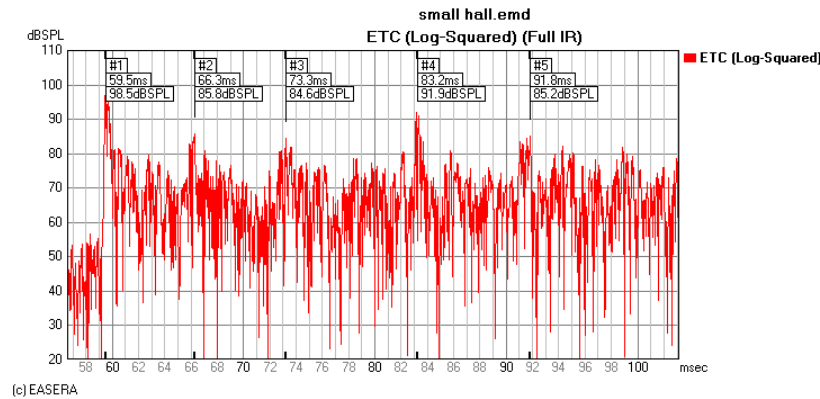
## Cursors

To mark certain spots in a diagram it is possible to set cursors. To do this one chooses either *Mouse / Peek* or one keeps the Shift key pressed while clicking on the desired location with the right mouse button.



The corresponding values of the X-axis (in this example the frequency) and of the Y-axis (in this example the level) of the curves are now shown at the cursor positions in the overlay.

Cursors are preferably also used in the ETC diagram to mark the individual reflections:



The maximum number of cursors can be set using the *Maximum Number of Cursors* option in the *Options (F9)* window VIEW & CALC | OPTIONS page in the *Cursors* section. Once the maximum number of cursors is reached, there are no more new cursors inserted by clicking with the left mouse button, instead the active cursor is relocated.

Cursors

Maximum Number of Cursors :

Values per Cursor :

Show Label (Default)

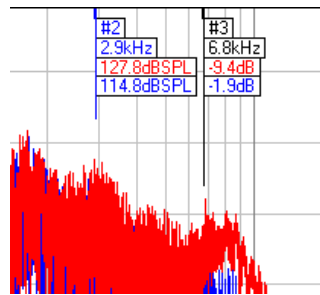
Show x - Value (Default)

## The List of Cursors

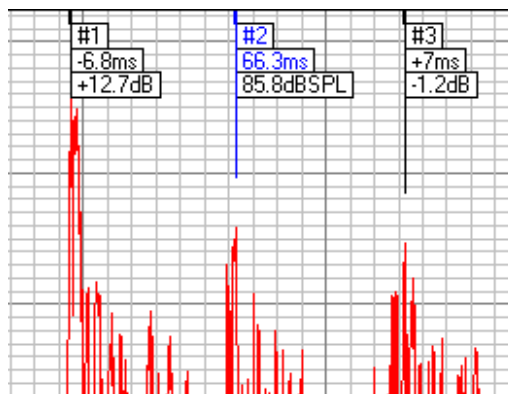
On the *Cursor* navigator page you find a list of the cursors. Here you may carry out the following actions:

C	A	R	V	Freq.	Label	
			X	204Hz	X #1	X
			X	2.9kHz	X #2	X
			X	6.8kHz	X #3	X

- To show or hide the whole cursor, click in the column *V* (=Visible) on the corresponding line.
- To show or hide the frequency display, click in the column after the *Frequency* column on the corresponding line.
- To show or hide the labeling, click in the column after the *Label* column on the corresponding line.
- You may have the values of the Y-axis shown relative to the values of a chosen cursor. To do this select a cursor by a corresponding mouse click in the column *R* (=Relative).



The frequency and the label of the selected cursor are now shown in blue in the diagram and the Y-values of the other cursors are shown relative to the selected cursor. In the time domain the time values are also shown relative to the cursor selected:



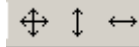
- To remove a cursor click with the right mouse button on the corresponding line. To remove all cursors click on the *Remove All* button below the list.
- In the *Sample* and *Mouse* modes one sets the cursor to the desired spot by clicking with the right mouse button. When using the *Snap* mode, the cursor is set to the maximum value within a range of +/- 3 pixels from the mouse position.

## Diagram region

To select the desired diagram region there exists quite a number of functions:

### Full

The *View / Full* function shows the whole diagram. Using the functions *Full Y* and *Full X* the whole range of the Y-axis or of the X-axis is shown, respectively. These functions can also be selected using one of the following tool buttons:



In the *Options (F9)* window one can preset on the VIEW & CALC | OPTIONS page as well as the VIEW & CALC | FULL VIEW page what is to be displayed for a complete representation.

If a time axis is involved, the total measuring period is generally shown, with a frequency axis the range shown is up to half the sampling frequency – the lower frequency can be set in the *Options (F9)* window VIEW & CALC | OPTIONS page in the *Lowest Frequency to Draw* section.

Lowest Frequency to Draw

0 Hz - Full Range

8 Hz                       32 Hz

16 Hz                       63 Hz

For the Y-axis range there are various possible settings for linear values and logarithmic values (dB scale):

Full View

<p><b>1</b> Linear Data</p> <p><input type="radio"/> Min. to Max. (asymmetric)</p> <p><input checked="" type="radio"/> Symmetric with Abs. Max.</p>	<p><b>4</b> Relative Border</p> <p>Horizontal [%]: <input type="text" value="5.0"/></p> <p>Vertical [%]: <input type="text" value="5.0"/></p>	<p><b>5</b> Result Data</p> <p>Use Relative Border:</p> <p><input checked="" type="checkbox"/> Reverberation Time RT</p> <p><input type="checkbox"/> Clarity C</p> <p><input checked="" type="checkbox"/> Definition D</p> <p><input type="checkbox"/> MTF/MTI</p> <p><input checked="" type="checkbox"/> Center Time</p> <p><input checked="" type="checkbox"/> Support ST</p> <p><input checked="" type="checkbox"/> Sound Strength G</p> <p><input checked="" type="checkbox"/> IACC</p> <p><input checked="" type="checkbox"/> Lateral Fraction LF/LFC</p>												
<p><b>2</b> Positive Linear Data</p> <p><input checked="" type="radio"/> Min. to Max.</p> <p><input type="radio"/> Keep Zero Visible</p>	<p><b>3</b> Logarithmic Data</p> <table border="0" style="width: 100%;"> <tr> <td>Maximum:</td> <td>Minimum:</td> </tr> <tr> <td><input type="radio"/> Round 1dB</td> <td><input type="radio"/> 60dB below Max.</td> </tr> <tr> <td><input type="radio"/> Round 5dB</td> <td><input checked="" type="radio"/> 90dB below Max.</td> </tr> <tr> <td><input checked="" type="radio"/> Round 10dB</td> <td><input type="radio"/> 120dB below Max.</td> </tr> <tr> <td><input type="radio"/> Relative Border</td> <td><input type="radio"/> 150dB below Max.</td> </tr> <tr> <td></td> <td><input type="radio"/> Auto Minimum (Rel. B.)</td> </tr> </table>	Maximum:	Minimum:	<input type="radio"/> Round 1dB	<input type="radio"/> 60dB below Max.	<input type="radio"/> Round 5dB	<input checked="" type="radio"/> 90dB below Max.	<input checked="" type="radio"/> Round 10dB	<input type="radio"/> 120dB below Max.	<input type="radio"/> Relative Border	<input type="radio"/> 150dB below Max.		<input type="radio"/> Auto Minimum (Rel. B.)	
Maximum:	Minimum:													
<input type="radio"/> Round 1dB	<input type="radio"/> 60dB below Max.													
<input type="radio"/> Round 5dB	<input checked="" type="radio"/> 90dB below Max.													
<input checked="" type="radio"/> Round 10dB	<input type="radio"/> 120dB below Max.													
<input type="radio"/> Relative Border	<input type="radio"/> 150dB below Max.													
	<input type="radio"/> Auto Minimum (Rel. B.)													

In the Linear Data (1) section one can adjust whether the scale range reaches from the minimum to the maximum or if it is arranged symmetrically about the 0-axis.

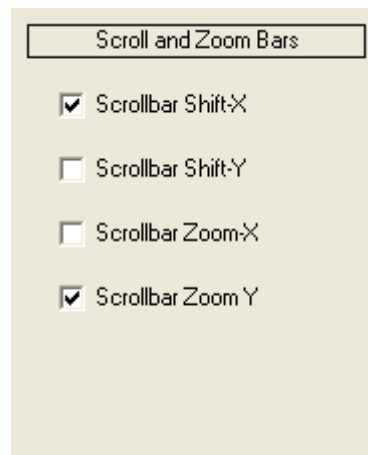
If only positive linear data are possible (2), one can adjust whether the 0-axis must always be shown or not.

With logarithmic data (3), the scale always exceeds the maximum up to the next rounded-off value. An option to round to 1dB, 5dB or 10dB can be selected. The minimum is then chosen as a differential value relative to the maximum.

For diagrams in which such settings are ineffective, it is possible using *Relative Border* to realize a kind of border around the curves shown (4). The value entered indicates the width of the border in percent of the diagram width or height. Under Result Data (5) it is then possible to determine the diagrams for which this border is to be used.

## Zooming

There are several options available to zoom into a diagram. First there are up to four scroll bars arranged around the diagrams. The upper and right scroll bars allow scaling the representation to the desired dimension, whereas the lower and left ones allow the representation to be shifted to the desired position. Which scroll bars are visible can be adjusted in the *Options* (F9) window VIEW & CALC | LAYOUT page in the *Scroll and Zoom Bars* section.

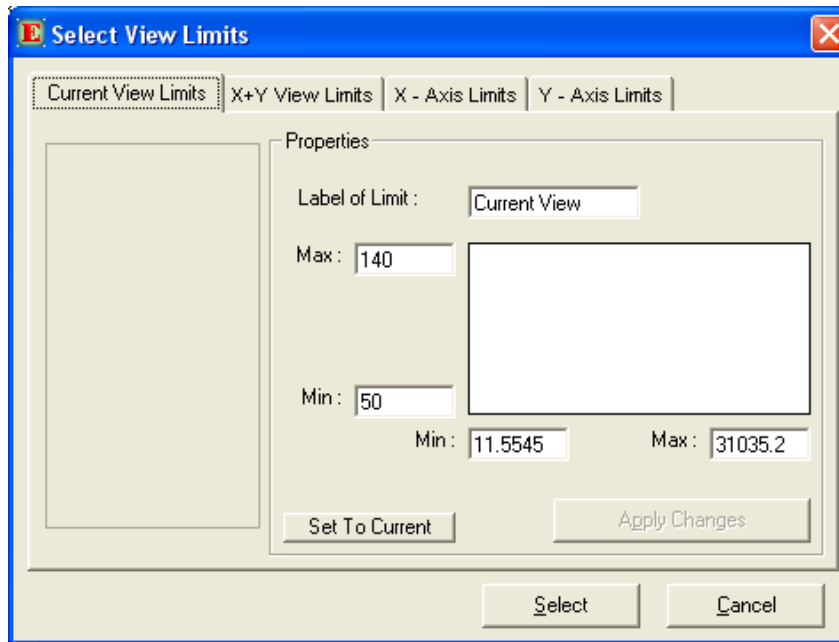


It is also possible to zoom by selecting the desired area with the mouse using one of two modes: With *Mouse / Zoom X / Zoom Y* one can choose the region to be shown separately for the two dimensions, i.e. with the left mouse button for the X-axis and with the right mouse button for the Y-axis.

If one chooses the *Mouse / Zoom* mode, the display is enlarged exactly to the frame drawn in the diagram. To avoid having to constantly switch the mouse modes, all available mouse modes can also be activated by control-key combinations: The *Zoom X / Zoom Y* function becomes available by holding simultaneously the Ctrl and Alt keys, and the *Zoom* function by holding simultaneously the Shift, Ctrl and Alt keys.

## View Limits

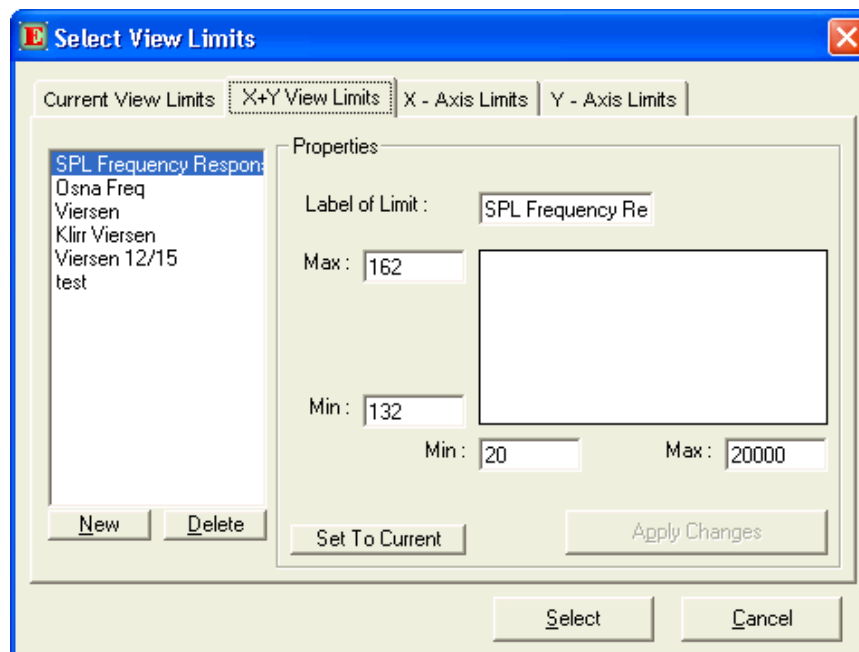
Especially when using diagrams for documentation, one usually wants to have identical view regions for all diagrams. This can also be easily done with EASERA: With a mouse click on one of the two scales the *Select View Limits* window is opened.



Now there are several tab pages available:

On the *Current View Limits* tab the actual diagram limits are shown. These can be overwritten by other values, but they are not stored.

If one wants to repeatedly assign the same limits, one should change to the *X+Y View Limits* tab:



Using the *New* button it is possible to create new data sets and to change their name in the *Label of Limit* field. The limit values are shown and modified in the fields *Min* and *Max*. These data are then stored by clicking on *Apply Changes*, causing them to be kept available.

To apply these stored limit values, they are then just selected from the list and the dialog is closed with the *Select* button. On the other two tabs it is possible to store and select limits for either the X-axis or the Y-axis separately.

### ***And now a still more simplified version***

Using *View / Store Current View Limits* it is possible to store one single set of limit values which can then be recalled using *View / Recall View Limits*. These items not only can be selected via the menu, but also – like many other menu items – via the button bar.


To return to the last chosen zoom region one may also use *View / Last View*.

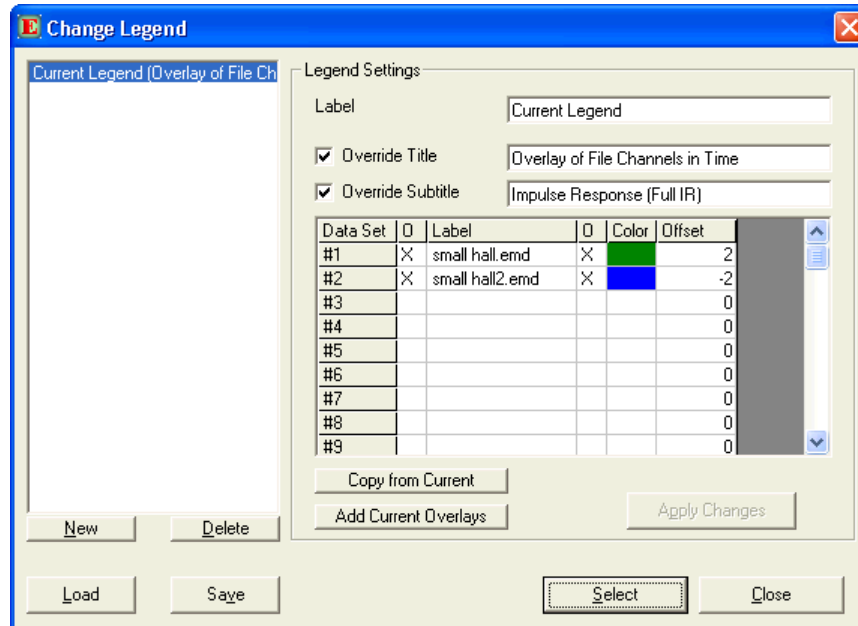
## Exporting diagrams

To export diagrams there exist on principle two ways:

- Using *File / Send Picture To / Clipboard* they are stored as a bitmap in the temporary memory.
- Using *File / Send Picture To / File* they are stored in the chosen graphic format. EASERA supports all common pixel graphics formats (bmp, jpeg, gif, png, tif, lwf, pcx, ico, emf, tga).

## Change Legend

Under *View / Change Legend* (Button ) there are further options to configure a diagram available:

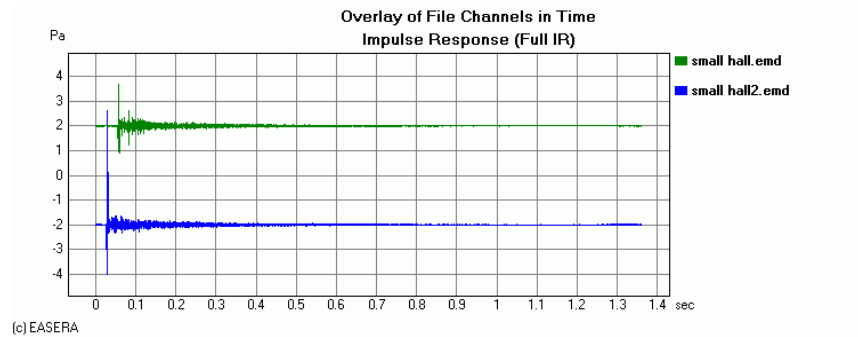



Here you can create several legend patterns which you may then assign to the respective diagram. These patterns may also be saved and transferred to another computer.

The *Label* of a legend is just its name for application in this window. It is not shown in any other place. The next two lines concern the title and the subtitle of the legend where it is also possible to determine whether the titles generated by EASERA are to be used or overridden.

The legend (*Label*) and color (*Color*) can be determined for each one of the 32 displayable channels. In the column *O* arranged before the Label or Color column one specifies whether the settings generated by EASERA are to be overridden in the course of the legend change.

It is also possible to activate an *Offset* function which allows the curve to be displaced vertically. This is very suitable to show several impulse responses, for example, one below the other without major processing:



The legend adjustments apply only to the present diagram. They are discarded when a new diagram is generated and they are also not restored when returning to the original diagram. Using the *View / Apply Last Legend* menu item (there is also a corresponding button  available on the button bar) it is, however, possible to recall the last legend setting very quickly.



# Lesson 8: Processing

Measurements not only capture the desired quantities, but often are also subject to the influence of disturbances. To minimize this influence, EASERA offers a variety of processing options.

Such processing not only allows minimizing disturbances, but also opens a wide scope for creative design: From the use of filters and windowing (this was already dealt with in Lesson 6) via mathematical operations up to time shifts, EASERA enables all manipulation options required in practice. These are classed under the term *Processing*.

As a rule similar measurements should be processed in the same way. EASERA will allow the processing used for one set of data to be used on other data.

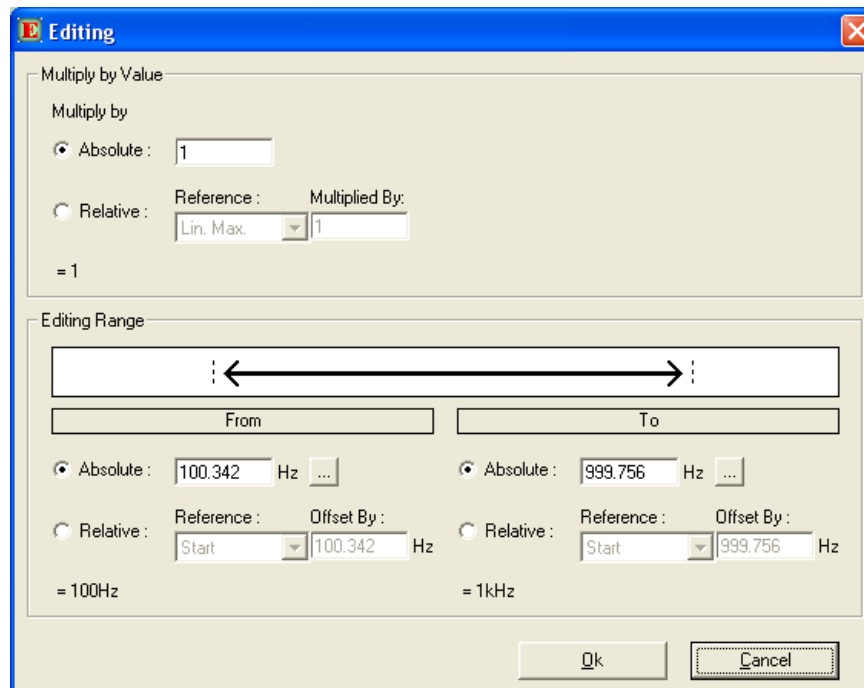
## Mathematical operations

The following mathematical operations are available:

- Setting to a fixed value
- Addition and subtraction
- Multiplication and division
- Exponentiation

In addition there are menu items which will initiate one of these operations by presetting parameters in a specific way. All operations work with linear data.

If one of these operations is to be carried out, the corresponding menu item opens a dialog that looks more or less as follows:



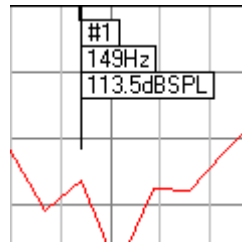
This dialog was opened from an unsmoothed frequency response using the menu item *Edit / Multiply and Divide / Multiply by Factor* with the markers set to 100 Hz and 1 kHz.

The first question that arises is why the entered 100 Hz and 1 kHz are not used. This requires some detailed explanation:

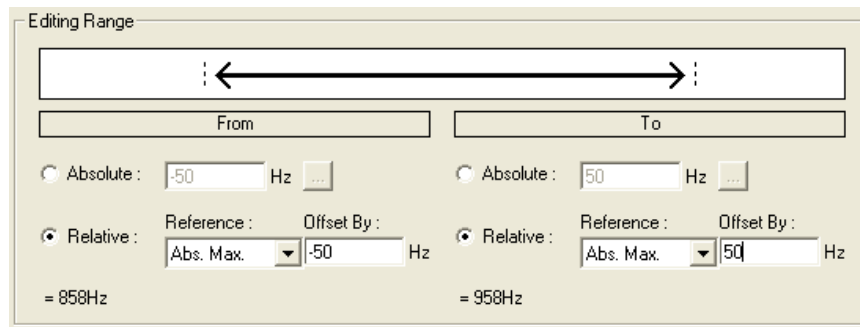
The data used here were recorded with a sample rate of 48 kHz and have a length of 1.365 s. There exist 65536 time samples (which are numbered from 0 through to 65535). If an FFT analysis is carried out with these data, one obtains  $x/2 + 1$  frequency samples (which are numbered from 0 through 32768). These frequency samples are evenly distributed over the range from 0 Hz to half the sample rate, i.e. 24 kHz. The frequency resolution is thus 0.732421875 Hz (rounded in the EASERA display to 732 mHz).

100 Hz divided by 0.732421875 Hz would amount to 136.53, but there are only whole-number samples. This means that EASERA has to round off and does so to 137 which results in a frequency of 100.3417969 Hz rounded to 100.342 when multiplied by 0.732421875 Hz. The value of 1 kHz divided by 0.732421875 Hz would yield 1365.3 which is rounded off to 1365. The actual frequency is thus 999.7558594 Hz, rounded to 999.756 Hz.

Cursor positions also lie on these actual frequency points although their specified frequency is appropriately rounded:



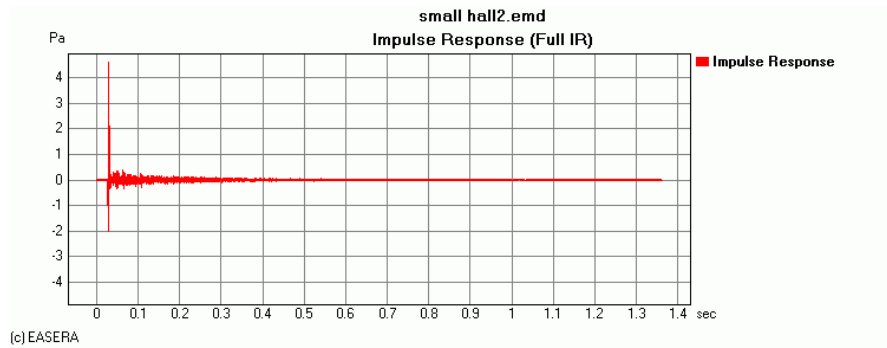
As long as not specified otherwise, the specifications are absolute values. But you may also refer your inputs to other points: In the next picture the frequency inputs are referred to *Abs. Max*, which is at 908 Hz. Processing took place in this instance within a range of +/-50 Hz of this frequency.



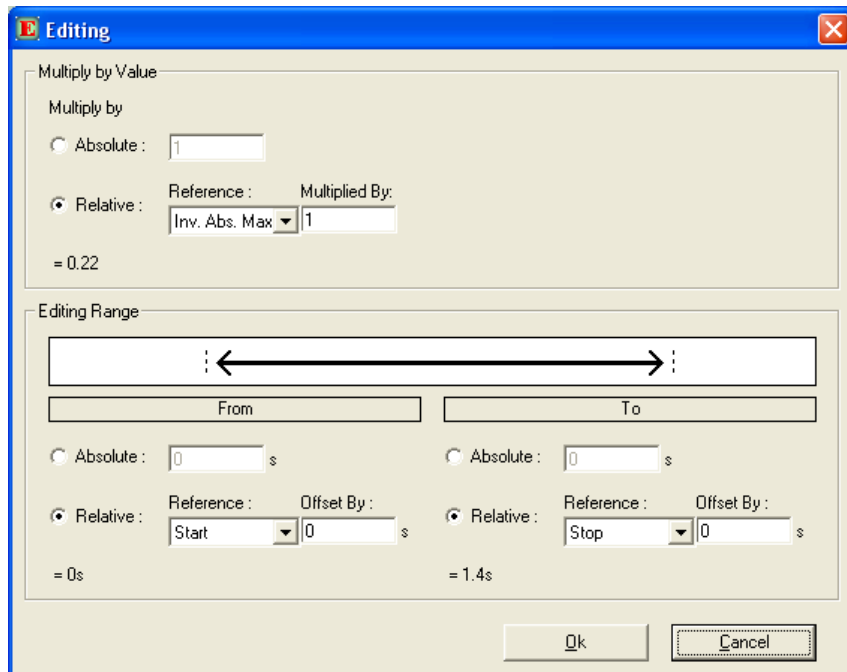
The value to be multiplied by is similar: If the values are to be "amplified" by 6 dB one enters the value 2 into the *Multiply By Absolute* field. (We deal here with voltages and not with power quantities. Doubling therefore corresponds to an increase of 6 dB and not to 3 dB.)

### Relative calculation values

The multiplication factor may, however, also be set relative to a value. Let us assume we wanted to standardize the following impulse response to 1 Pa (i.e. the maximum shall also be 1 Pa):

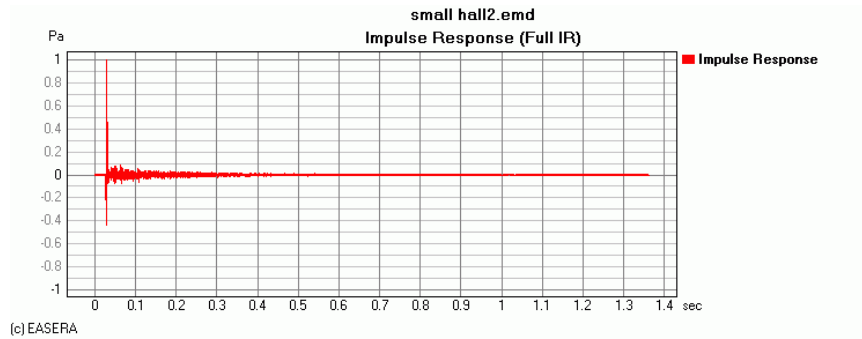


To simplify matters we select the menu item *Edit / Multiply and Divide / Scale Abs Max to Value (by Multiplying)*:



First let us deal with *Range*: Whenever we are working in the time domain without having set any markers, EASERA uses all data for processing. In this case *From* is set to 0 under *Absolute* and *To* to the maximal value, which results here in the temporal end of the data range.

Multiplication takes place with the value 1 which is set relative to the reciprocal value of the absolute maximum (*Inv. Abs. Max.*). With a maximum of 4.6 this results in a reciprocal value of 0.2173913 which is indicated here in rounded form. After closing of the dialog the impulse response is normalized to 1:



## Remove DC

In some configurations it may happen that the impulse response gets displaced on the Y-axis. In the realm of analog circuitry one would suppose a DC offset to be responsible. EASERA offers here a simple way of correction: Using *Edit / Add and Subtract / Remove DC* the subtraction dialog is adjusted in such a way that one equal portion gets eliminated:

Subtract Value

Subtract

Absolute :  Pa

Relative : Reference :  Offset By:  Pa

= -136.63 $\mu$ Pa

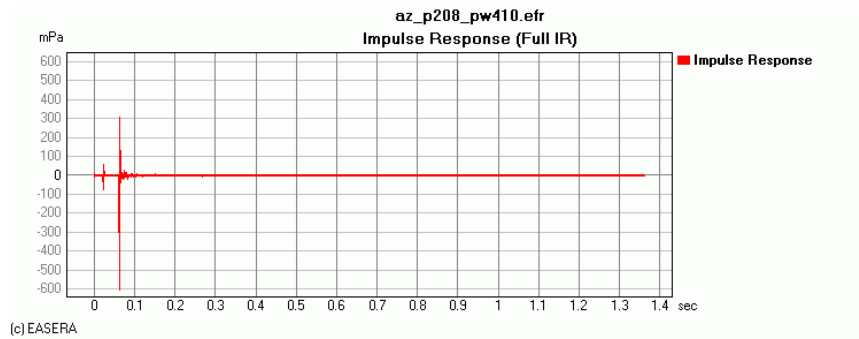
EASERA computes here the average of all values (which with a pure AC signal should be 0) and offers to subtract this average from all values. (With a new average computation the result would be 0.)

## Displacements

If measurements are carried out at points varying in distance from the sound source, the impulse arrives at different moments. From these temporal differences it is possible to determine the range differences with some accuracy. If evaluations in the time domain are to be compared, however, these differences are not very welcome. For creating a phase response diagram it is, moreover, necessary to remove such run-time differences, since the results would otherwise be superposed by a run-time phase.

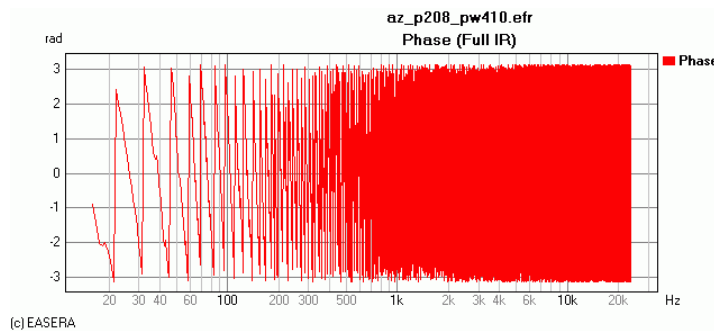
Displacements on the time axis are always of cyclical nature: What at one end of the diagram is "pushed over its edge" is rejoined at the other end. This means the data before the time that we move to time 0 will now appear at the end of the impulse response. Should this be undesirable, the components rejoined at the other end have to be removed using *Edit / Set To / Zero* or a corresponding window.

The following diagram shows the impulse response of a three-way system at a distance of about 20 m (The measurement has been made in a half-open hall by using the mic MBC-550 on a stand at ear height. The speakers p208 and pw410 have been arranged with the tweeter on top of the woofer.):

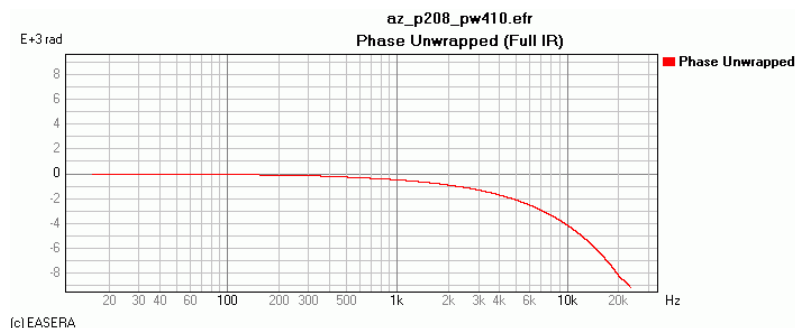


The fact that the largest impulse is negative is due to the polarity of loudspeaker and/or measuring microphone and does not matter at all. In front of the impulse proper we find two further ones, a very small one and one of approx. -74 mPa – these are distortion components. Later on we will deal with this topic in detail.

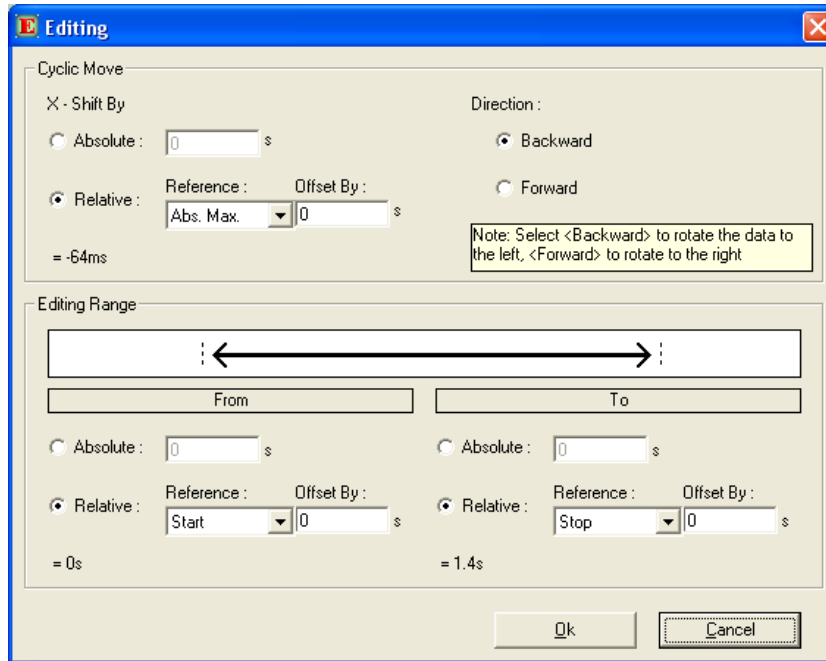
If we are now going to show the phase response using the function *Phase*, this is determined mainly by the delay up to the arrival of the main impulse:



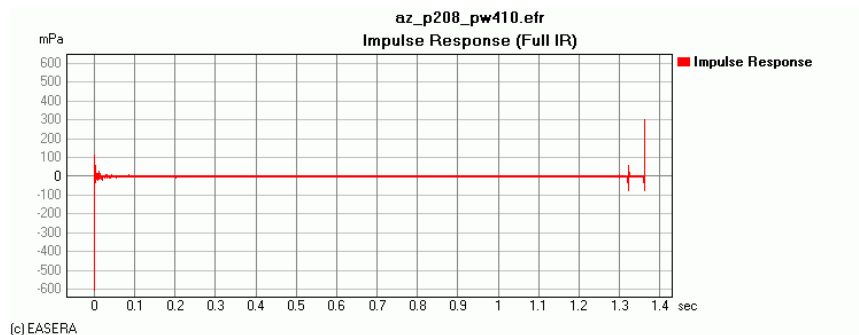
This becomes clearer if we use *Phase Unwrapped*:



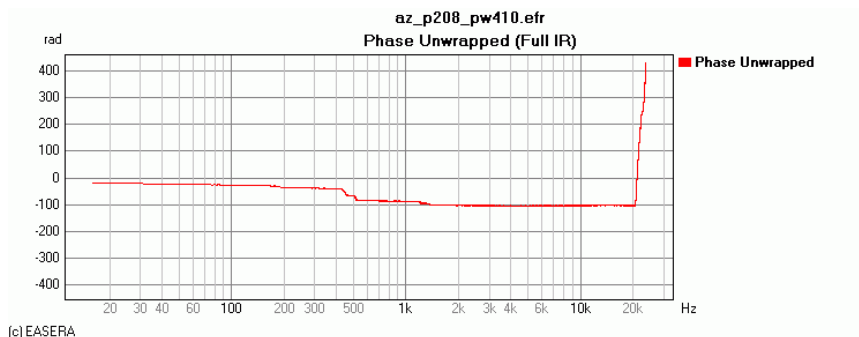
This diagram does not allow us to gather any information on the phase behavior of this sound reinforcement system. Therefore we are now going to select the menu item *Edit | Cyclic Move | Move Abs Max to Zero* (directly from the impulse response):



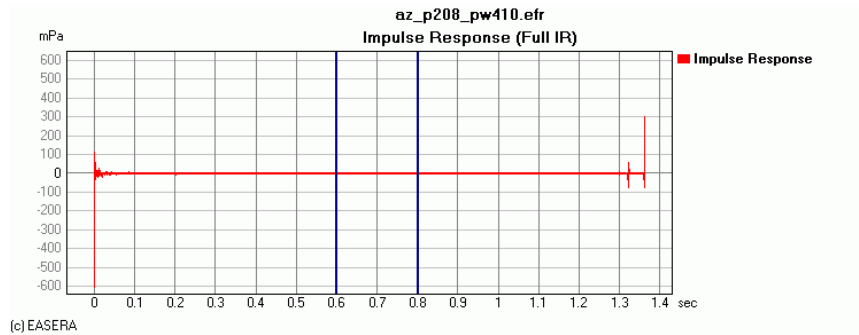
The dialog is preset in such a way that the absolute maximum – which at the moment is still at 64.05 ms – will be shifted to time 0. Thus we need only to confirm with *Ok*.



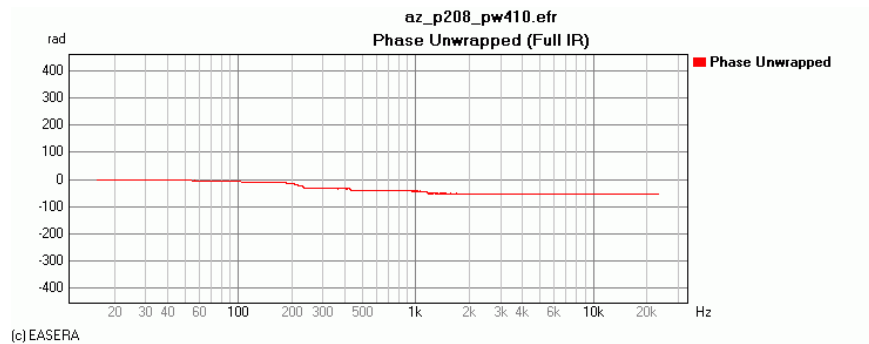
Since the main impulse is several samples wide, we now have a part of it at the end of the impulse response, and in front of this, clearly distinguishable, the 2<sup>nd</sup> harmonics.



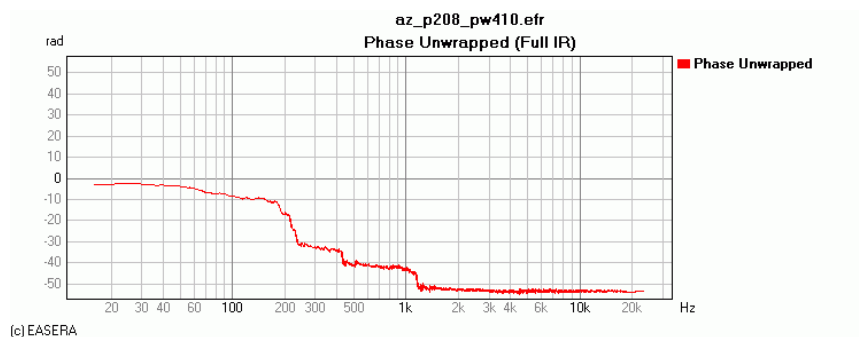
The phase response is now a bit more linear, but shows a steep rise shortly after 20 kHz. We want to also explore this state of affairs and set the markers in the impulse response to about 0.6 s and 0.8 s.



Now we window out the end of the impulse response using *Edit / Window (Type Right-Half)*. First we view the phase response with identical axial scaling:



This really looks much more linear. Even when amplified to *Full Scale* we still have a rather unspectacular behavior:



We now see rather distinctly the two frequencies at which the crossover filter separates: 220 Hz and 1.2 kHz.

To enable conclusions to be drawn about the phase response of a loudspeaker or loudspeaker system the travel-time phase needs to be removed first, otherwise the diagram would be wholly dominated by the same.

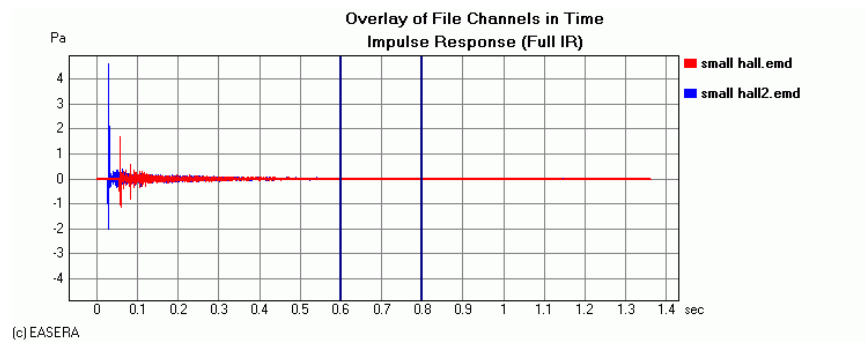
## Move Arrival to Zero

Whereas the absolute maximum is usually shifted to 0 when phase responses are desired, with room acoustical measurements the interest centers primarily on the *Arrival*, i.e. the first appreciable amplitude. Instead of *Move Abs Max to Zero* we use here *Move Arrival to Zero*. This avoids shifting significant components of the main impulse to the end where they corrupt the results. Especially when calculating the reverberation time the result can be very erroneous if *Move Abs Max to Zero* is used.



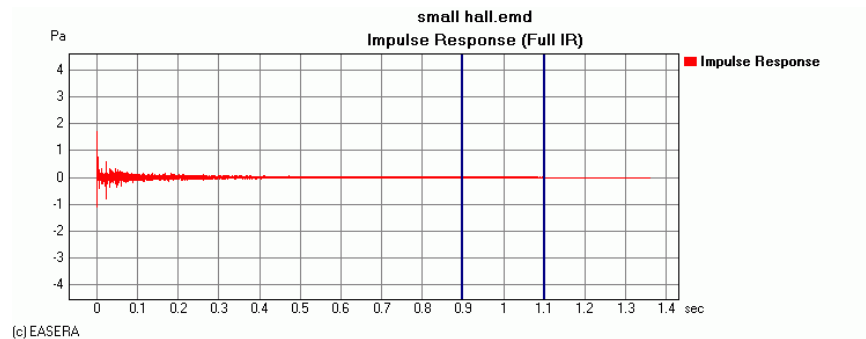
## Editing Sequence

In the following diagram we see an overlay of two measurements:

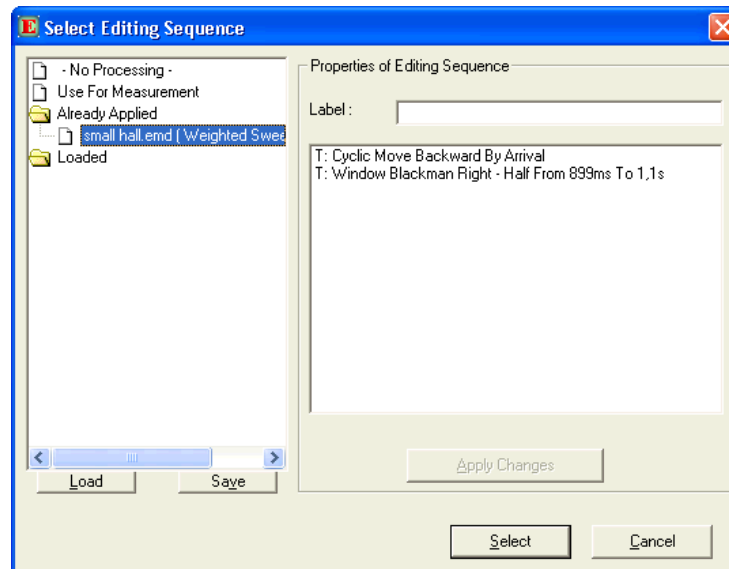


Now we use one of the two curves to perform the following processing steps:

- Using *Edit / Move Arrival to Zero* we shift the curve "to the beginning".
- To avoid the influence of the distortion components, the markers are set to 0.9 s and 1.1 s. Then the rest of the curve is windowed out (menu item *Edit / Window*, Type *Right-Half*, the window type is insignificant in this context).



Now we are going to also apply these two processing steps to the other curve. We choose this curve and select the menu item *Edit / Editing Sequence* to open the *Select Editing Sequence* dialog box.



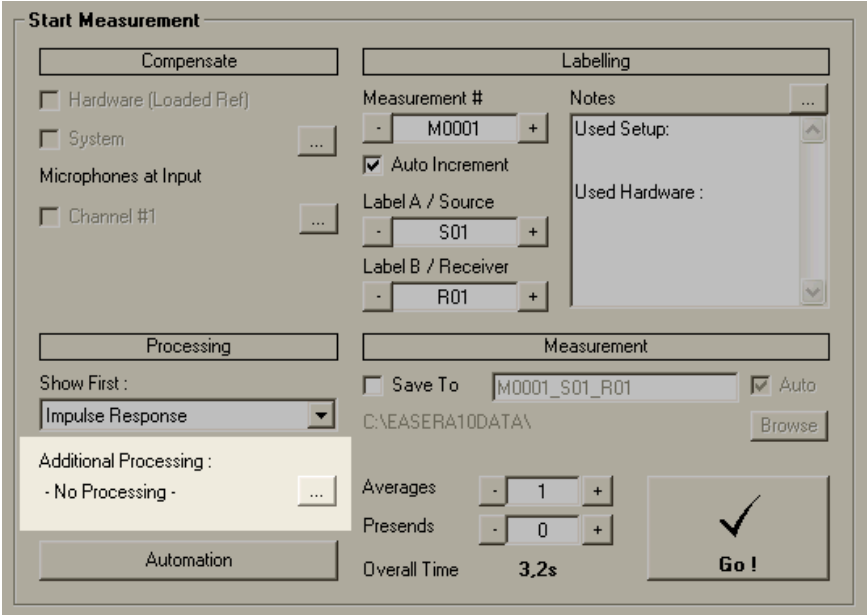
Under *Already Applied* we find all processing sequences that were applied to the existing curves – in this case only one. If we choose this item, all the processing steps are shown on the right-hand side, in this case the shifting of the arrival and the window that was applied. Using the *Select* button we may now apply the processing sequence to the active curve.

Such a processing sequence can also be stored by clicking on *Save* (and then reloading with *Load*) and thus remain available after closing the program. (Prior to this it is possible to assign a name to the sequence in the *Label* field and confirm it with *Apply Changes*.)

## Processing for any new measurement

If the same processing is always to be applied to any new measurement, it is very helpful if the processing steps do not have to be performed one by one, but can be done by simply assigning a processing sequence. In the long run, however, this is also cumbersome, especially if a lot of measurements are involved. This is why EASERA offers the option of assigning a processing sequence a priori to all new measurements.

To do this we go to the *Measure* page in the *Start Measurement* section and click on the button with the three dots in the *Processing* area. This will open the just used dialog box.



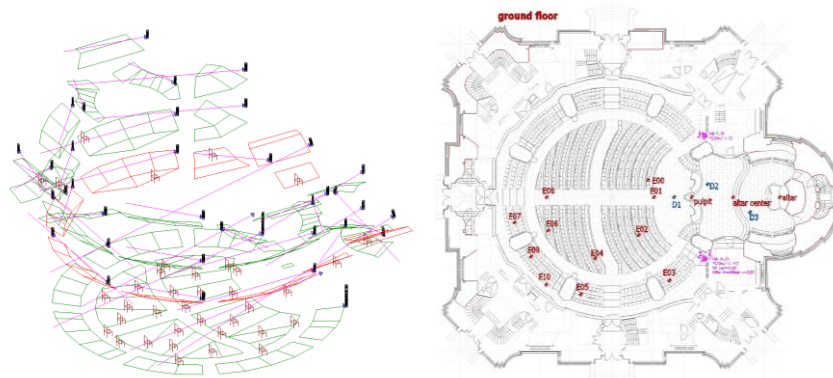
Even if the processing is assigned right after completion of the measurement, the original data remain available nevertheless and the processing can also be eliminated at any moment. Undo/Redo also remain available!

# **Lesson 9: Measurements at diverse room positions**

In this lesson we will see how (room) acoustical measurements are carried out and evaluated.

## Frauenkirche Dresden

The church room of the Frauenkirche in Dresden (Germany) shall serve as our measuring subject. The following picture shows a wire-frame model taken from the EASE simulation program depicting the arrangement of the source and measuring positions in the parquet:



The room acoustical part of the measurements was carried out with a dodecahedron as the source and dummy heads as dual-channel receiving and recording devices. Then the sound reinforcement system was tuned using recordings that were made once more with the dummy head as well as with an omnidirectional measuring microphone. The measurements were taken at representative listener positions. For better orientation the measuring positions in the parquet of the church nave are depicted in the plan view above.

### Realization of the measurements

Acoustical measurements are carried out to acquire the usual room acoustical quantities (reflections, reverberation, clarity values, speech intelligibility). These values are independent of the absolute level value (this assumes a low noise floor). In this case a calibration is recommended, but not required.

The mentioned quantities were acquired at 14 selected positions in the parquet. If the soundcard used provides a sufficient number of input channels and moreover, if a sufficient number of measuring microphones are available, one can install a microphone at each of these positions and measure them all simultaneously.

If this possibility is not available (and that was the case when making the measurements in the church), all locations are measured successively.

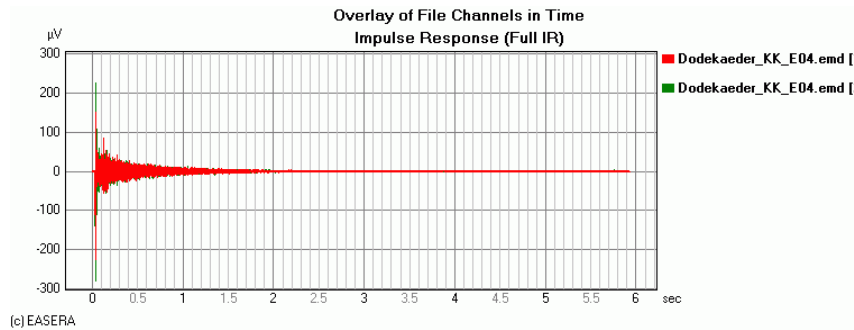
A Weighted Sweep of 5.9 s duration was used as the stimulating signal. Since no significant disturbing noises were to be expected, we averaged over two measurements only.

## Evaluation of the measurements

We shall now have a closer look at the dodecahedron measurement with the dummy head at the selected position E04, see figure above. The dodecahedron was set up near the pulpit.

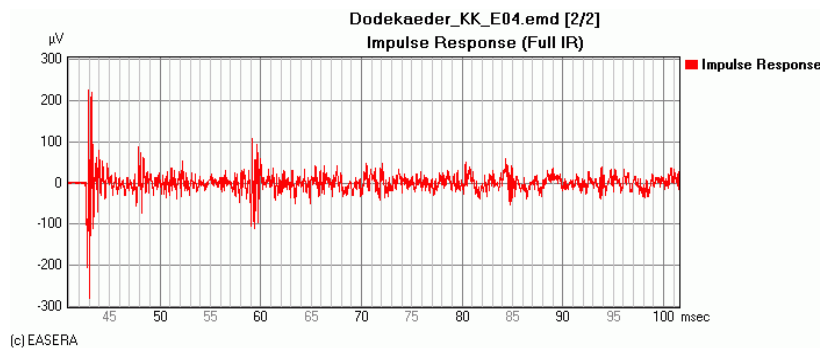
### The reflections

Our first glance is at the impulse response:



Viewing it in full length, there are no peculiarities to be seen. Beware of any attempt to read the reverberation time directly from this diagram. The resolution of the screen is far too coarse for this purpose. For further evaluations we will use the impulse response of the right ear.

We now want to see if there are isolated significant reflections and select the region from 40 ms to 100 ms. To zoom into the diagram *Zoom X / Zoom Y* is activated by a mouse click and the corresponding area is selected on the time axis while keeping the left mouse button pressed.



The first peak is at 43 ms; multiplied by the sound velocity this would result in a distance of 14.62 m between sound source and microphone.

The first reflection is at 48 ms. Using the *Peek* mouse mode (or the Shift key) it is possible to easily measure the time difference from the direct sound which is 5 ms here and corresponds to 1.7 m. This could be a reflection from a pillar.

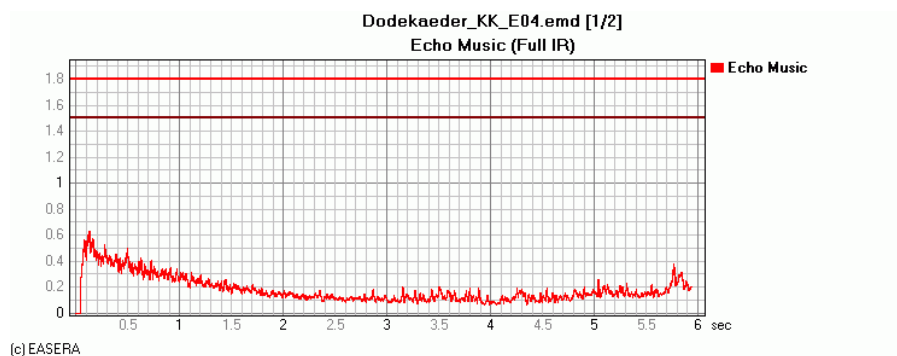
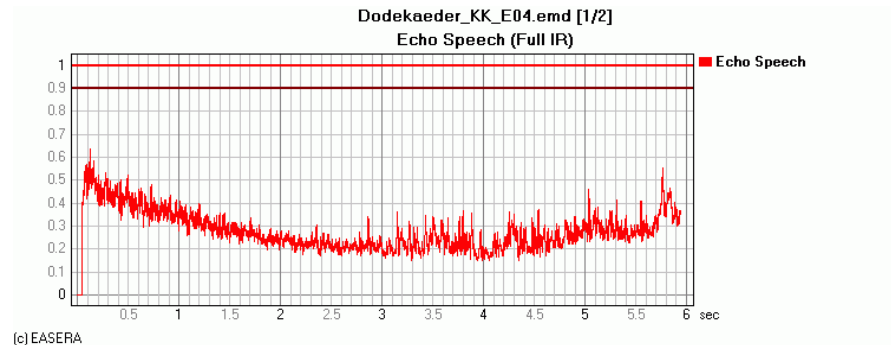
The next distinct reflection is at 59 ms, which corresponds to a difference of 16 ms from the direct sound – this converts into a distance of 5.44 m. Using trigonometric functions it was possible to identify another pillar behind the recording position as the reflection surface. During the measurements it is also possible to localize possible reflection surfaces by using a mobile absorber material (plastic panel or large piece of garment) to provoke directionally dependent acoustical shadowing effects. The disturbing reflecting surface can then be easily localized by the visualized level attenuation.

Since the room temperature is rarely recorded with all these measurements, there reigns a slight uncertainty as to the actual sound velocity. Considering that slight inaccuracies cannot be avoided with the positioning of loudspeakers and microphones, the sound velocity also cannot be determined from the instant of arrival of the reflections.

## Echo criterion

Even if the first reflection develops a rather distinct peak, a distinct echo is not to be expected, because the reflection follows too close to the direct sound.

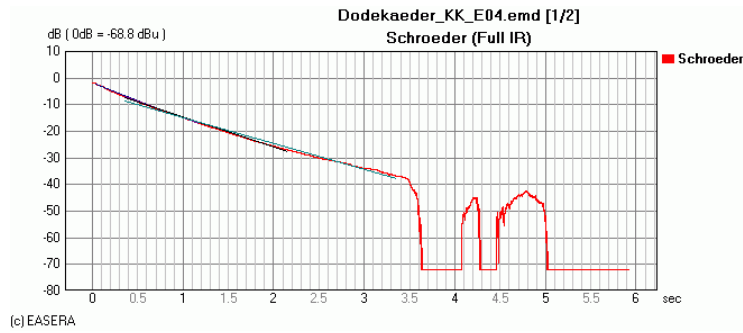
To complete things let us have a look at the echo criteria for speech and music:



As long as the curve remains below the brown line, no echoes can be discerned even by trained ears (the red line would apply to untrained ears), and our curve is still far below either of these lines.

## The reverberation time

Next we shall have a look at the reverberation time; to do this we select the *Schroeder RT* item in the *Calculation* section on the *Graphs* navigator page.



The broad-band reverberation time is determined from the Schroeder diagram in which the energy is summed starting from the end, back over the whole duration of the measurement.

To this integral there are four different tangents applied. These tangents are extrapolated to  $-60$  dB to allow an assessment of when the diffuse sound field has decayed by 60 dB.

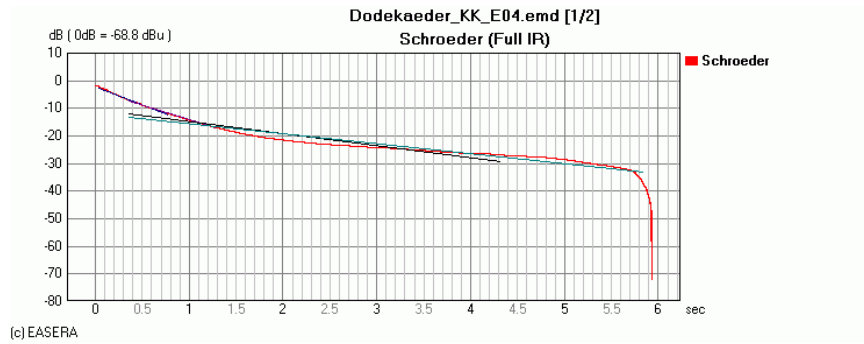
	RT	
EDT	4.38	s
T10	5.00	s
T20	5.36	s
T30	6.13	s
Time From	0	s
Time To	0	s
Noise Comp.:	ON	

The EDT (“Early Decay Time”) is obtained from the first 10 dB of level decay, T10 from the level decay between  $-5$  dB and  $-15$  dB, T20 from the level decay between  $-5$  dB and  $-25$  dB and T30 from the level decay between  $-5$  dB and  $-35$  dB.

When looking at the diagram you may perhaps be confused at first sight: If the energy is summed from the end to the beginning, it should also constantly increase from the end to the beginning – it cannot decrease anyway, because the energy is always positive. In the time range greater than 3.5 s, however, this continuous behavior is not seen. The reason for it lies in the *Noise Compensation* which detects the noise in the signal and eliminates it from the signal. This noise component to be deducted is a statistical mean of the noise that actually occurs and sometimes may well be stronger than the measured signal itself. If this is the case, we have to live with such "cancelled" areas in the Schroeder diagram which, however, do not influence the reverberation times obtained – on the contrary: if the noise would not be compensated a measuring error would result, since the noise level would cause the integrated energy to become too large.

As a trial we will now switch off the noise compensation by opening the *Measurement File Properties (F4)* dialog box: We find the corresponding option on the *General Processing* page in the *Schroeder Integral* section. Now the integral really does rise continuously from the end to the beginning:



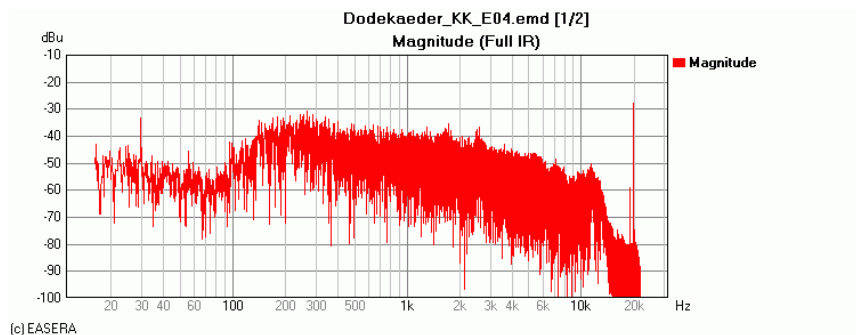


It is quite understandable that now the calculated reverberation times show strong deviations, which in effect are not correct:

	RT	
EDT	4.57 s	
T10	5.56 s	
T20	13.60 s	
T30	16.30 s	
Time From	0 s	
Time To	0 s	
Noise Comp.:	OFF	

## Frequency response

Now let us also look in passing at the frequency response (*Magnitude*). Since we have not carried out any compensation for the loudspeaker, we shall not attach any major importance to it:



Since we have not calibrated the inputs either, there can be no statement given about the level of the measurement.

## Some acoustical values

If we choose the *Arrival*, *C50*, *D/R*, *S/N* item on the *Graphs* navigator page in the *Calculation* section, several basic room acoustical quantities are calculated and displayed:

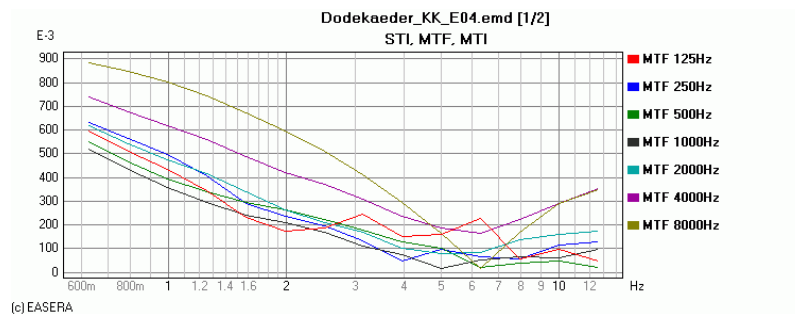
Data		
Arrival	42.83	ms
Distance	14.72	m
C50	-6.1	dB
C80	-4.4	dB
Ltotal	67.1	dB SPL
Center Time	316.93	ms
Noise	8.4	μPa
Abs. Max	5.57	mPa
SNR	20.4	dB
Section		
From	0	s
To	5.9	s
Abs. Max	5.57	mPa
SNR	20.4	dB
D/R Ratio	Markers?	
Split Time	35	ms

Among other things we have here the measure of definition according to Ahnert (*C50*), the measure of clarity according to Abdel Alim (*C80*) and the calculated noise component. The details of these quantities are described in the manual.

## Speech intelligibility

If the speech intelligibility is measured for alarm systems and similar purposes, the sound emission level and the level of the expected ambient noise play a very significant role.

In the following example the point is rather how well one could follow a sermon with the listening congregation being mainly quiet. We choose on the *STI* page in the *Measurement File Properties (F4)* dialog box a signal level of 70 dB and a S/N of 30 dB for all octave bands. *Signal Masking* and *Noise Levels* shall be taken into account by selecting the respective check boxes. Then we select the *STI*, *STIPa*, *RaSTI* item in the *Calculation* section.



The diagram shows the modulation transfer functions: Poor MTF values are the main result when low frequencies are quickly modulated.

These results are numerically presented on the *Details* navigator page:

STI	0.354
AICons [%]	25.097
STI (Male)	0.346
STI (Female)	0.361
RaSTI	0.301
Equiv. STIPa (Male)	0.360
Equiv. STIPa (Female)	0.373

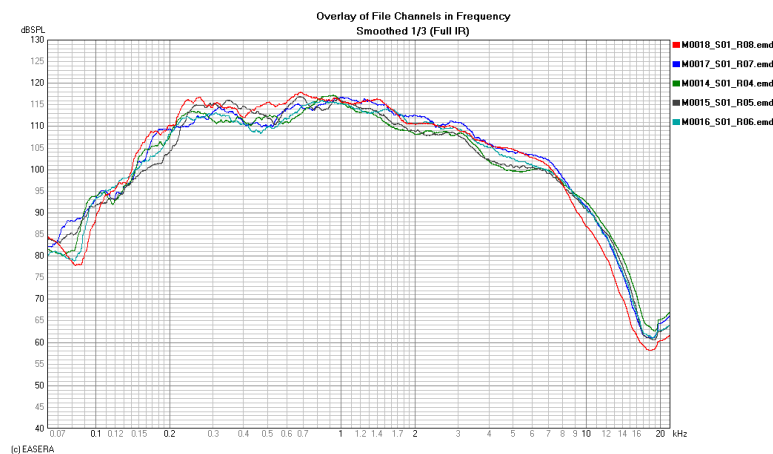
We see a STI value of below 0.4, which confirms that a sermon would not be understandable without a sound reinforcement system.

Since the modulation transfer functions produce poorer results with low frequencies it is not surprising that male voices have slightly poorer speech intelligibility values than female voices.

## The sound reinforcement system

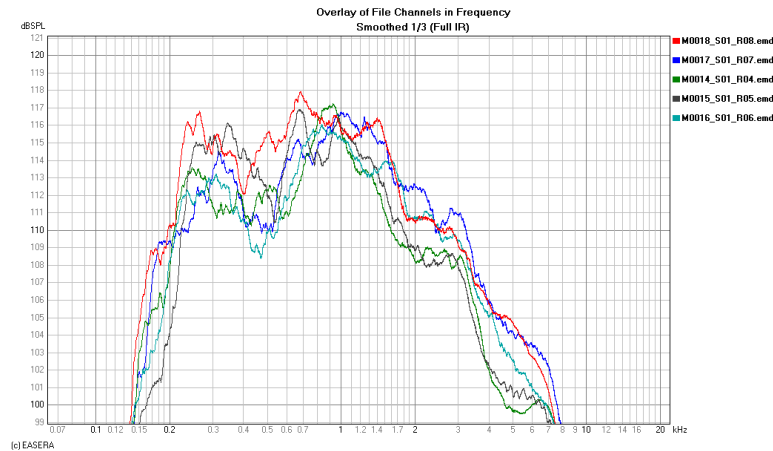
Finally we want to see if the sound reinforcement system needs to be (differently) equalized. In this case it does not make sense to try to obtain an optimum result for just one microphone position – the objective must be, however, to achieve a balanced as possible frequency response in the whole room.

First we will have a look at an overlay showing the frequency response of five positions in the parquet of the church (only sound system in the parquet active – S01 and measurement places R08 = E08 etc.). To do this we choose the first measurement and select the *Smoothed 1/3* item in the *Frequency / More* section on the *Graphs* navigator page. Finally we choose the *Add To Overlay* function and also add the other measurements.



As can be seen, all transfer curves in the considered frequency range are similar.

If the frequency responses look quite smooth here, this is due mainly to the large diagram range of more than 60 dB. After making sure that the mouse function *Zoom X / Zoom Y* is activated, we select a region from 100 dB to 120 dB with the right mouse button:

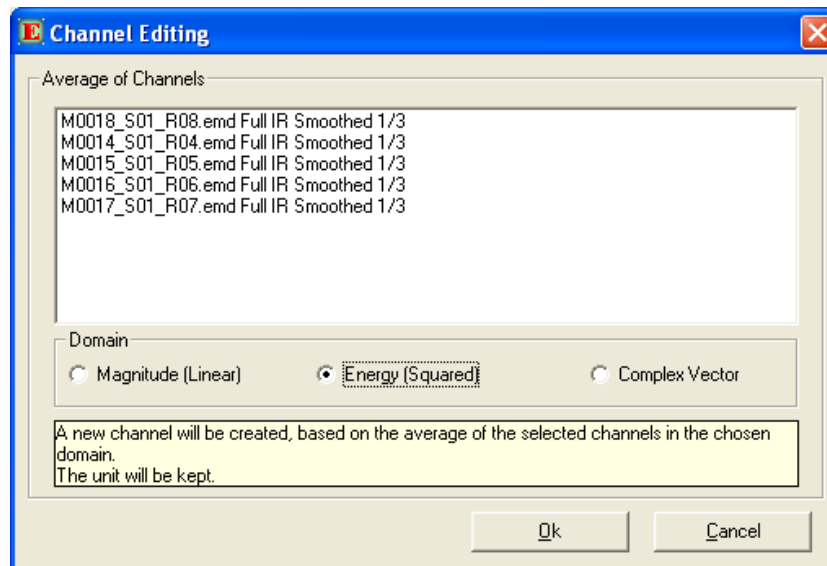


Now we see that the frequency responses are not quite as balanced (also remember that they have already been smoothed to 1/3 octave). Additionally there is a large dip in level in the region between 300 Hz and 600 Hz that should be compensated.

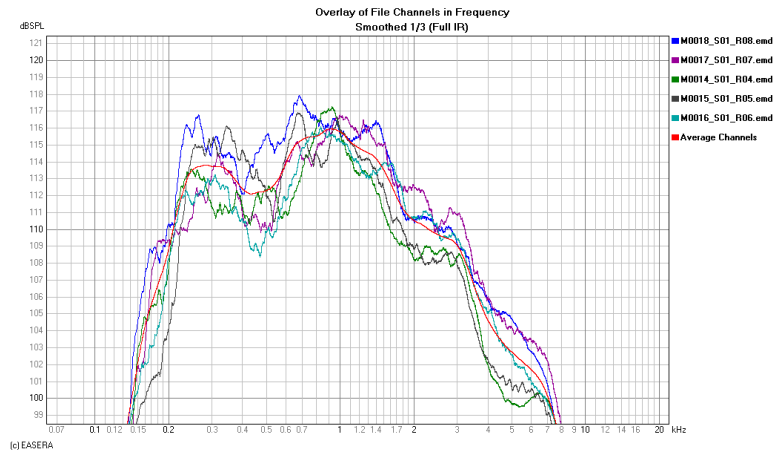
Towards the ends of the main frequency range we see large fall-offs in level. Here one should also abstain from equalizing, since one simply cannot force any loudspeaker by EQ to comply with requirements it simply cannot deliver.

## Averaging curves

For completeness sake we are now going to average the five curves by choosing the *Edit / Average Channels* menu item:



The five curves are to be averaged energetically (for details regarding averaging see lesson 5). We will then include the new curve in our overlay:



As we see, in the range from 300 Hz to 1 kHz we have a fluctuation of only +/- 2 dB. This is certainly too little to absolutely justify the use of an equalizer. As we have just seen, the averaged frequency response is a good clue for judging the resulting variations and for drawing adequate conclusions regarding a possible equalization.

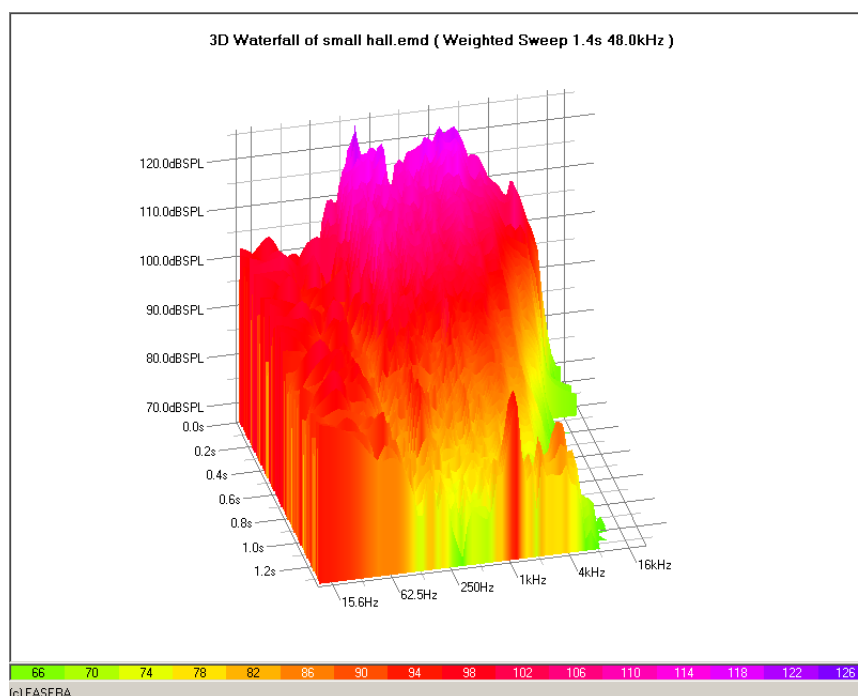
# Lesson 10: Waterfall Diagrams

The *View & Calc* page allows viewing only two-dimensional diagrams that are frequently used for showing the level over time (ETC) or the level over frequency (frequency response). Using a three-dimensional presentation, it is possible to show the level simultaneously over time and frequency. A diagram of this kind often assumes the appearance of a waterfall and that is what it is called.

Waterfall diagrams have the "advantage" of looking spectacular and, if colors are assigned, rather colorful. Drawing useful and acceptable conclusions from them, however, requires some experience.

## Creating a waterfall diagram

For these investigations we are going to use the file *small hall.emd* of the Brandenburg Academy of Arts in Berlin. In the *View & Calc* page we click on *Calculation | Waterfall* in the *Graphs* navigator page (second item from bottom). EASERA changes to a new page (*Waterfall*) and we see a diagram which, depending on the settings made, looks more or less as follows:



A waterfall diagram is a three-dimensional diagram in which the level is represented as a function of time and (!) frequency. Thus one gets the time behavior (mostly the decay behavior) of the system for each and every frequency or, if you want to see it from another point of view, the development of the frequency response over time. In this respect one has to consider, however, that it is not possible to resolve time and frequency simultaneously to the same degree of fineness. A high time resolution always permits only coarse frequency resolution and vice versa:  $\Delta t \sim 1/\Delta f$ . (This is the uncertainty relation of acoustics.) The level is indicated not only by a corresponding position in the diagram, but also by the color.

## Positioning of the diagram

Below the diagram there is a scroll bar used to turn the diagram horizontally. On the *View* navigator page there is the field *Hor Angle [°]*, in which it is also possible to preset the turning angle numerically.

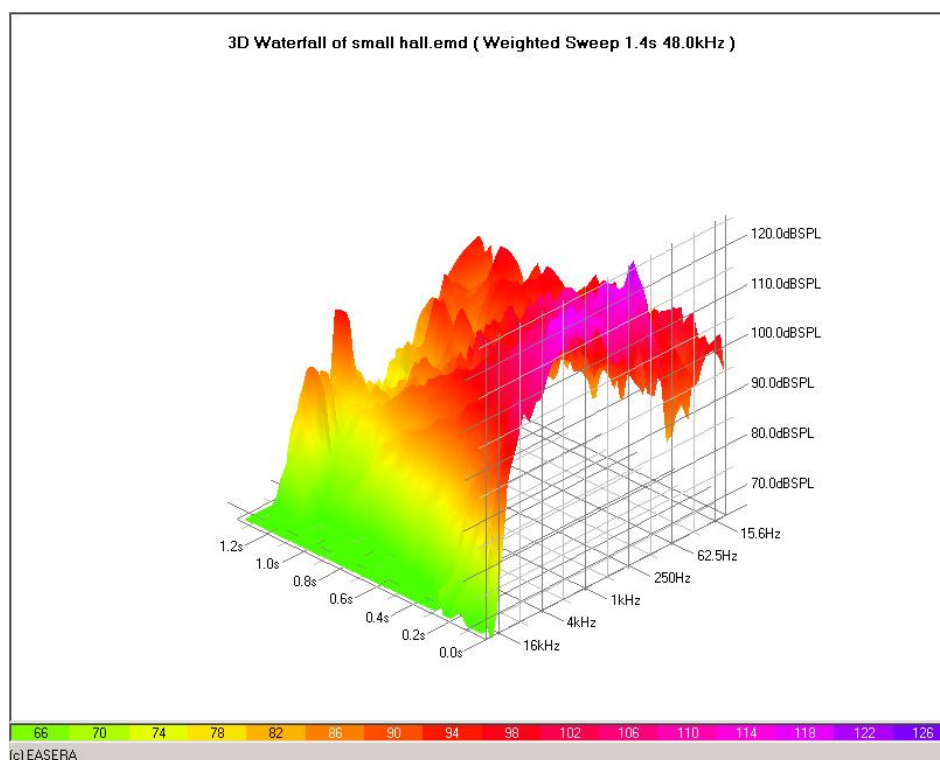
To turn the diagram vertically there is a scroll bar on the right of the diagram. In the field *Ver Angle* [°] it is also possible to preset the turning angle of the window.

The diagram can be enlarged or reduced, using either the scroll bar at the left of the diagram or by entering the zoom factor in the *Zoom* [%] field.

A particularly nice animation is hidden beneath the *View / Spin* menu item: It enables the diagram to be put into a permanent spin.

## Level peaks at the end of the diagram

If we position the diagrams appropriately (by setting *Hor Angle* [°] to 222, *Ver Angle* [°] to 16, *Zoom* [%] to 136), we detect level peaks at the end of the diagram:



One could now be tempted to believe having detected room reflections at certain frequencies. The origin is different, however: Due to the use of sweep excitation signals the distortion products (harmonics) concentrate in front of the proper impulse from where they are transferred by the cyclical movement to the end of the impulse response.

Thus we do not have room reflections, we rather have distortion products.

## Two-dimensional view

Using the *View/Side View*, *View/End View* and *View/Plan View* menu items it is possible to turn and swivel the diagram in such a way that each time a two-dimensional view is produced. Please note that the arrangement of the scales may not necessarily be as expected. With *Side View*, for instance, it may be that the low frequencies are on the right and the high frequencies on the left. This may, however, be inverted by selecting the function again.

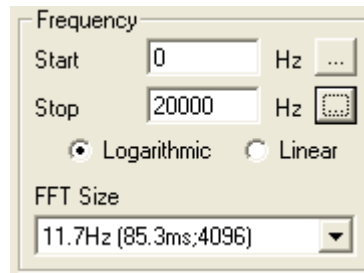


## Fine adjustment of the presentation

We have already seen how the diagram can be swiveled and turned, but, EASERA provides a lot of further adjustment possibilities:

### Frequency range and smoothing

On the *Axes* navigator page in the *Frequency* section it is possible to adjust the start and stop frequencies.



These frequencies may be entered numerically or by using the buttons on the right side of the data fields. These buttons will open a selection list where it is possible to look up and choose the ISO-frequencies.

Below the frequency data field one can determine whether the frequency scale shall be linear or logarithmic and finally it is still possible to set the FFT size. With the sample frequency of 48 kHz used in this example and an FFT size of 4096 points one obtains a frequency resolution of 11.7 Hz. This is certainly sufficient for inspecting the whole spectrum, but for assessments in the sub-bass range it will be necessary to have access to higher FFT resolutions.



To avoid making the representation inaccessible by too fine details, one should adequately smooth in the frequency range. The corresponding menu items are available in the *Graphs* menu (and, like all other menu items, also on the button bar), alternatively to this you find on the *View* navigator page the selection list for *Average*.

### Time range

On the *Axes* navigator page it is possible in the *Time* section to set the start and stop points of the display. Use *Steps* to specify the resolution of the display in the time domain and *Window* to specify the type of window to be used for the FFT analysis (Further details regarding the window types are given at the end of Lesson 6).

Time

Start  s

Stop  s

Steps  ▾

Window  ▾

## Level range

On the *Axes* navigator page one can determine in the *Magnitude* section what level range to use for the display and if a linear or a logarithmic scaling is to be used.

Magnitude

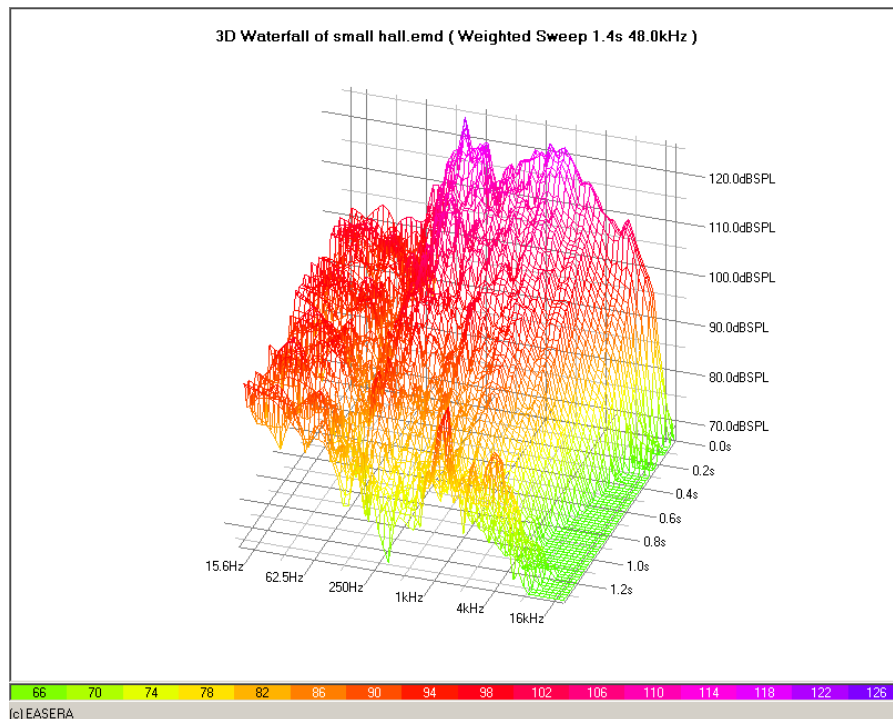
Max  dB SPL

Min  dB SPL

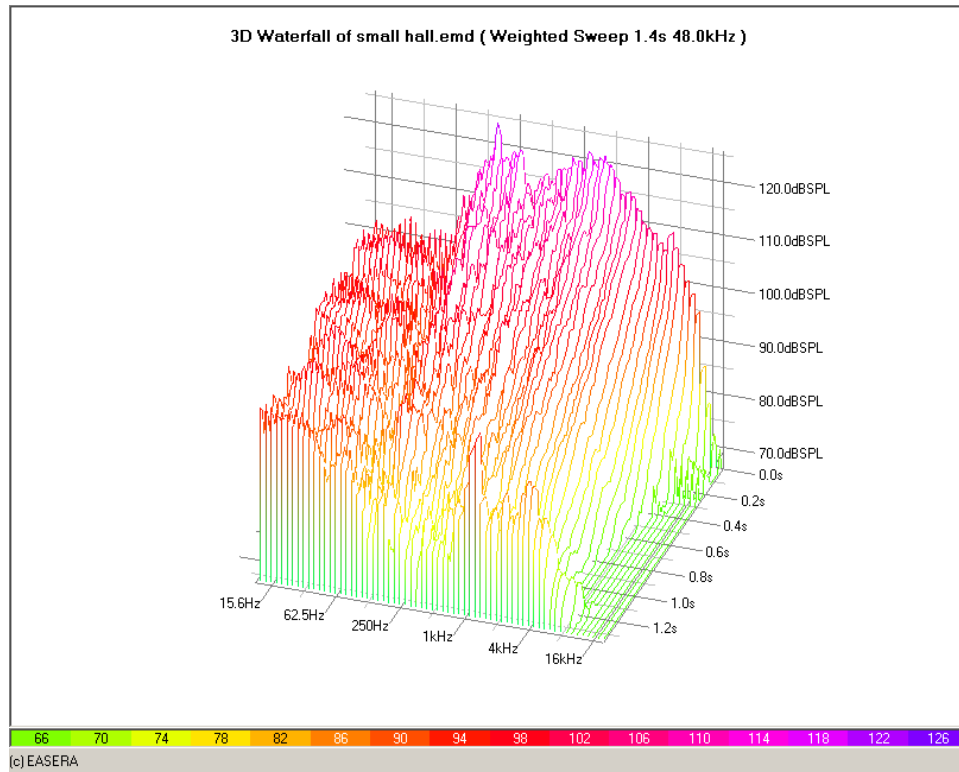
Logarithmic  Linear

## Slices or grid

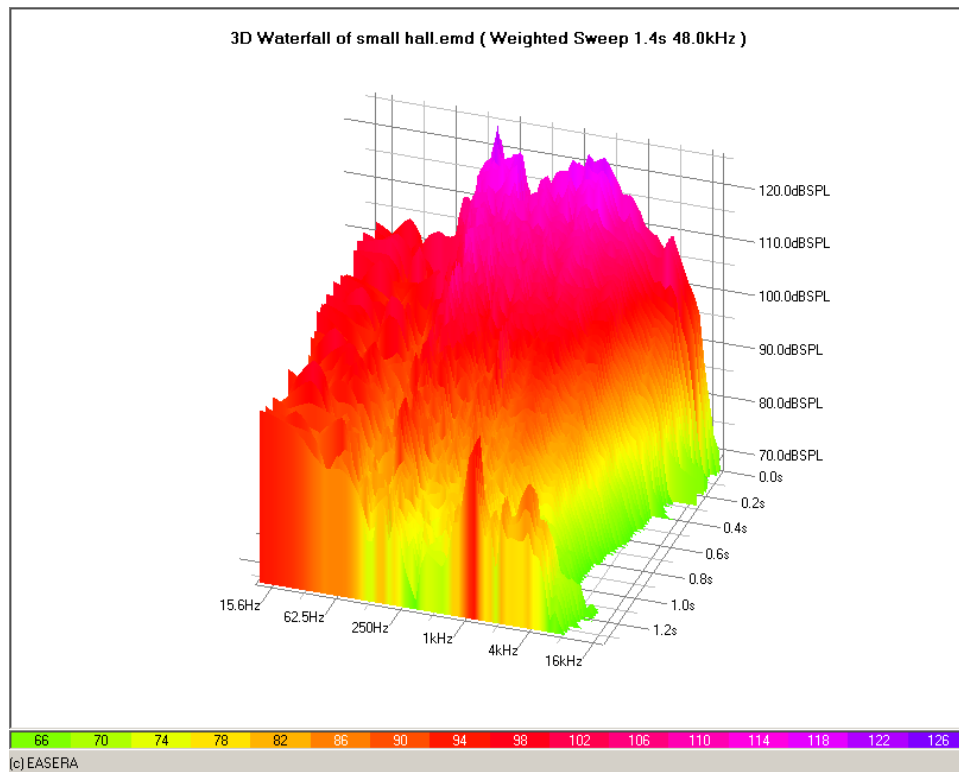
To show the diagram as a grid select the *View / Grid* menu item (the diagram below shows an unfilled grid):



The diagram can also be shown in the form of slices for the individual frequencies by selecting the *View / Frequency Slices* menu item (in the diagram below the slices are unfilled):



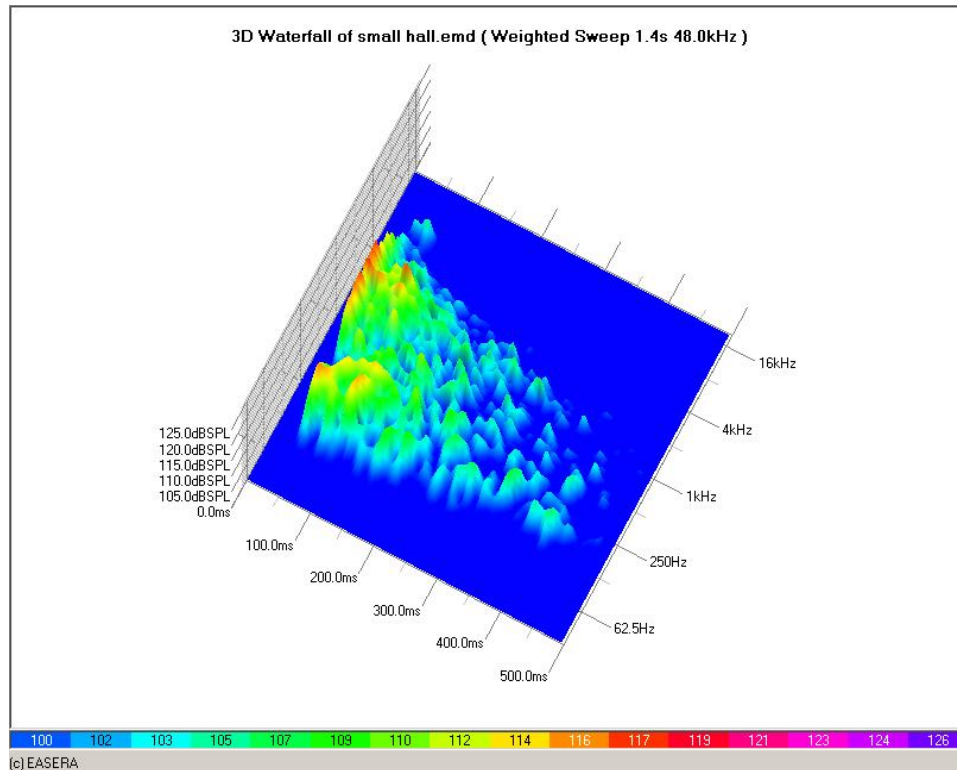
Alternatively to this slices can be shown for individual times by selecting *View / Time Slices* (in the diagram below the slices are filled):



The *View/Fill* menu item enables switching between the filled and unfilled representation. All of these settings can also be applied using the *View* navigator page.

## Detail examinations with waterfall display

It is often desirable to display clearly the frequency response of the direct sound and individual strong reflections. This is easily possible by choosing the proper parameters.



This presentation allows showing the details as in the waterfall display above. To show this the following parameters are set:

The screenshot shows a settings dialog box with the following parameters:

- Time:** Start: 0 s, Stop: 0.5 s, Steps: 5.3ms (256), Window: Blackman
- Frequency:** Start: 0 Hz, Stop: 24000 Hz,  Logarithmic,  Linear, FFT Size: 23.4Hz (42.7ms;2048)
- Magnitude:** Max: 126 dB SPL, Min: 66 dB SPL,  Logarithmic,  Linear

Buttons at the bottom: Default, Apply

One recognizes the zoomed level range, the high resolution in the time domain (small steps) and the adapted frequency resolution, the value of which can slightly vary depending on the actual measurement (actual FFT size 23.4 Hz). The coloring was also adapted appropriately using the *Legend Colors* section on the GENERAL | COLORS page in the *Options (F9)* dialog box.

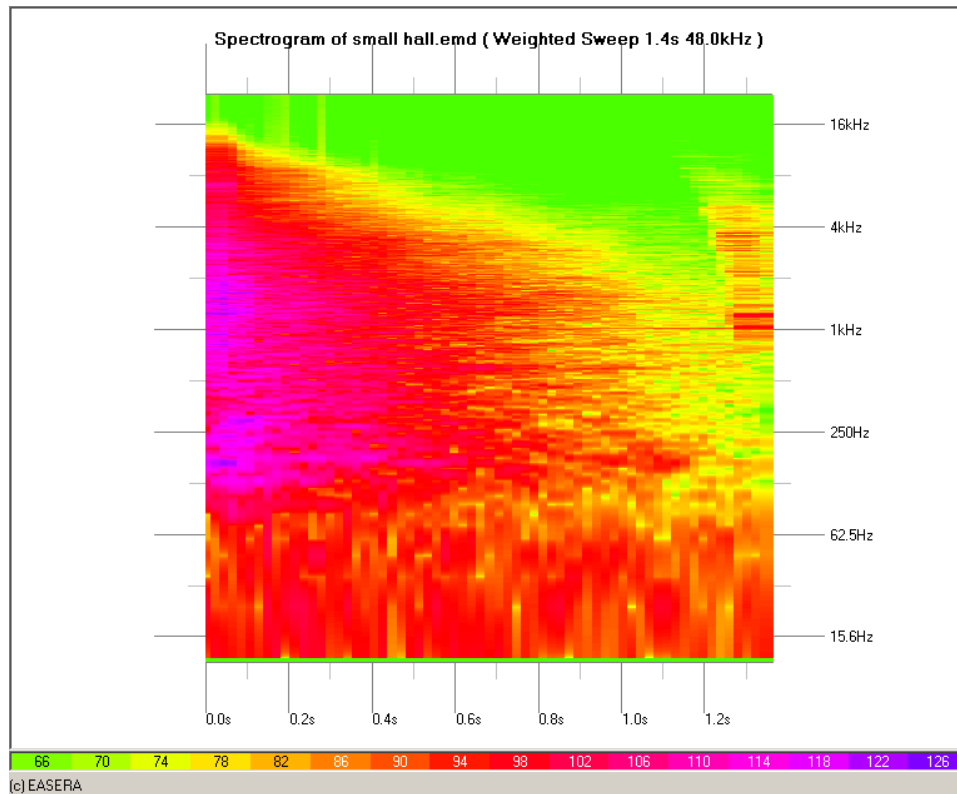
Using this display one can clearly recognize that higher levels are given at 1 to 2 kHz in the range of the direct sound and the first reflections. Between 100 and 250 Hz there are also visible higher-level short-time reflections.

Such evidence is very welcome with room acoustical examinations. As a rule there will be other resolution settings necessary for electro acoustical examinations, since the interesting events take place in another frequency and time frame. In order to accurately resolve e.g. the resonance of loudspeakers, one will in this case typically choose rather long FFT blocks. On the other hand the time behavior of a loudspeaker is simple as compared with room responses, since it is usually no more than a (frequency-dependent) decay process. If the *Steps* are sufficiently small it is possible to follow adequately without having to reduce the FFT block size.

With all variants one must, however, always keep in mind that the level values (direct sound, reflections) may be displaced in time by a constant value in the waterfall display. This is due to the method of successive FFT blocks. The displacement can amount to half the length of an FFT block.

## Spectrogram

The diagram can be shown not only as a waterfall diagram, but also in the form of a spectrogram. A spectrogram is a waterfall diagram viewed from above: The level no longer can be seen by its height in the diagram, but only by the color. But on the other hand the “hills” no longer obscure any “valleys”, so that fine details can be much more easily discerned.



One can clearly see here the faster decay in the treble range (the green wedge-like area on the right side), which is due to the damping behavior of the wall materials at high frequencies and to the air damping.

# Lesson 11: Speech intelligibility

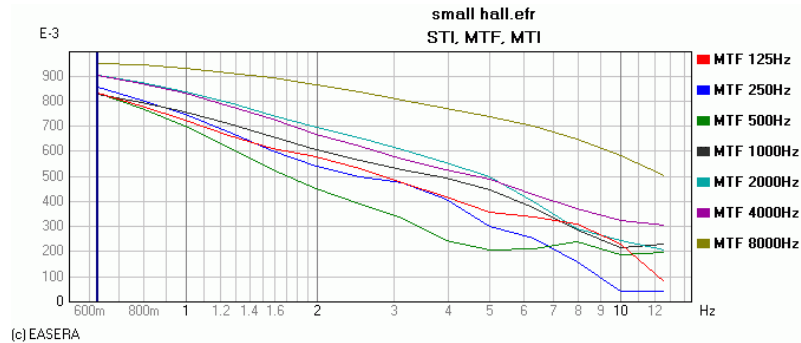
Speech intelligibility plays an important role for assessing the acoustical properties of rooms. EASERA offers ample functionality in this respect.

## The modulation transfer functions

The most common measure for speech transmission is the STI, or *Speech Transmission Index*. The principle used for determining the STI stems from the era of analog measuring instruments: To calculate the STI the signals in the frequency bands of 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz and 8 kHz were modulated with frequencies between 0.63 Hz and 12.5 Hz (The signal intensity fluctuates between 0% and 100%). A modulation loss may occur due to reverberation, echoes or noise in the system under test (the signal intensity at the reception point then fluctuates between x% and 100%, because signals also occur during the “breaks”). The ratio between the modulation factor at the listener seat and the modulation factor of the stimulating signal then results in the modulation transfer function.

In EASERA, however, all modulation factors are derived from the impulse response according to SCHROEDER. There no longer are 98 measuring operations necessary – as is the case with the analog procedure – but only one. Both procedures (the analog procedure, as well as the derivation from the impulse response) are described in the IEC 60268-16 (2003) standard where the individual quantities are also defined.

To determine the STI from one measurement click in the *Graphs* navigator page on the item *Calculation / STI, STIPa, RaSTI*. The modulation transfer functions are then shown in a diagram:



Modulation transfer factors are restricted to between 0 and 1 (corresponding to 0% and 100%), the Y-axis is scaled here in 1/1000th. To cover different octave bands the corresponding curves are combined per overlay. The modulation frequency is found on the X-axis.

In the navigator we switch to the *Details* page for numerical display of the calculated values:

STI (Mask.)	0.550
AICons [%]	8.661
STI (Male)	0.559
STI (Female)	0.567
RaSTI	0.506
<hr/>	
Equiv. STIPa (Male)	0.562
Equiv. STIPa (Female)	0.573

The following quantities are shown there in detail:



- **STI:** *Speech Transmission Index*, result value of the modulation transfer functions of all frequencies and modulation frequencies. The suffix (*Mask*) will be explained later.
- **AlCons (%)**: Articulation loss of consonants to be expected, calculated from the *Speech Transmission Index* (STI) (according to Farrel-Becker).
- **STI (Male) and STI (Female)**: Result value of the modulation transfer functions of all frequencies and modulation frequencies, in which the individual values are weighted according to IEC 60268-16 (2003) for the male voice (*Male*) and the female voice (*Female*).
- **RaSTI:** With the RaSTI procedure the octave bands of 500 Hz and 2 kHz are used exclusively. The 500Hz band is modulated with the frequencies 1 Hz, 2 Hz, 4 Hz and 8 Hz, and the 2 kHz band with the frequencies 0.7 Hz, 1.4 Hz, 2.8 Hz, 5.6 Hz and 11.2 Hz. Thanks to the reduced number of octave bands and modulation frequencies, the time for a discrete measurement is reduced considerably. Since all values in EASERA are derived from the impulse response, there is no advantage by comparison with the STI. Due to its limited conclusiveness for only two frequency bands, the use of RaSTI is continually diminishing.
- **Equiv. STIPa (Male) and Equiv. STIPa (Female)**: With the STIPa procedure only 2 instead of 14 modulation frequencies are used for each of the seven octave bands, therefore the time for discrete measurements gets reduced accordingly (Since all values in EASERA are derived from the impulse response, there is no advantage by comparison with the STI). But as the procedure requires external signal stimulation, the STPa values obtained via the impulse response are given only as equivalence values. With *STIPa (Male)* the weighting factors for the male voice are used, with *STIPa (Female)* those for the female voice.

## The register page Results

For some evaluations EASERA opens the *Results* page on which further data can be found – as shown here.

small hall.emd : Measures						
	MTF 125Hz	MTF 250Hz	MTF 500Hz	MTF 1000Hz	MTF 2000Hz	MTF 4000Hz
0.63 Hz	0.832	0.896	0.834	0.885	0.903	0.90
0.8 Hz	0.779	0.805	0.773	0.850	0.873	0.87
1 Hz	0.724	0.747	0.701	0.806	0.838	0.83
1.25 Hz	0.666	0.679	0.618	0.756	0.796	0.78
1.6 Hz	0.612	0.601	0.524	0.702	0.746	0.72
2 Hz	0.576	0.540	0.450	0.648	0.697	0.66
2.5 Hz	0.532	0.498	0.392	0.604	0.652	0.61
3.15 Hz	0.476	0.475	0.332	0.563	0.606	0.57
4 Hz	0.418	0.411	0.242	0.526	0.556	0.52
5 Hz	0.399	0.301	0.208	0.477	0.497	0.48
6.3 Hz	0.337	0.252	0.209	0.403	0.402	0.42
8 Hz	0.310	0.151	0.241	0.305	0.292	0.37
10 Hz	0.229	0.040	0.186	0.232	0.245	0.32
12.5 Hz	0.080	0.041	0.196	0.246	0.206	0.30
MTI	0.492	0.451	0.449	0.951	0.968	0.57
STI (Mask.)	0.590					
AlCons (%)	0.661					
STI (Male)	0.559					
STI (Female)	0.567					
RaSTI	0.506					
Equiv. STIPa (Male)	0.562					
Equiv. STIPa (Female)	0.573					
STI (Modified)	0.540					
STI (Unweighted)	0.545					
STI (Custom)	0.567					
RaSTI (weighted)	0.513					
STIPa (Modified)	0.562					
STIPa (Unweighted)	0.564					

All 98 modulation transfer function values (MTF) as well as the modulation transfer indices (MTI), are numerically shown here.

On the left side there is a list in which the actual result can be stored. To do this click in the last line (*[Empty]*) on the column *Add/Set*. To overwrite a data record in the memory table click on the same column, but in the corresponding row.

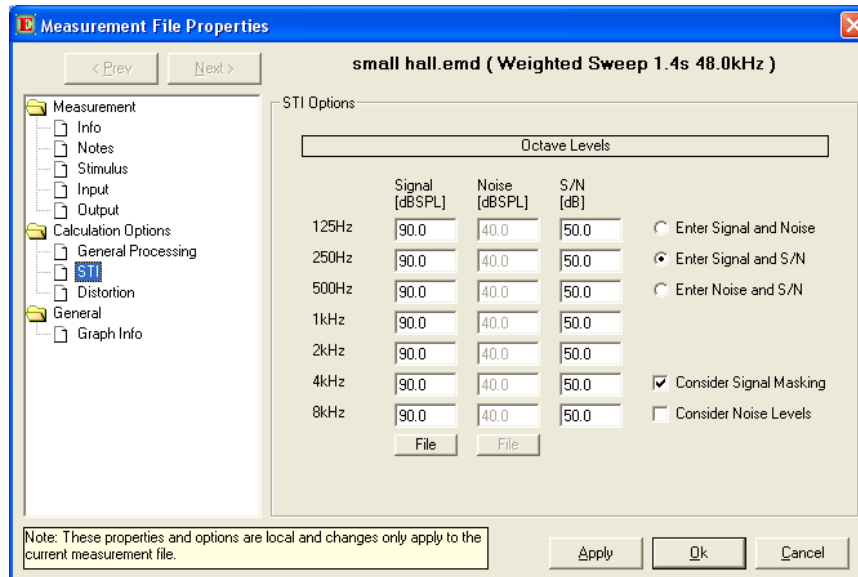
V	Table	Add/Set
	STI, MTF, MTI	<< <<
	STI, MTF, MTI	<< <<
X	STI, MTF, MTI	<< <<
	[ Empty ]	<< <<

Use *File / Set Table To / Clipboard* to copy the table into the clipboard, and use *File / Set Table To / File* to store the table in a file; in both cases the different columns are separated by tabs.

## Considering ambient noise

Ambient noise impairs speech intelligibility. But the extent to which it does so depends not only on its level, but also on its spectral composition and its relation to the desired signal.

To enable speech intelligibility to also be determined with due consideration of ambient noise, EASERA allows S/N values to be entered. Use the CALCULATION OPTIONS | STI page in the *Measurement File Properties (F4)* dialog box. (The settings in the window *Measurement File Properties (F4)* are valid only for the current file. Should pre-settings be made for all future measurements, these are to be defined in the *Options (F9)* dialog box on the PROCESSING | STI OPTIONS page.)



Here one can freely choose which combination of quantities to enter. In practice one would normally first measure the desired signal (e.g. using the Live module of EASERA) and then the disturbing noise. On the right side one would then choose *Enter Signal and Noise* and thereafter enter the values of the different octave bands in the columns *Signal [dBSPL]* (desired signal) and *Noise [dBSPL]* (disturbing noise). For the octave spectra *Signal* and *Noise* it is also possible using the *File* button to load octave spectra in the \*.els format. These files can be created with the Live module of EASERA.

Finally, if the option *Consider Noise Levels* is enabled, EASERA then calculates the STI with consideration of the ambient noise.

STI +N (Mask.)	0.534
AICons [%]	9.426
STI (Male)	0.543
STI (Female)	0.551
RaSTI	0.493
Equiv. STIPa (Male)	0.547
Equiv. STIPa (Female)	0.557

The details now show *STI +N* instead of *STI* (for the example shown the S/N was set to 15 dB). Please keep in mind that the weighted signal-to-noise ratio should be more than 10 dB to not deteriorate the speech intelligibility significantly.

## Considering masking effects

Masking effects are taken into account when the option *Consider Signal Masking* is enabled. Thus the masking of low-level signal components by high-level components of adjacent frequencies is duly considered.

If such consideration is not desired, this option has to be disabled, in which case (*No Mask*) appears in the line *STI*.

## Custom frequency weighting

In the calculation of the STI not all octave bands are weighted in the same way – this is the case even with the standard STI, as well as with the *Male* and *Female* options.

In order to also enable weightings here, although not yet standardized, EASERA offers the option of entering one's own frequency weightings, using the *Options (F9)* dialog box PROCESSING | CUSTOM STI page.

Custom STI Parameters

Octave Weight		Redundance Weight	
125	0.130	125	0.000
250	0.140	250	0.000
500	0.110	500	0.000
1000	0.120	1000	0.000
2000	0.190	2000	0.000
4000	0.170	4000	0.000
8000	0.140		

The weightings of the individual octave bands can be entered in the *Custom STI Parameters* section, and in *Redundance Weight* the redundancy correction factors can be entered which weight the influence of adjacent octave bands (for details see IEC 60268-16 (2003) A.2.3). Use the *Standard*, *Male* and *Female* buttons to pre-

fill the different fields with the values taken from the standard (in the present example the setting would be *Standard*).

To get the values calculated with these weighting factors displayed at all it is necessary to activate the option *Show Non-Standard STI Values*. This option is available in the *Options (F9)* dialog box on the PROCESSING | STI OPTIONS page.

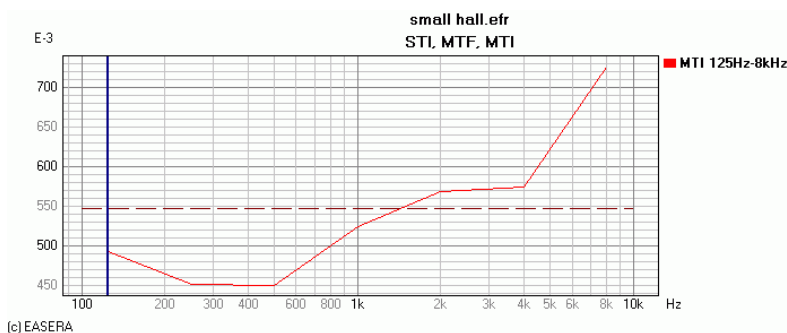
STI +N (Mask.)	0.534
AICons [%]	9.426
STI (Male)	0.543
STI (Female)	0.551
RaSTI	0.493
Equiv. STIPa (Male)	0.547
Equiv. STIPa (Female)	0.557
STI (Modified)	0.526
STI (Unweighted)	0.529
STI (Custom)	0.551
RaSTI (Weighted)	0.500
STIPa (Modified)	0.547
STIPa (Unweighted)	0.549

Also on the STI OPTIONS page it is possible to access the same settings as in the *Measurement File Properties (F4)* dialog box (*Signal, Noise, S/N...*). The settings made there serve as presets for new measurements.

## The modulation transfer index (MTI)

The modulation transfer indices are calculated from the impulse response in the individual octave bands between 125 Hz and 8 kHz, the modulating frequencies are located between 0.63 Hz and 12.5 Hz. The procedure is accurately described in IEC 60268-16 (2003). From the individual MTI's the STI is derived as a weighted average.

To show the MTI curves choose the *Calculation / MTI and STI* item on the *Graphs* navigator page.



For comparison the STI is shown as a dashed line.

In this diagram one can easily discern the relative transmission quality of the “octave channels”.

# **Lesson 12: Explanation of further measuring quantities**

We have already dealt with a part of the measuring quantities calculated by EASERA and the diagrams shown by EASERA. Now we are going to have a look at the remaining quantities.

## Diagrams of the time domain

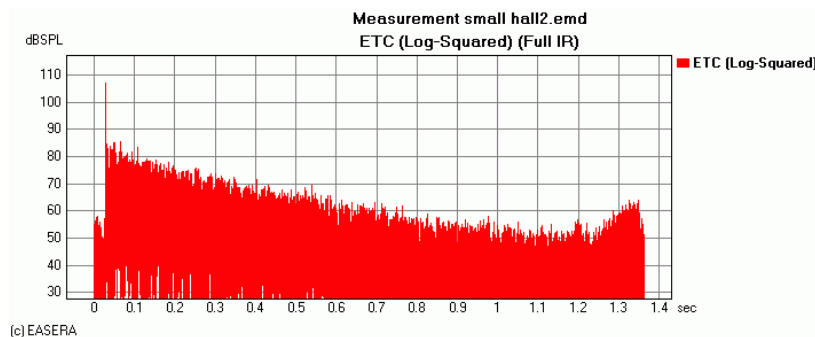
The time domain diagrams are shown on the *Graphs* navigator page, all in the *Time* folder. These diagrams can be shown for the unfiltered and unwindowed data (Full IR), for individual one third octave or octave bands as well as for windowed and postprocessed data.

### Impulse response

The impulse response has already been extensively described in Chapter 4.

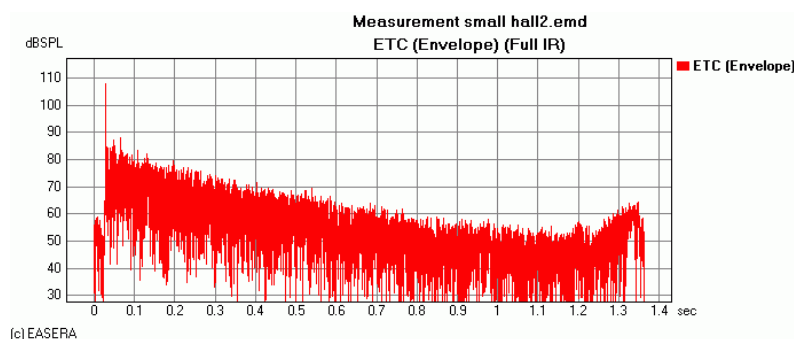
### ETC (log-squared and envelope)

The Energy Time Curve (ETC) shows the energy decay over time. Since the sound power is calculated from the square of the sound pressure, the impulse-response values are squared and shown as the log-squared ETC. The result is shown on a logarithmic scale:



The square of a value is always positive – negative energy values just do not exist in classical physics. At those points, however, at which the impulse response crosses the zero line, the square is also zero. Thus the energy drops to zero with corresponding frequency in the log-squared ETC. Though being physically correct, this is not necessarily helpful for evaluation.

This is why EASERA also offers the envelope ETC diagram: In this case the analytical function is formed from the impulse response with the help of a Hilbert Transformation (complex extension in the time domain). Then the energy is calculated from this function and finally displayed as an envelope, i.e. as a smoothed and squared ETC:



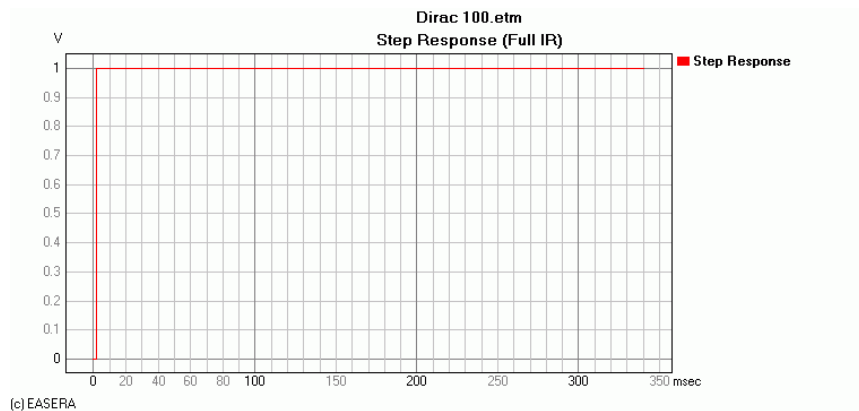
These energy-time diagrams are primarily used for localizing conspicuous reflections (mainly in rooms, but also in other media). If the energy rises a bit at the end of the diagram, it is normally due to the distortion components being settled there by the sweep during measurements – so this has nothing to do with a reflection. (It would be better to perform the measurement as disturbance-free as possible or to window at the end.)

## Schroeder Integral

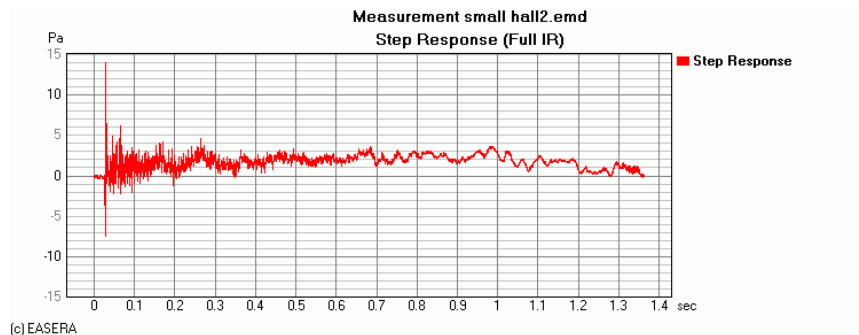
The Schroeder Integral has already been extensively dealt with in Chapter 9.

## Step Response

The step response is the reaction of the room to a step function. The step function would look as follows:

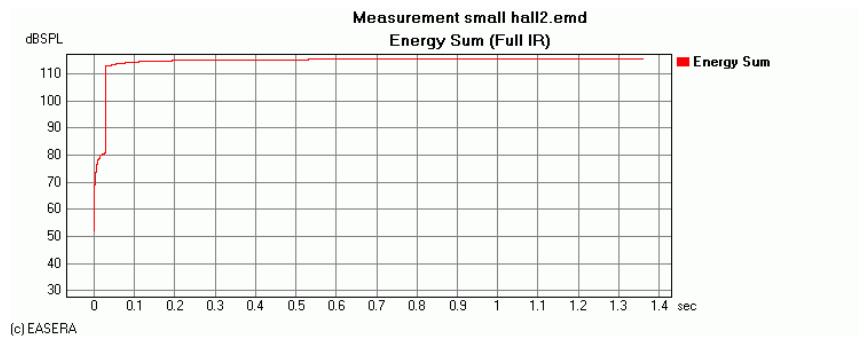


The step response of a room acoustical measurement could look as follows:



The step response is rarely used for room acoustical measurements, but very often for showing the transient and decay response of loudspeakers. The corresponding presentations are correlated with quality features of loudspeakers.

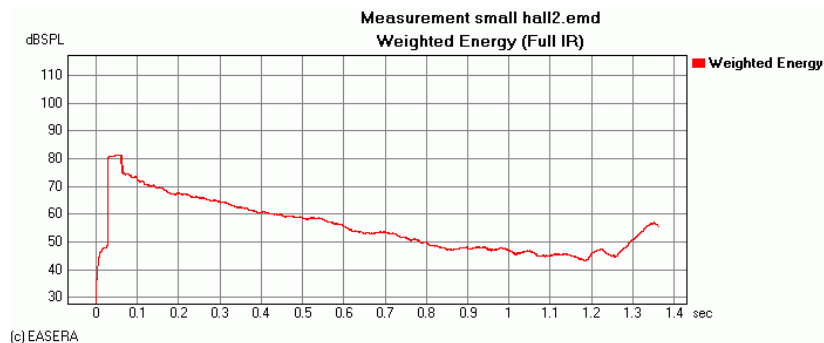
## Energy Sum



The summed energy is the forward integral of the squared signal, displayed on a logarithmic scale. Normally one finds a fast rise in these diagrams at the moment of arrival of the main impulse of the impulse response and then a very quickly flattening trace. At those points where the distortion components are located in measurements with sweeps, one may notice a slight rise.

In room acoustical assessment, this behavior occurring during the first 200ms is used for characterizing transient response of rooms. If the behavior is as shown in the diagram above, one speaks of a fast sound build-up; the room in question is very suitable for speech transmission. A slow sound build-up is characterized by a constant slope of the summed-up energy. Rooms of this kind are preferred for musical performances.

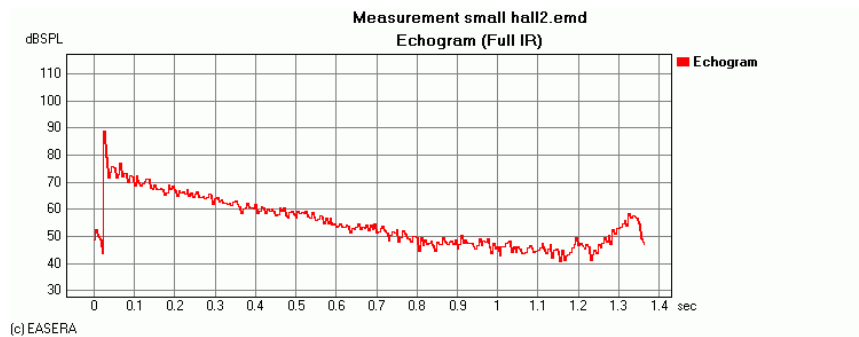
## Weighted Energy



The weighted energy is the short-term integral of the energy over the sliding time window  $\Delta t$ . The integration period  $\Delta t$  can be set in the *Measurement File Properties (F4)* dialog box in the *Weighted Energy* section on the CALCULATION OPTIONS|GENERAL PROCESSING page. The default value is 35 ms. This 35 ms corresponds to the inertia of the ear. Using this procedure, it is possible to conveniently analyze how our ears resolve acoustical structures over time.

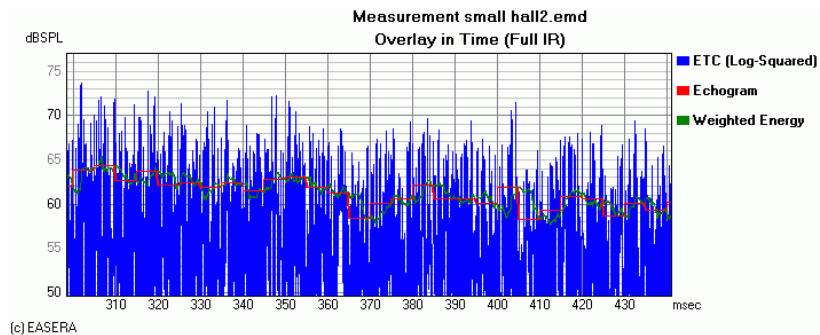


## Echogram



To create an echogram, the composite partial energy is ascertained for periods of time  $\Delta t$ . The integration period  $\Delta t$  may be set in the *Measurement File Properties (F4)* dialog box in the *Echogram* section on the CALCULATION OPTIONS|GENERAL PROCESSING page. The default setting is 5 ms.

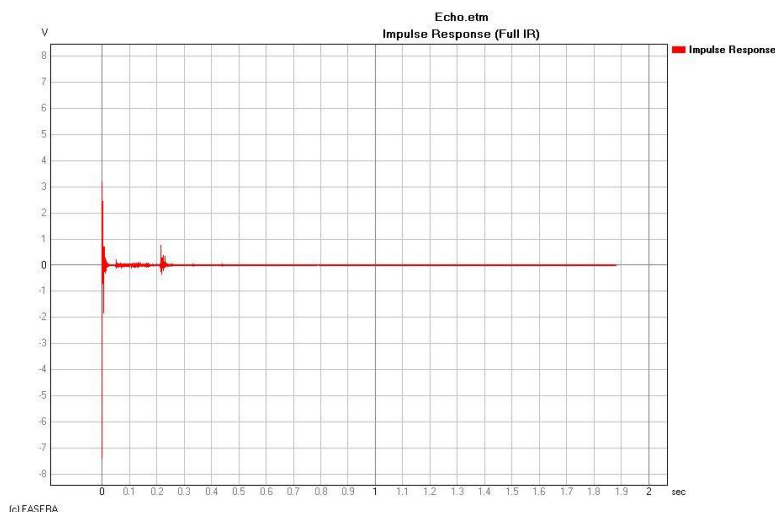
In contrast to *Weighted Energy*, calculation and representation do not slide with this function, but are summed in steps (in the diagram below, the two integration periods have been set to equal values of 5 ms).



Both of these energy representations may be considered the same way as the ETC envelope, as an envelope of the fast oscillating impulse response and can be used for analyzing the decay process of the inherent energy.

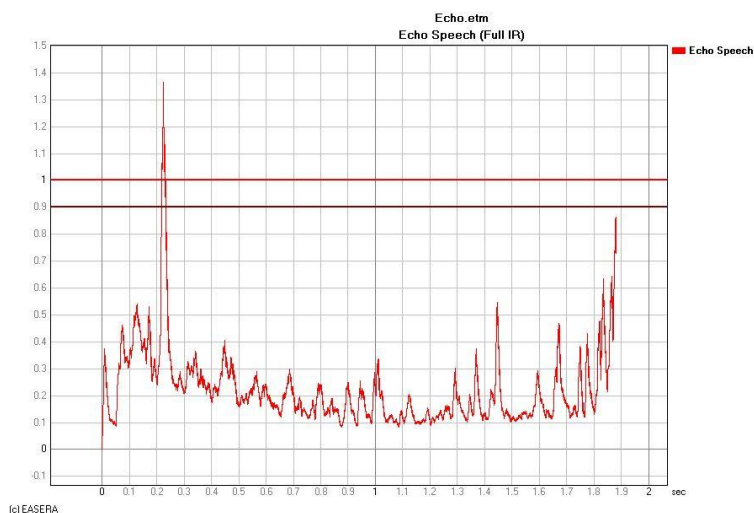
## Echo (Speech and Music)

The next diagram shows an impulse response measured in a theatre with a heavily delayed reflection, which is perceived as an echo by most of the listeners.



In EASERA the *Echo Speech* function allows confirmation of this circumstance of double hearing in an objective way. The echo criterion for speech according to Dietsch and Kraak is produced by selecting this function. If the trace of the curve exceeds the value 1 (for a trained ear a value of 0.9), these reflections are perceived as an (disturbing) echo.

If the measurement is carried out in the presence of major disturbing noise, sometimes heavily delayed artifacts appear in the diagram, which do not have anything to do with real existing echoes.



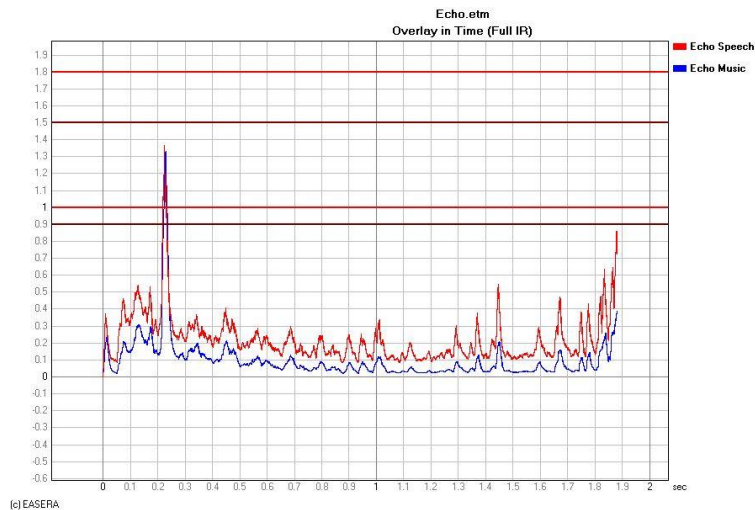
The generally perceived echo is thus objectively confirmed and should be heard by any listener and not only by trained listeners.

Select the *Echo Music* function to produce the echo criterion for music according to Dietsch. If the trace of the curve exceeds the value 1.8 (for a trained ear a value of 1.5) these reflections are perceived as an (disturbing) echo, if the music is of accordingly critical nature.

The value of 1.8 applies to “classical” music (e.g. Mozart), with “dense” music (e.g. Wagner) the value is much higher.

The overlay of the two curves shows that their behavior is similar in quality, but not identical. For speech reproduction, more severe limit values definitely apply. The peak at 1.85 s is in this case is most likely caused by distortion components and would, of course, not be perceived. Keep in mind that the difference between

*Echo Speech* and *Echo Music* is not only due to differing limits of perceptibility, but also to different weightings applied in the calculation.

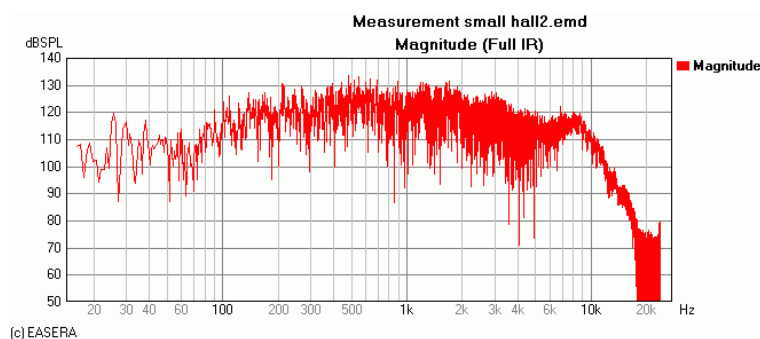


## Diagrams of the frequency domain

The frequency domain diagrams are found on the *Graphs* navigator page, all in the *Frequency* folder. These diagrams can be shown for the unfiltered and unwindowed data (Full IR), for individual one third octave or octave bands (which is normally not very sensible), as well as for windowed and processed data.

### Magnitude

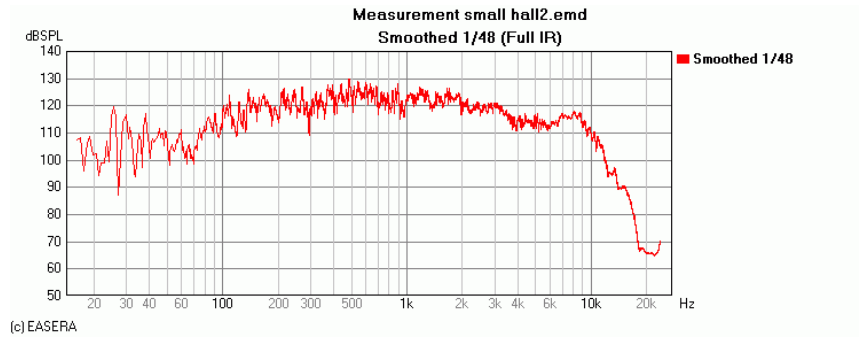
Select the *Magnitude* function to show the unsmoothed frequency response. The resulting spectrum of an FFT analysis is always evenly and linearly distributed over the frequency range, whereas the frequency response is always shown on a logarithmically scaled diagram. This implies that with low frequencies there are fewer points per octave shown than with high frequencies.



### Smoothed

Select one of the *Smoothed* functions to show the smoothed frequency response. EASERA allows smoothing from one octave (*Smoothed 1/1*), through one third octave (*Smoothed 1/3*), a great second (*Smoothed 1/6*), a small second (*Smoothed 1/12*), down to one 96<sup>th</sup> octave (*Smoothed 1/96*).

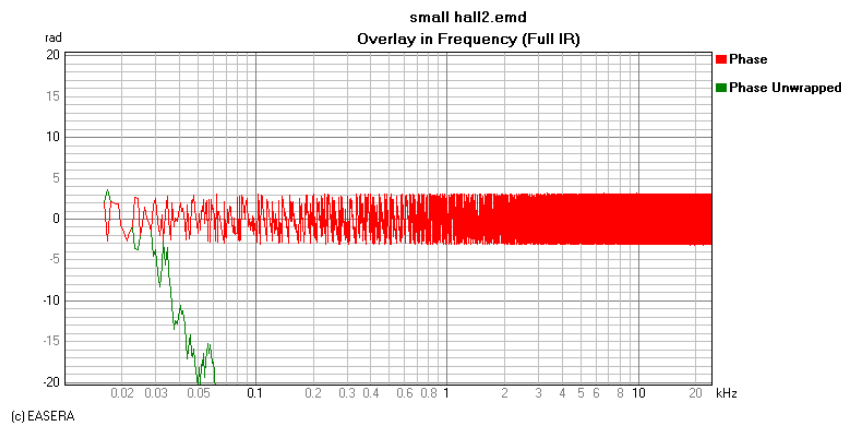
To show the frequency response, smoothing to one 48<sup>th</sup> octave is normally suitable (*Smoothed 1/48*), but often the one third octave smoothing is used, since small dips in the curve which one would probably not hear anyway, tend to disappear.



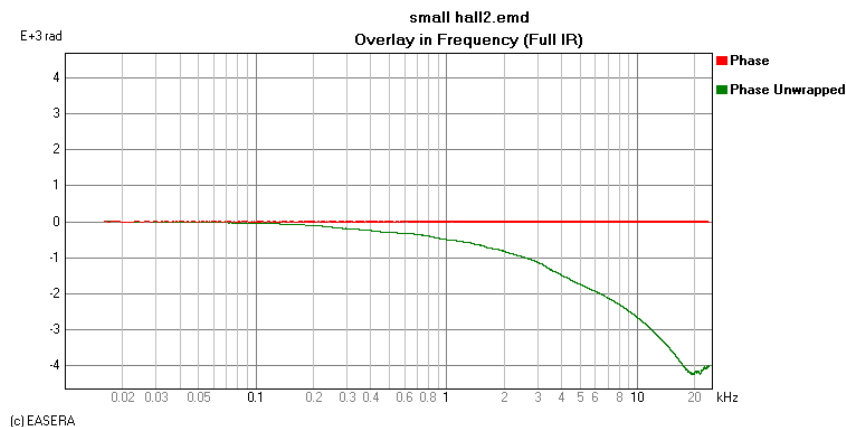
## Phase

The phase is represented in radians. Pi – i.e. approx. 3.1415927 – corresponds to 180°. Usually the wrapped *Phase* is shown, where the value is always between +/- Pi.

The phase behavior is in principle always the superposition of a run-time phase and the actual phase response of the source signal. The run-time phase is a fixed delay usually caused by the physical offset between the source and receive positions for the measurement.

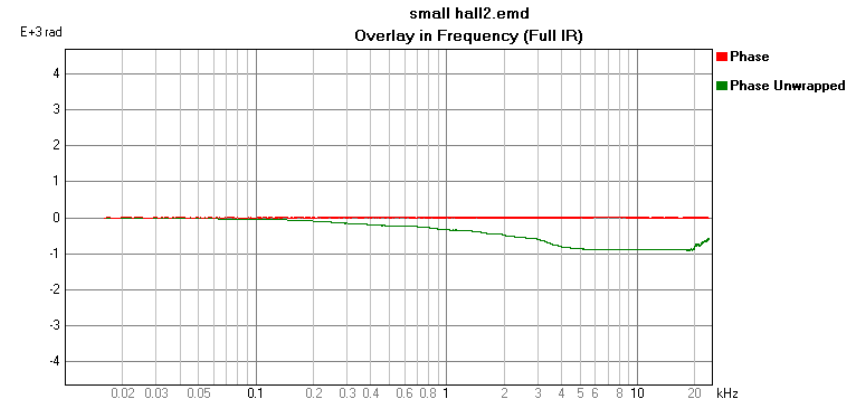


The phase may also be shown unwrapped, in which case the absolute phase behavior is shown:



The red curve shows the wrapped representation with an amplitude of  $\pm\pi$  radians, whereas the unwrapped phase shows negative values of more than 4000 radians ( $\Phi = 2 * \pi * t$ ). Therefore the wrapped phase curve looks like a horizontal trace.

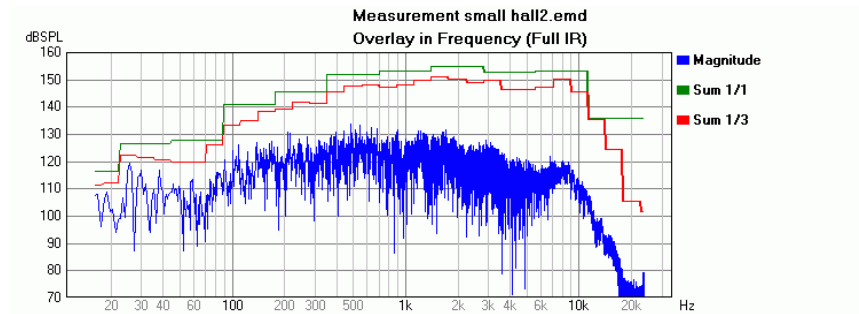
The phase behavior depends primarily on the position where the main impulse is located within the impulse response. The following diagram shows the same phase behavior after *Edit/Cyclic Move/Move Abs Max to Zero* has been selected.



(c) EASERA

## Sum and average

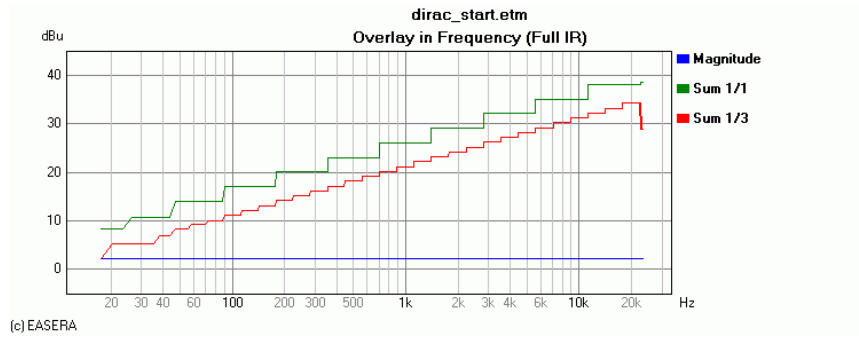
By selecting *Sum 1/1* or *Sum 1/3* it is possible to show the sum of the individual signal components for the individual octave or one third octave bands.



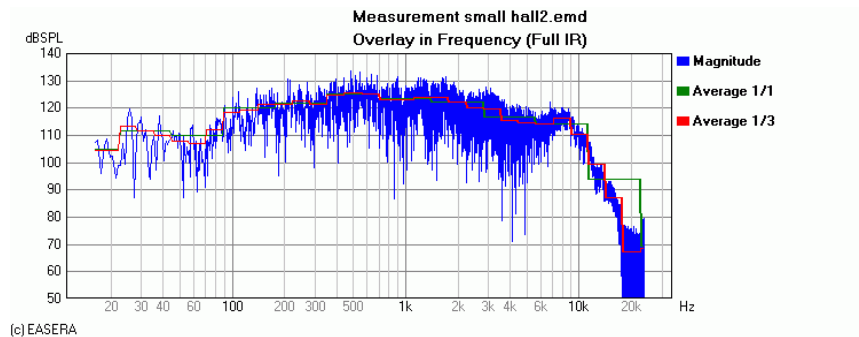
(c) EASERA

The sum in one octave band is always higher than that in the three individual one third octave bands, which in its turn is higher than that of the individual values in the frequency response.

As mentioned before, the result of an FFT analysis is linearly distributed over the frequency range, meaning that for a constant function there is the same energy in every frequency span (1 – 2 kHz, 2 – 3 kHz, 3 – 4 kHz). When you sum this energy and divide it into equal one third or one octave bands then you will have a band energy that increases linearly with the frequency in every octave band (1 – 2 kHz, 2 – 4 kHz, 4 – 8 kHz). This then results in a display that increases in level with an increase in frequency.



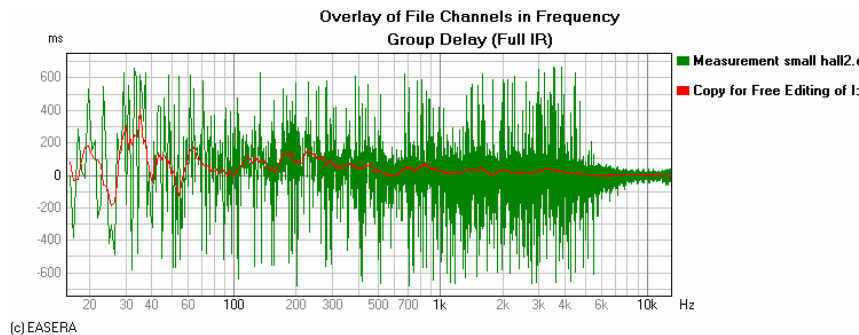
The result would look quite different with the averages (*Average 1/1* or *Average 1/3*):



The frequency response as well as the average (in the one third octave as well as the octave bands) form a mutual curve here.

## Group delay

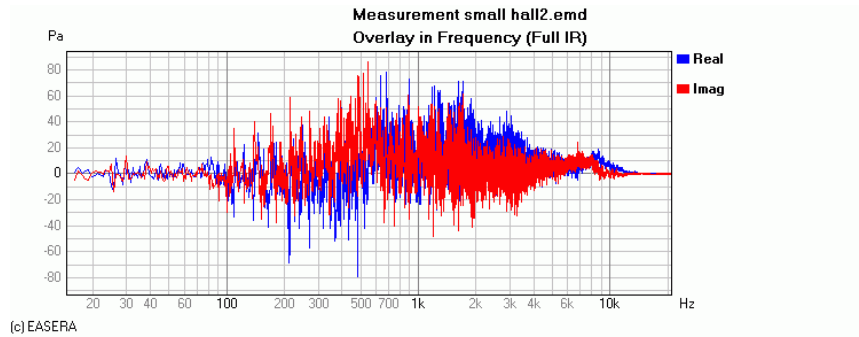
The *Group Delay* represents the delay of the signal dependent on the frequency.



To obtain usable results from acoustical measurements, the signal will need to be smoothed considerably. Such smoothing is allowed by EASERA, but a copy for free editing must be created first by selecting *Edit/Duplicate File*, then select *Duplicate For Free Editing* in the dialog box and click *OK*. To smooth the signal select *Edit/Smooth*, then select *1/3rd Oct.* for the *Standard Logarithmic Smoothing* in the dialog box and click *OK*.

## Real and imaginary parts

By FFT analysis, it is possible to derive a complex frequency response from an impulse response. Usually it is shown as modulus (frequency response) and phase information. It is, however, also possible to visualize it as a real and an imaginary part.



## Calculated values and diagrams

EASERA is able to derive all relevant acoustical quantities from the impulse response and display them either numerically or in a diagram. These calculated functions are located on the *Graphs* navigator page in the *Calculation* folder.

### Arrival, C50, D/R, S/N

The results which can only be numerically stated, are summarized under the item *Arrival, C<sub>50</sub>, D/R, S/N*. These are shown on the *Details* navigator page, while the active curve is displayed in one of the time diagrams (impulse response, ETC...). To specify which detailed information is to be displayed on the *Details* page, check the boxes for the desired items in the *Options (F9)* dialog box in the *Details Table* section on the *VIEW & CALC|LAYOUT* page.

Data		
Arrival	28.73	ms
Distance	9.87	m
C50	4.5	dB
C80	6.2	dB
C35	3.3	dB
D	0.740	
L7	113.1	dB SPL
L50	114.1	dB SPL
L80	114.5	dB SPL
L35	113.7	dB SPL
Ltotal	115.4	dB SPL
Center Time	57.23	ms
Noise	7.07	mPa
Noise DC	225.45	μPa
Mean (DC)	0	aPa
Abs. Max	4.6	Pa
RMS	46.01	mPa
SNR	16.3	dB
Crest	40	dB
Section		
From	0	s
To	1.4	s
Mean (DC)	0	aPa
Abs. Max	4.6	Pa
RMS	46.01	mPa
SNR	16.3	dB
Crest	40	dB
D/R Ratio	Left<Arrival	
Split Time	35	ms

Most of the values are also shown on the *Results* page.



## Arrival & Distance

The arrival time is the period between the start of the measurement and the arrival of the signal, in milliseconds. The arrival of the signal is assumed to be the moment when the signal has reached a certain level above noise. This level may be preset in the *Options (F9)* dialog box in the *Arrival Time* section on the PROCESSING|PROCESSING page and the default value is 35 dB. The distance calculated from the *Arrival* time (using a temperature of 20° C) is displayed as *Distance*.

$$C_{50}, C_{80}, C_7, C_{35}, C_{split}$$

The quantity  $C_{50}$  (measure of definition according to Ahnert) describes the ratio between the energy that arrives during the first 50 ms after *Arrival* and the energy arriving thereafter.

$$C_{50} = 10 \log \left( \frac{\int_0^{50 \text{ ms}} p^2(t) \cdot dt}{\int_{50 \text{ ms}}^{\infty} p^2(t) \cdot dt} \right) \text{ dB}$$

The quantity  $C_{80}$  (measure of clarity according to Abdel Alim) describes the ratio between the energy that arrives during the first 80 ms after *Arrival* and the energy arriving thereafter.

$$C_{80} = 10 \log \left( \frac{\int_0^{80 \text{ ms}} p^2(t) \cdot dt}{\int_{80 \text{ ms}}^{\infty} p^2(t) \cdot dt} \right) \text{ dB}$$

The higher this clarity value, the higher the intelligibility.  $C_{50}$  is used for speech,  $C_{80}$  for music.

$C_7$  is also called direct sound measure and describes the ratio between the energy arriving during the first 7 ms and the energy arriving thereafter. This value is frequently negative and lies between -10 and -15 dB in typical rooms.

The quantity  $C_{split}$  describes the ratio between the energy arriving before the split time after *Arrival* and the energy arriving thereafter. This split time can be set in the *Options (F9)* dialog box and is 35 ms by default – where the default quantity could then be specified as  $C_{35}$ .

## D

The quantity  $D$  (*Definition* according to Thiele) describes the ratio between the energy arriving during the first 50ms after *Arrival* and the overall energy. From this measure derived in the 1950's the  $C_{50}$  measure was developed.

$$D_{50} = \frac{\int_0^{50 \text{ ms}} p^2(t) \cdot dt}{\int_0^{\infty} p^2(t) \cdot dt}$$

**$L_7$ ,  $L_{50}$ ,  $L_{80}$ ,  $L_{35}$  /  $L_{Split}$** 

$L_7$  is the level during the first 7ms.  $L_{50}$  and  $L_{80}$  correspond to the level during the first 50 ms or 80 ms, respectively, and  $L_{Split}$  is the level during the split time – or 35 ms by default.

**Total SPL**

This is the integrated overall level in dB.

**Center Time**

The *Center Time* (according to Kuerer) is the center time of the integrated energy. This is the point in time where the energy received before the time is equal to the energy received after the time. It is calculated according to the following formula:

$$T_s = \frac{\int_0^{\infty} t \cdot p^2(t) \cdot dt}{\int_0^{\infty} p^2(t) \cdot dt}$$

**D/R Ratio**

*D/R Ratio* calculates the ratio between the direct sound energy and the reflected sound energy. Put the markers at the beginning and the end of the reflections in the impulse response (the markers define the reverb component which is then related to the direct component. If one sets the left marker at + 50 ms and the right one at the end of the IR, the result should be approx.  $C_{50}$ ).

**Early and Reverb**

*Early* calculates the sound pressure up to the left marker, and if the markers are set appropriately, the direct-sound component. *Reverb* calculates the sound pressure in the range between the two markers, and if the markers are set appropriately, the reverberation component.

**Noise**

The noise floor is automatically determined by comparing the late part of the impulse response with the very early part before the arrival peak. For this purpose several time sections with an approximately flat average amplitude are compared and the part with the lowest signal is identified as being the background noise. (Due to the nature of this algorithm, sometimes wrong estimates are possible, for example if the room response does not sufficiently decay over the time period covered by the impulse response. If the program is not able to unambiguously detect it, the noise is usually displayed as being zero.)

**DC (Noise) and Mean / DC**

*DC (Noise)* is the steady component of the noise floor, *Mean / DC* is the steady component of the overall signal.

**Abs. Max and RMS**

*Abs. Max* is the size of the signal peak, with standardized impulse responses of 1.00 Val, and *RMS* is the effective value of the signal.

**INR and SNR**

*INR* is the impulse-peak/noise ratio in dB, and *SNR* is the signal-to noise ratio in dB.

**Crest**

*Crest* is the ratio between peak and effective values in dB.

**Section**

*Section* shows information on the signal when setting the two markers:

**Section from**

Position of the left marker in ms.

**Section To**

Position of the right marker in ms.

**Section Mean (DC)**

The steady component of the signal between the two markers.

**Section RMS**

Effective value between the markers.

**Section SNR**

Signal-noise ratio between the markers.

**Section Crest**

Ratio between peak value and effective value, in dB, calculated between the two markers.

**Split Time**

*Split Time* is set in the *Options (F9)* dialog box in the *Calculation* section on the PROCESSING|PROCESSING page and serves as the basis for  $L_{Split}$  and  $C_{Split}$ . This value is shown for the sake of information.

**Schroeder RT**

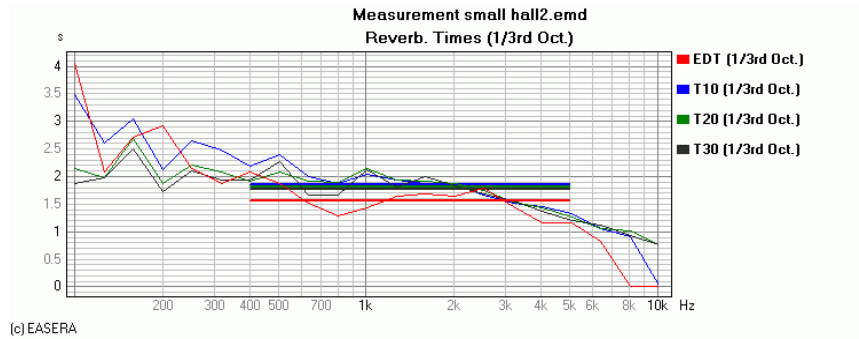
The function *Schroeder RT* was already described in chapter 9.

**STI, STIPa, RaSTI, MTI**

The speech intelligibility functions were already described in chapter 11.

## EDT, RT

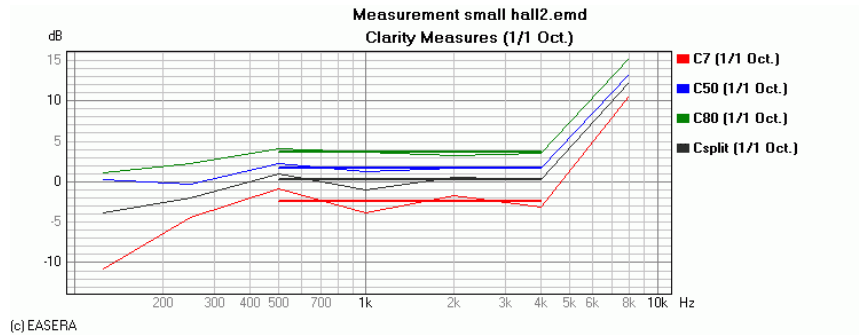
The reverberation times  $T10$ ,  $T20$  and  $T30$  as well as the early decay time  $EDT$  can be shown in broadband, as well as for every octave band ( $EDT$ ,  $RT$  (Octaves)) or one third octave band ( $EDT$ ,  $RT$  (1/3rd)).



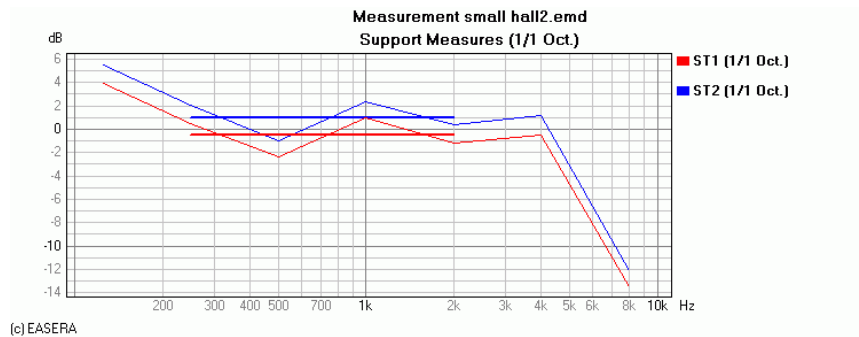
The display of broadband values as horizontal lines can be enabled or disabled in the *Options* (F9) dialog box in the *Show Lines* section on the VIEW & CALC | OPTIONS page.

## $C_{50}$ , $C_{80}$

Using the function  $C_{50}$ ,  $C_{80}$  it is possible to graphically display the C values ( $C_7$ , measure of definition according to Ahnert  $C_{50}$ , measure of clarity according to Abdel Alim  $C_{80}$ ,  $C_{split}$ ) for the octave bands ( $C_{50}$ ,  $C_{80}$  (Octave)) or the one third octave bands ( $C_{50}$ ,  $C_{80}$  (1/3rd)).



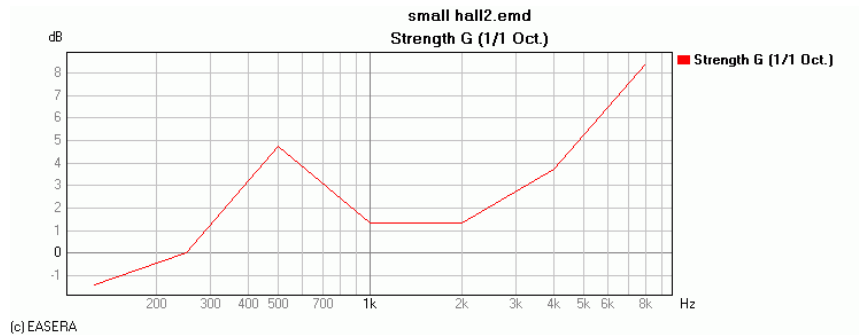
## Support ST



*Support ST* (according to Gade) is a measure for the acoustical support by the surrounding room in the stage area. In this context ST1 is a measure for the mutual hearing on stage and ST2 is a measure showing the degree in which the room supports the musicians by reflections (room response). These measures are recorded on the stage with the microphone at 1m distance from the sound source. Typical values in a concert hall are about  $-15$  dB to  $-12$  dB.

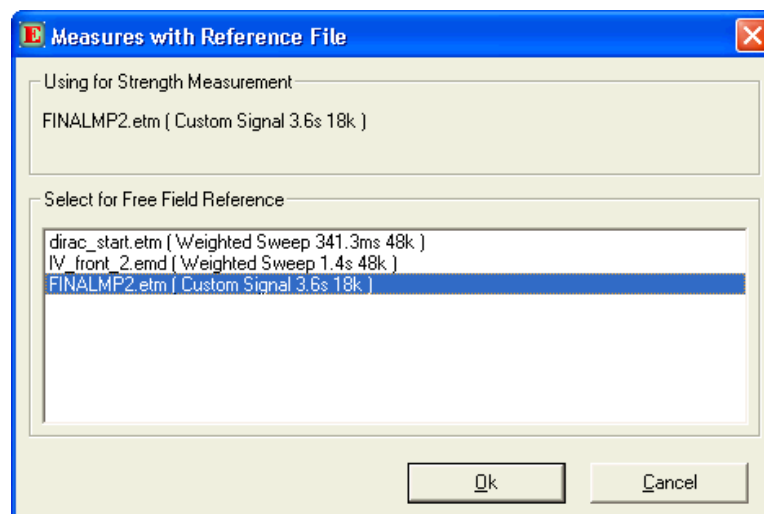
The support quantities can be indicated by EASERA in octave as well as one third octave bands.

## Strength G



The *Strength* measure  $G$  (according to P. Lehmann) is the ratio between the sound energy at the measuring location and a specified reference (according to DIN ISO 3382, measured at 10m distance from the sound source in the free field). In general terms  $G$  is thus understood as a measure for the overall gain of the emitted sound by the surrounding room as compared with the free field.

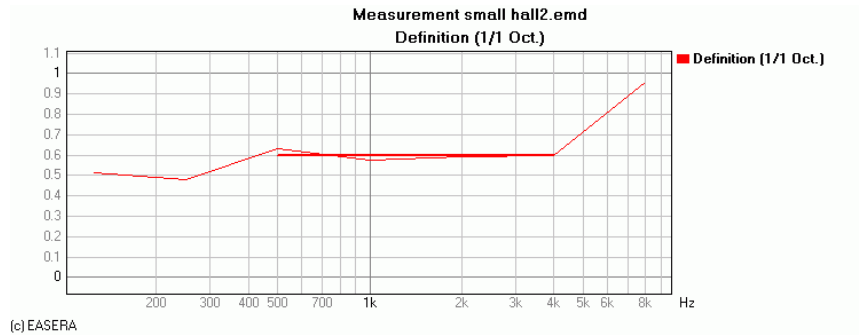
When first selecting this function for a measurement, it is necessary to provide a second measurement as the reference.



If another file is to be used later as the reference, the existing reference must be removed in the *Measurement File Properties (F4)* dialog box in the *Measures With Reference File* section on the CALCULATE OPTIONS|GENERAL PROCESSING page. These values are also shown on the *Results* page.

In the medium frequency range (500 Hz to 1 kHz)  $G$  should be above 0 dB. The *Strength* quantity can be shown by EASERA in octave as well as one third octave bands.

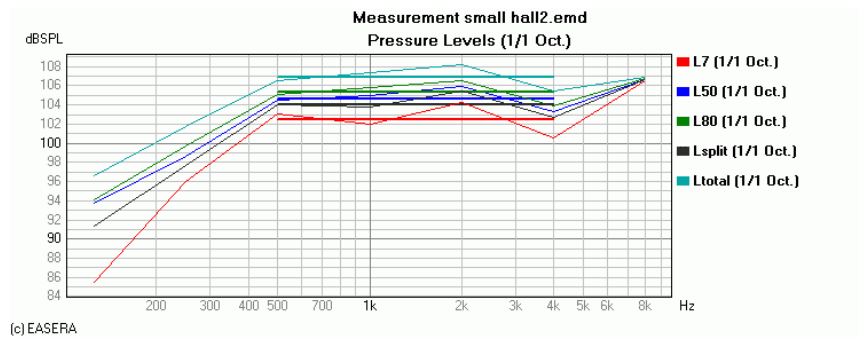
## Definition



The definition  $D$  (also  $D_{50}$ ) according to Thiele is the ratio between the energy received during the first 50 ms and the overall energy. For the sake of good speech intelligibility the value of  $D$  should lie above 0.5 for all frequencies.

The *Definition* quantity can be shown by EASERA in octave bands as well as one third octave bands.

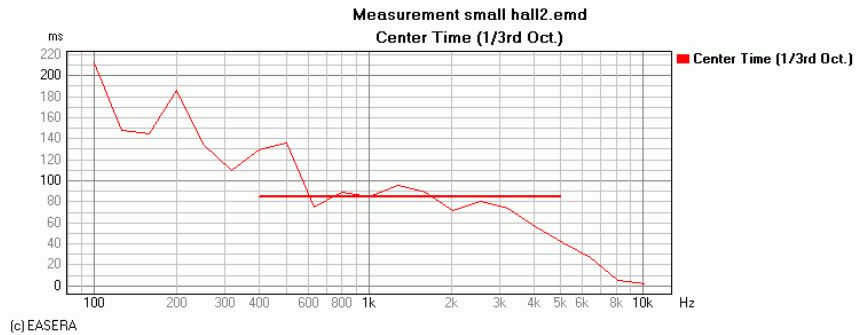
## $L_{50}$ , $L_{80}$



Using the level functions it is possible to calculate the sound pressure levels from an impulse response. Displayed quantities are in octave or one third octave bands:

- the level within the first 7 ms ( $L_7$ ).
- the level within the first 50 ms ( $L_{50}$ ).
- the level within the first 80 ms ( $L_{80}$ )
- the level within the split-time ( $L_{Split}$ )
- the overall level (*Total SPL*)

## Center Time



The *Center Time* (according to Kuerer) is the center of the integrated energy. This is the point in time where the energy received before the time is equal to the energy received after the time. It is calculated according to the following formula:

$$T_s = \frac{\int_0^{\infty} t \cdot p^2(t) \cdot dt}{\int_0^{\infty} p^2(t) \cdot dt}$$

Here it is shown for the individual octave bands. EASERA is also capable of displaying it for one third octave bands. To obtain satisfactory speech intelligibility it should be below 130 ms.

It goes without saying that the evidence of the center time corresponds also with other acoustical measures, e.g. with the above explained energy sum, the development of which correlates on principle with the position of the center time. Both are indicative of whether the energy arrives "temporally diffuse", i.e. partially delayed, which for music can be rather acceptable or even desirable (soft sound build-up), whereas for speech a higher temporal concentration is advantageous (hard sound build-up).

# Lesson 13: In situ measurements

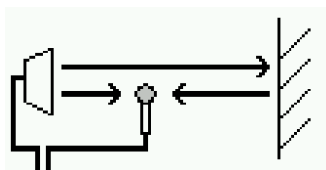


Knowledge of the reflection behavior of all larger surfaces is essential for room acoustical simulations. But corresponding information is often not available, particularly for existing buildings, and in most cases it is also not possible to take a sufficiently large material sample to perform appropriate lab investigations. In such cases, EASERA offers the very helpful possibility of determining the reflection coefficients “live” in situ.

## The measuring procedure

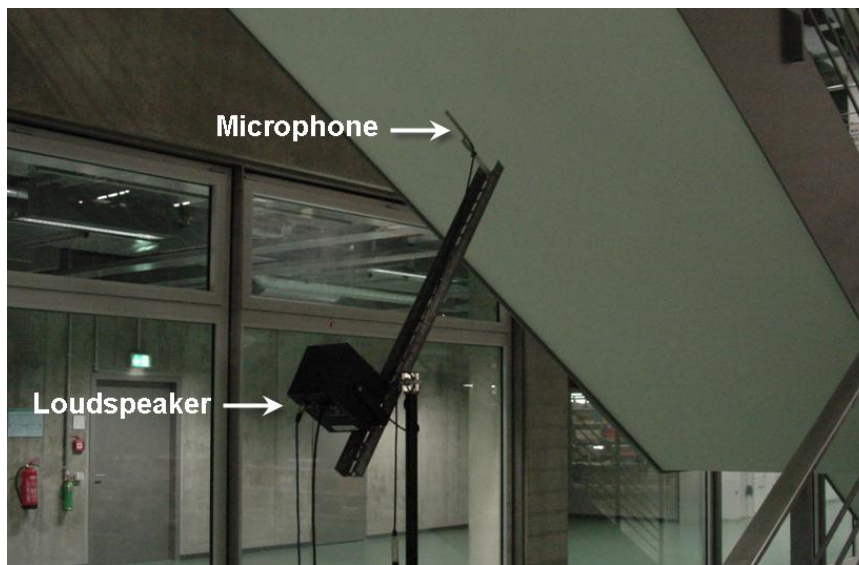
The reflection coefficients are determined in post-processing by two measurements: That of the impulse response in front of the test object and that of the impulse response in the free-field.

For the measurement in front of the test object a loudspeaker is directed perpendicular to the surface in question and the measuring microphone is arranged between the loudspeaker and the surface.



This will obtain an impulse response with two significant impulses: One from the direct sound and one from the reflected sound.

The following photograph shows an arrangement for a measurement carried out in the VW Library of the TU/UdK in Berlin.

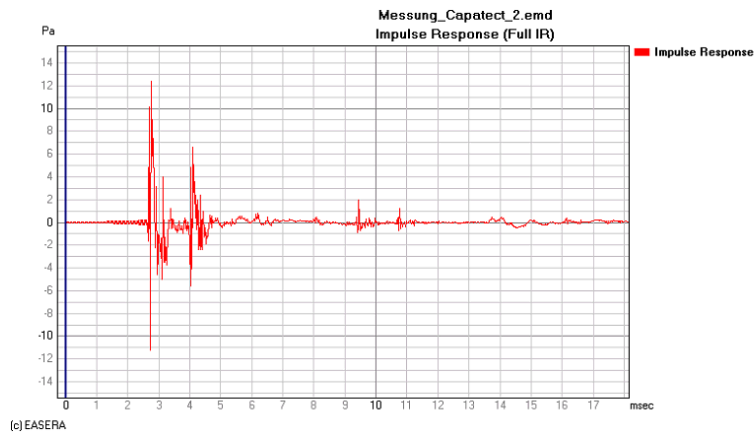


The surface to be tested must be plane and sufficiently large to also obtain realistic values for low frequencies.

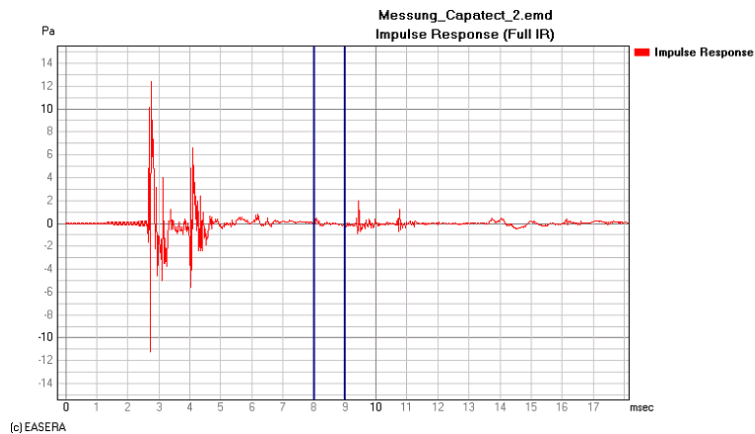
## Making the measurements

For the first measurement, the loudspeaker-microphone combination is directed at the surface to be tested. Then the measurement is carried out in the usual way.

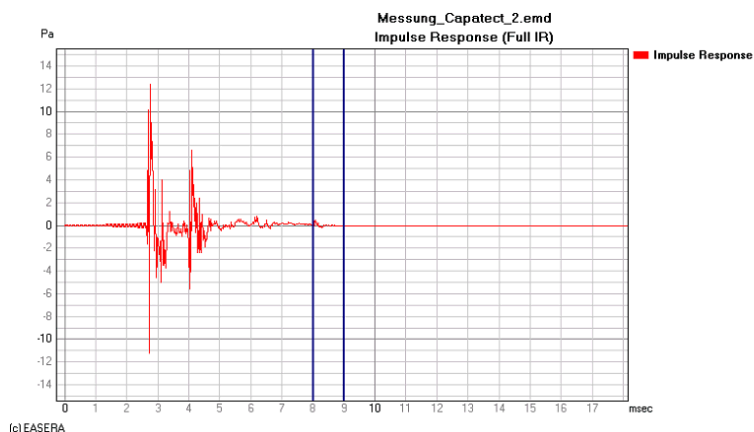
### Measurement in the direction of the surface to be tested



Here we can clearly see the two impulses, the first from the direct sound, and the second from the reflection. Then reflections follow from other surfaces. These later reflections are undesirable and need to be windowed out.



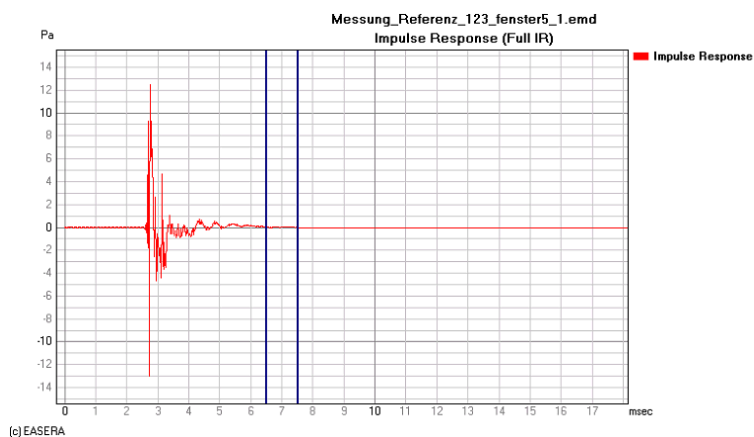
The markers are set so that they limit the range to be evaluated to just before the first disturbing distinct reflection



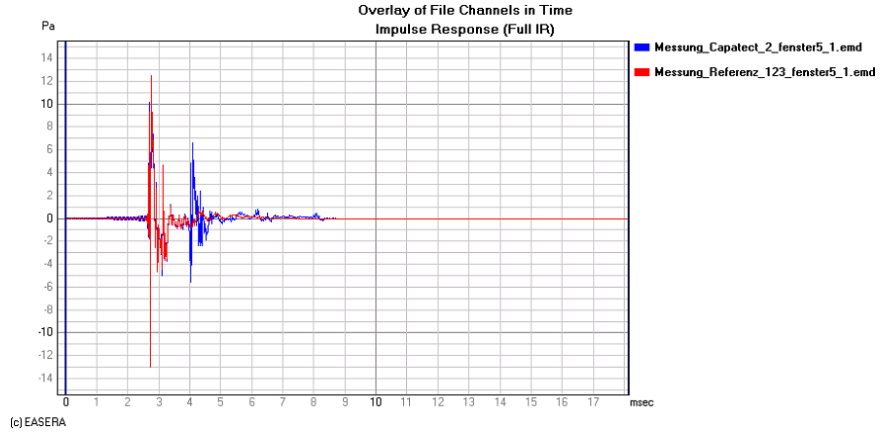
The windowed impulse response should be about 5 ms long. Therefore the low-frequency limit can be expected to be approximately 200 Hz.

## Measuring the free field

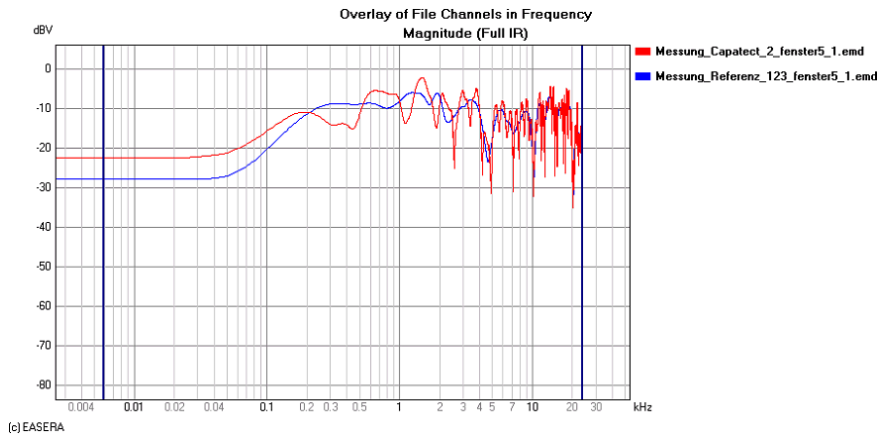
Now the measuring arrangement is turned away from the wall (e.g. into the room), so that the surfaces facing the microphone are as far away as possible. Then the measurement is carried out again and also here appropriately windowed.



Let us briefly compare the two measurements in an overlay of the impulse response. It is not astonishing that the first impulse of the two measurements is found at exactly the same place if one remembers that the loudspeaker and microphone are firmly mounted on a common rail.

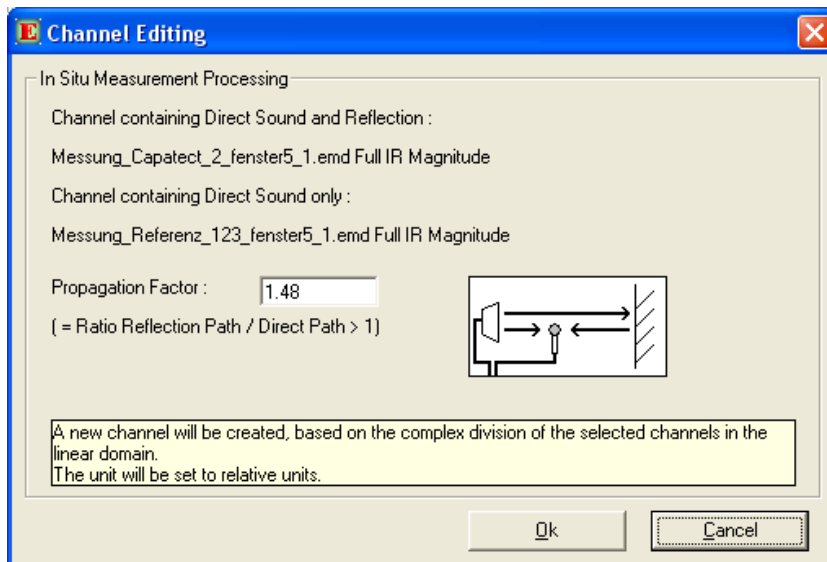


And now an identical overlay for the frequency response. Not altogether unexpected, there are two responses showing more and sharper interference effects by comparison with the response of the freefield measurement.



## Calculation of the transmission behavior

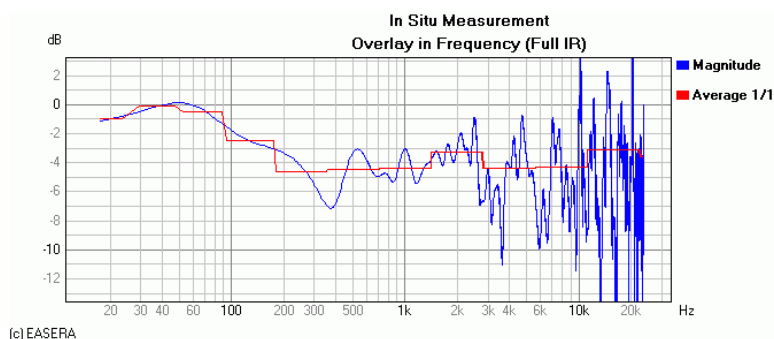
Based on the two measurements, EASERA calculates the complex transfer function of the examined surface. To do this please create an overlay of the two measurements and take care that the measurement directed towards the wall is selected as active. Then select *Edit / More / In Situ Measurement Processing*.



Please check here if the measurements were correctly chosen and enter the *Propagation Factor* – this is the quotient of the two travel distances from the loudspeaker to the microphone, the reflection path being the numerator and the direct path the denominator.

The impulses can be localized at 2.735 ms and 4.05 ms. Since only the ratio of the path lengths is interesting for the propagation factor, a conversion of the times into path lengths can be omitted and the quotient be calculated directly from the times.

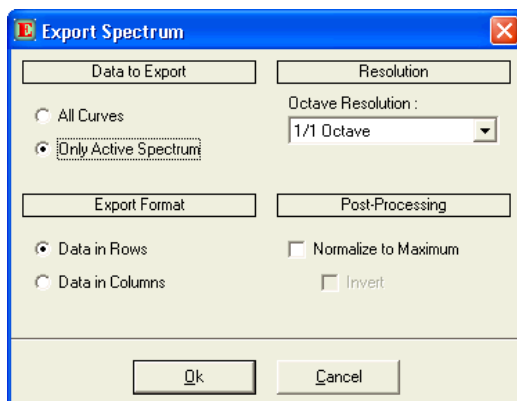
As a result we obtain the reflection factor in dB.



In the unsmoothed representation we obtain values above 0 dB for the individual frequencies. Since a wall cannot reflect more energy than is directed at it, these results cannot be considered as real. They are due to interferences and the resulting narrow level notches in the direct sound.

More sensible is the representation of the reflection factor  $r$  in octaves or third octaves, because with these such errors are smoothed out. The absorption coefficient can be calculated by means of the formula  $\alpha = 1 - |r|^2$ .

To do this it is possible to export the spectrum using *File / Export Spectrum* so that further processing can be continued, for instance, in a spreadsheet program (Excel or the like):



## Accuracy of the procedure

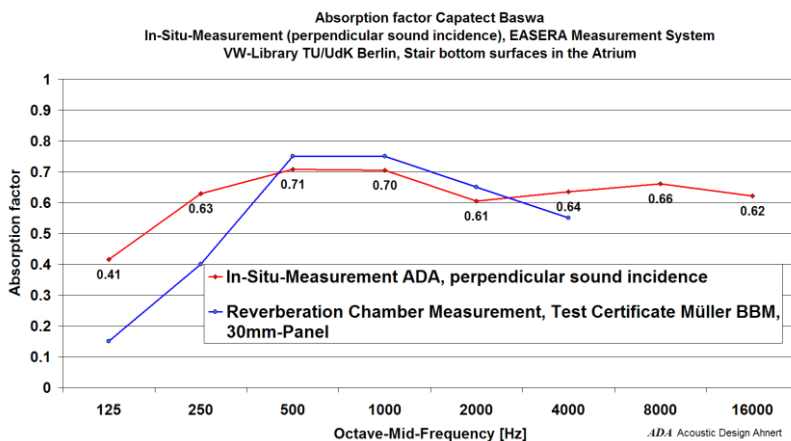
Absorption coefficients are generally measured in a reverberation chamber or in a Kundt tube. The determination of the absorption coefficients by the in situ procedure does not lead to exactly identical values.

As a consequence of the principle used, the absorption coefficient is determined here for a plane frontal sound incidence perpendicular to the surface, while in the reverberation chamber there prevails a random sound incidence.

The surface of the object to be examined must be plane and perpendicular sound incidence has to be guaranteed. Furthermore, the reflections from other surfaces must be sufficiently small or late enough to be windowed out.

The lower limiting frequency results on the one hand from the size of the surface (should be larger than the wave length) and on the other hand from the position of the window. With low frequencies the inaccuracies increase.

The following diagram compares an in-situ measurement with a measurement in a reverberant chamber. As can easily be seen, the deviations are below 10% with high frequencies and thus tolerable. With low frequencies the deviations are, however, considerable. The lower limit of approximately 200 Hz estimated by us for this case becomes quite evident.



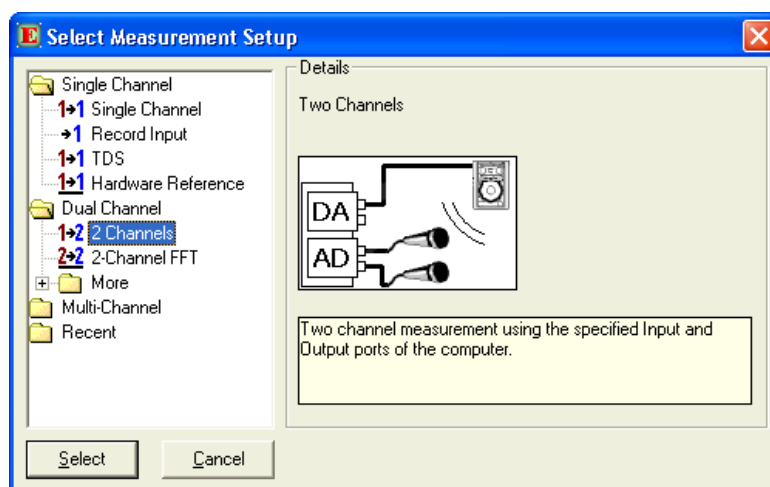
# **Lesson 14: Binaural measurements**

The previous measurements were mostly carried out with just one microphone, neglecting the reality that people hear with two ears. It is true that this is unimportant for quantities like reverberation time or frequency response, but EASERA also allows measurements which objectify binaural hearing.

## Two-channel measuring

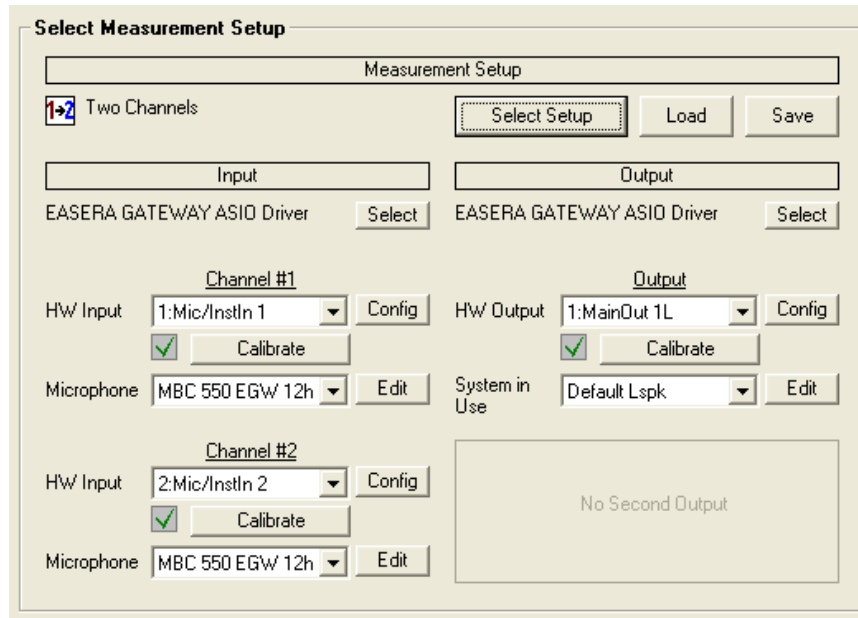
Functions like the *Interaural Cross Correlation (IACC)* or the *Lateral Fraction* correlate two impulse responses. In the case of the interaural cross correlation it is the left and the right channels of a dummy head. It would not be a good idea to measure the two channels one after the other, since just a slight time shift would likely produce unrealistic results.

To perform a two-channel measurement, we go to the *Measure* page in the *Select Measurement Setup* section and choose the *Select Setup* button. This opens the *Select Measurement Setup* dialog box where we select the option *2 Channels*.



Now we may choose the hardware individually for two channels and, of course, also calibrate the two channels individually:





Now if we perform a measurement, a results file that contains two channels appears in the file list on the *View & Calc* page:



These channels can be individually shown and also combined in overlays.

## Interaural cross correlation

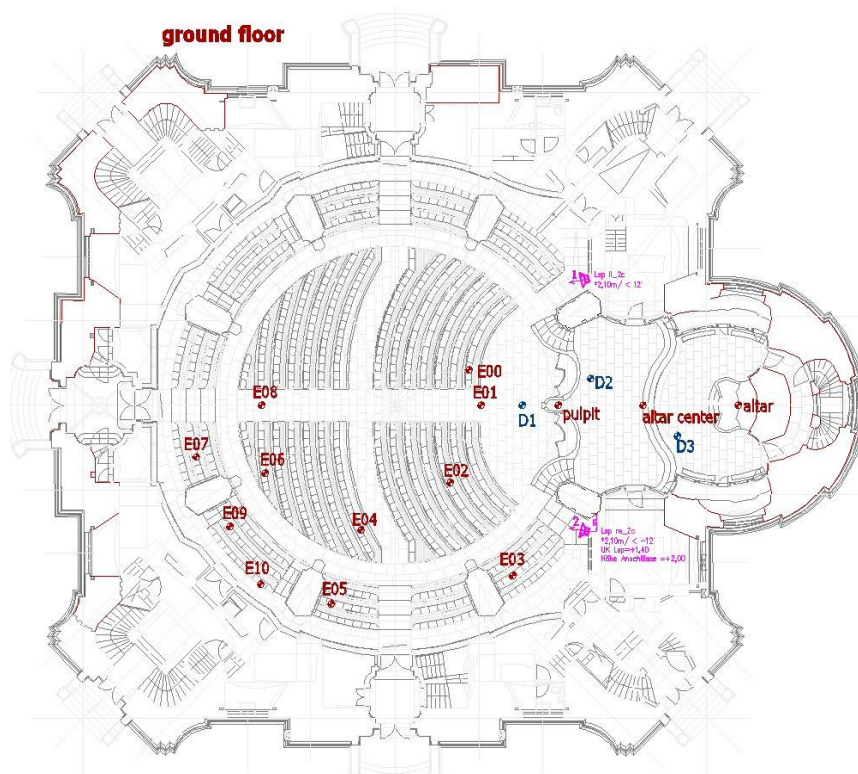
The interaural cross correlation is measured using a dummy head and compares the phase positions of the left and right channels (similarly to a correlation degree meter in studio engineering). With a correlation degree of 1 we have full correlation of the signals and with a degree of 0 there is no correlation.

The interaural cross correlation according to ISO 3382 is calculated for the individual octave bands using three separate time periods of the impulse response. EASERA calculates the first 80 ms of the impulse response (“Early”,  $IACC_E$ ), the part after 80 ms of the impulse response (“Late”,  $IACC_L$ ), as well as for the overall impulse response (“Full”).

According to Beranek, the value  $1 - IACC_E$  corresponds to the perceived width of the auditory impression, whereas the value  $1 - IACC_L$  correlates with the perceived “envelopment”. The octave bands 500 Hz, 1000 Hz and 2000 Hz are considered to be the most important quantities.  $IACC_E$  values between 0.28 and 0.38 are considered to be excellent to outstanding,  $IACC_E$  values between 0.39 and 0.54 as good to excellent, and  $IACC_E$  values between 0.55 and 0.59 as acceptable.

## Measurement in the Frauenkirche Dresden

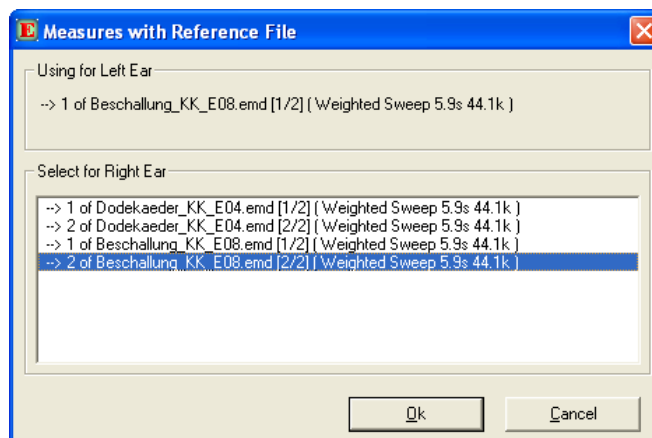
The following figure shows the ground floor plan of the Frauenkirche Dresden. This church was erected between 1726 and 1742 as the first baroque church in rotunda design. This famous edifice by George Bähr was completely destroyed in 1945 and reconstructed true to the original between 1992 and 2005.



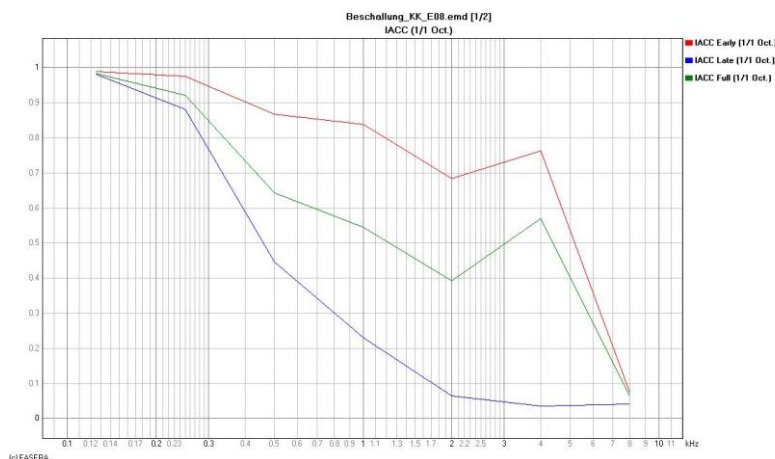
For the dodecahedron measurement presented here the sound source was located at position D1 in front of the pulpit and the dummy head was located at position E04. When measuring with the sound reinforcement system the talker microphone was placed in the altar area at position D2 and the dummy head at position E08.

## Calculating the interaural cross correlation

To calculate the interaural cross correlation one chooses the first channel (Left Ear) for this measurement and then selects the function *IACC (Octave)*. Then EASERA shows a dialog box to select the second channel (Right Ear):

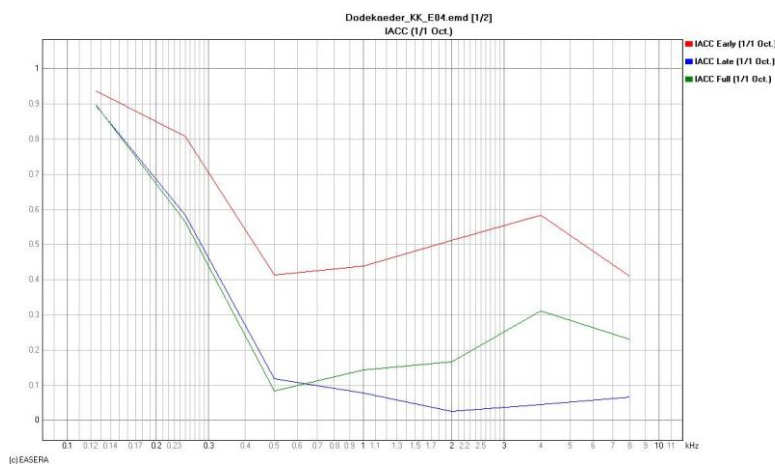


The calculation takes a few seconds, and then EASERA displays the result:



With the sound reinforcement system (many sources!) we obtain high  $IACC_E$  values, which is indicative of a high correlation of the two ear signals during the first 80 ms (E08 is a location in the center of the church). The high correlation is certainly due to the two line arrays symmetrically arranged on the ground floor. These sources dominate the early sound for this location. Only with the arrival of delayed sound-field components from the galleries does the correlation diminish (the  $IACC_L$  values are lower).

For comparison, we show the curve obtained with the dodecahedron measurement at position E04:



$IACC_E$  shows good values in this diagram. The correlation for the late signal components is, however, too low, which must be expected in highly reverberant churches.

# Lesson 15: Measurement on electronic circuits

Using EASERA it is not only possible to perform acoustical measurements, but also measurements on electronic and electro-acoustical devices. In this lesson we would like to measure a stereo equalizer (the BSS Varicurve).

## Hardware reference

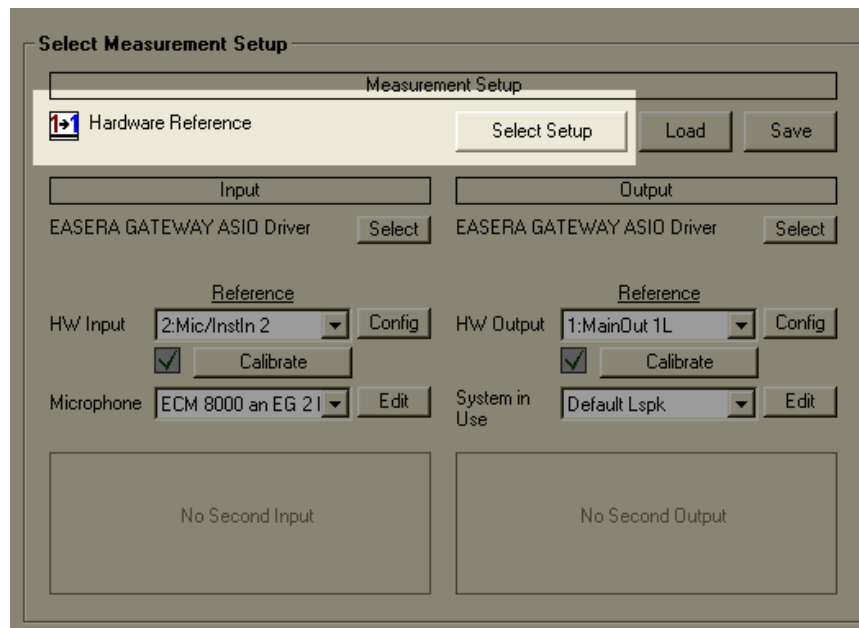
If room acoustical measurements or measurements on loudspeakers are carried out, the transmission properties of the soundcard used must be at least by one order of magnitude better than those of the object to be measured. Deficiencies of the soundcard (frequency response, harmonic distortion factor, noise) are then negligible.

This does not hold true, however, for measurements on electronic devices where the results could be considerably influenced by the properties of the soundcard. Just connect the input of the soundcard using a short cable with the output and perform a measurement. The result could be disillusioning.

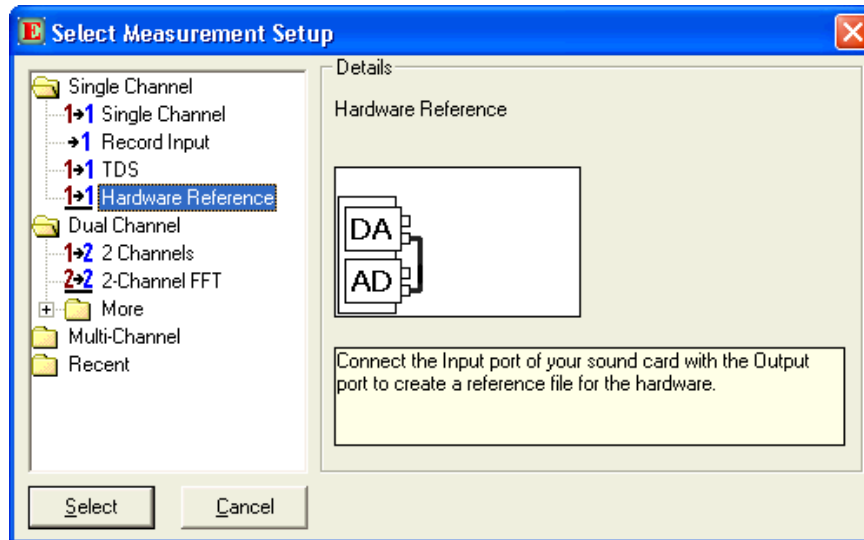
To minimize such interference of the measuring results EASERA enables using a hardware reference: To do this the soundcard output is connected to the input and a measurement carried out. The deviations from ideal transmission are then detected and will be compensated in the subsequent measurements.

## Creating a hardware reference

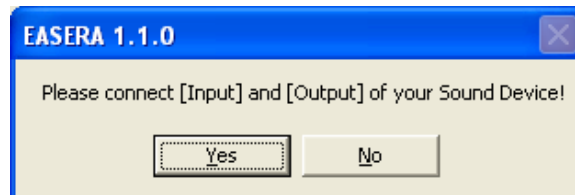
To create a hardware reference please go to the *Measure* page and in the section *Select Measurement Setup* click on *Select Setup*.



Now choose the *Hardware Reference* item under *Single Channel* in the tree:

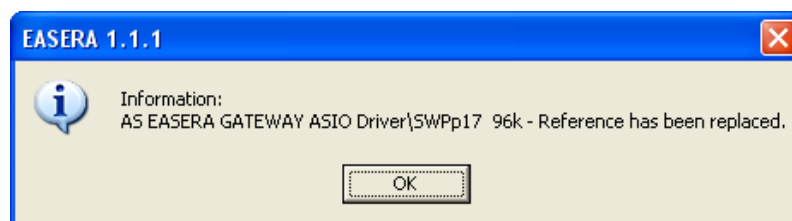


Since reference measurements are to be created specifically for the signal to be used for the later measurement, first adjust the parameters of this signal (signal type, sample rate and signal length, e.g. a *Log-Sweep* with 96 kHz and 1.4 s). Please also make sure that the levels are correctly set (first of all in order to avoid clipping) and then start a measurement.

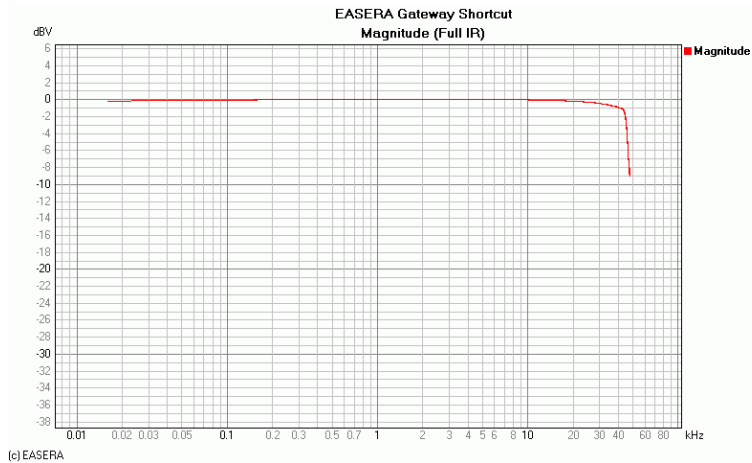


EASERA reminds you to connect the input with the output. Do so if necessary and then click on *Yes*.

Then a measurement is made and if applicable you are informed that previous reference files have been replaced.

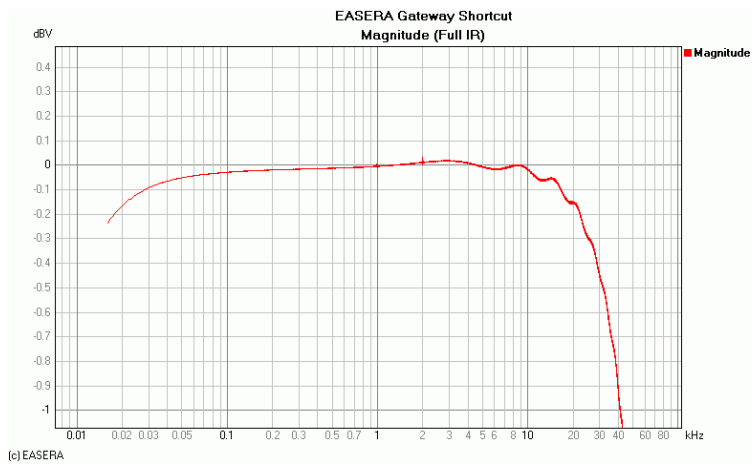


Let us have a look at the result of this hardware reference. First the frequency response: The audio range is linear, then a steep drop at 48 kHz, which is to be expected with a sample rate of 96 kHz.

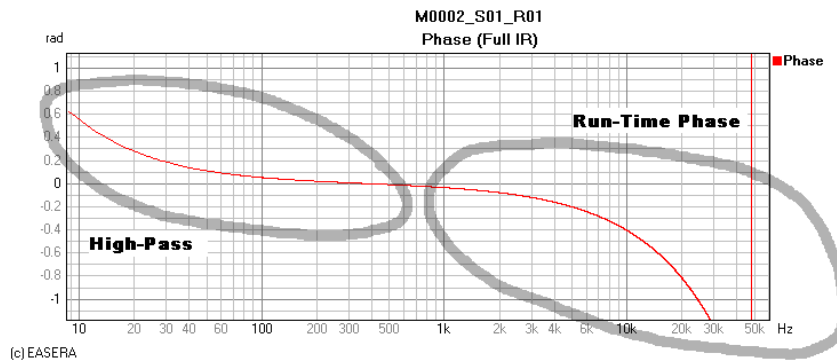


Let us have a closer look at this linearity: The curve is no longer quite so impressive. In the bass range the curve drops by a few tenths of dB and in the treble range it also falls a bit.

(By the way: If you detect imperfections at the multiples of the mains frequency (in USA 60 Hz) with an EASERA Gateway interface, you should try using a DC power unit.)



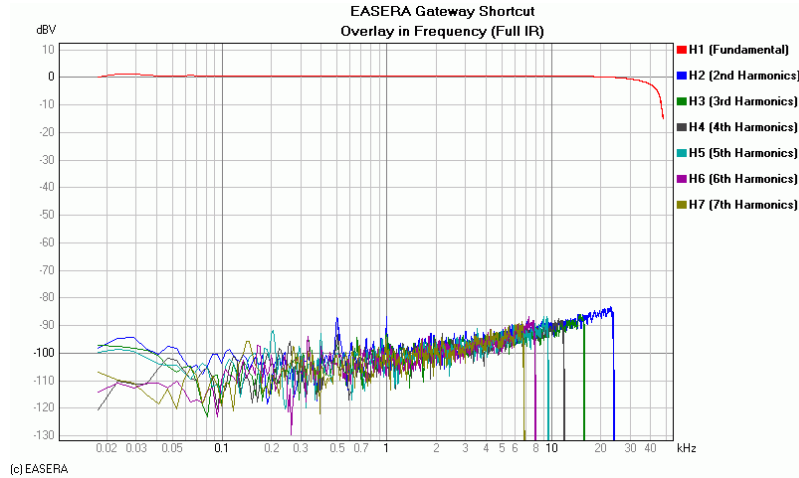
The next glance is at the phase response.



The experienced user would recognize here that the deviation from an ideal phase behavior has two reasons: With the low frequencies we have a high-pass behavior, but without the second reason the phase would be

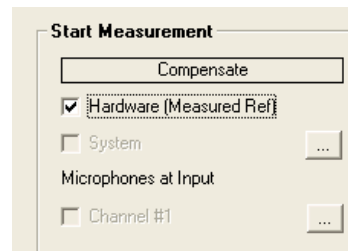
equal to zero for all frequencies above 500 Hz. The second reason is a run-time phase attributable to the fact that the peak of the impulse-response impulse does not lie on the first time sample (instant 0), but on the second time sample (instant 10.4  $\mu$ s; a sample rate of 96 kHz equates to a sample period of 10.4  $\mu$ s.)

Last but not least we look at the distortion spectra. For reference a value of -80 dB would correspond to a harmonic distortion of 0.01%.



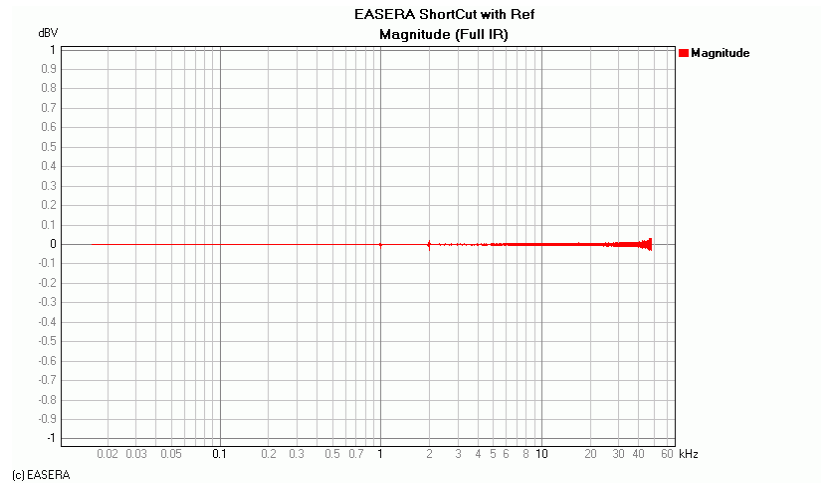
## Using a hardware reference

We leave the cable which we have used to loopback the soundcard output to the input in place to perform another measurement. But first we activate the *Hardware (Measured Ref)* option in the *Start Measurement* section on the *Measure* page.

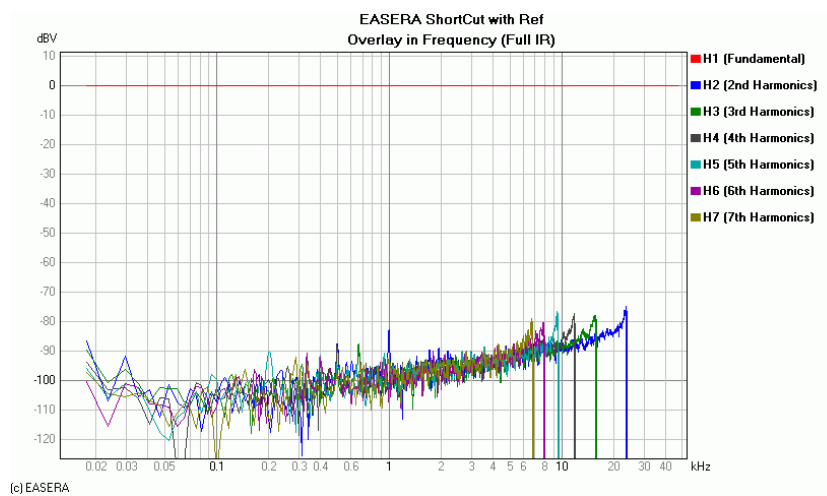


Then click the *Go* button to make another measurement. If we now look at the frequency response, all looks pretty linear – which is indeed not astonishing after having just compensated precisely for this hardware. The phase behavior would be similarly linear.





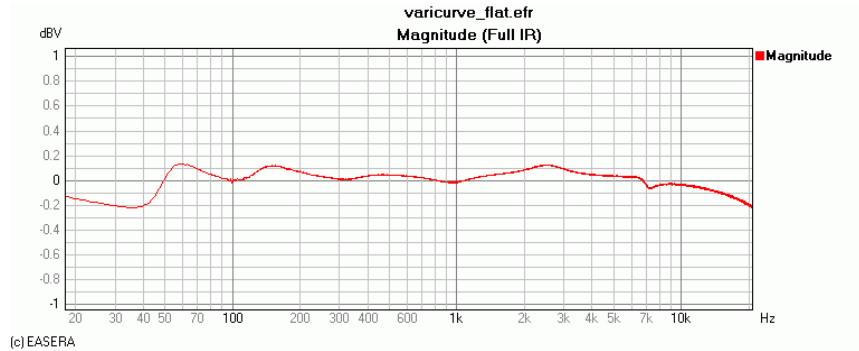
We again look at the distortion spectra:



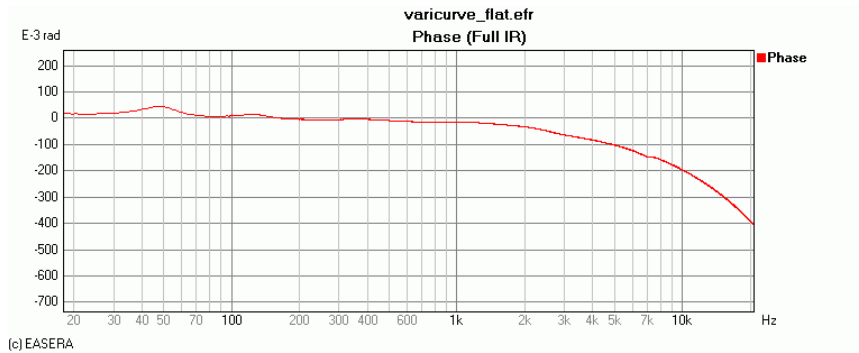
Now we observe that the compensation had negligible influence on the distortion. The non-linearities increasing at higher frequencies is typical and with a maximum of  $-80$  dB they are low in the range relevant to us.

## Measurement of the device

The BSS Varicurve is a digitally controlled analog stereo equalizer, one channel of which we now want to connect into the signal path. First we set to the equalizer for a flat frequency response, and then we perform another measurement and first view the frequency response:



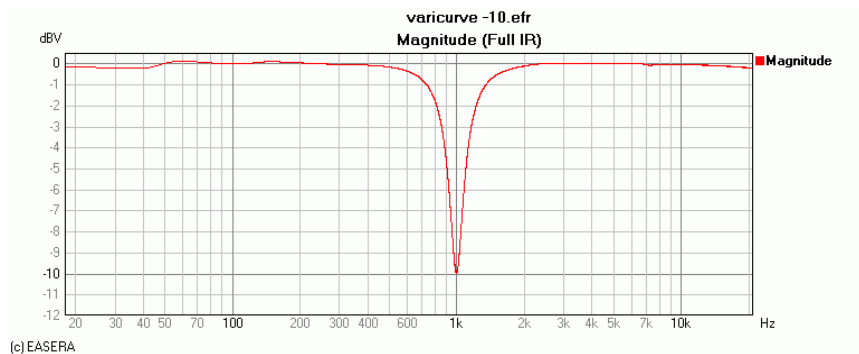
We see a ripple of +/- 0.2 dB here, which is very suitable. Equally perfect is the phase behavior.



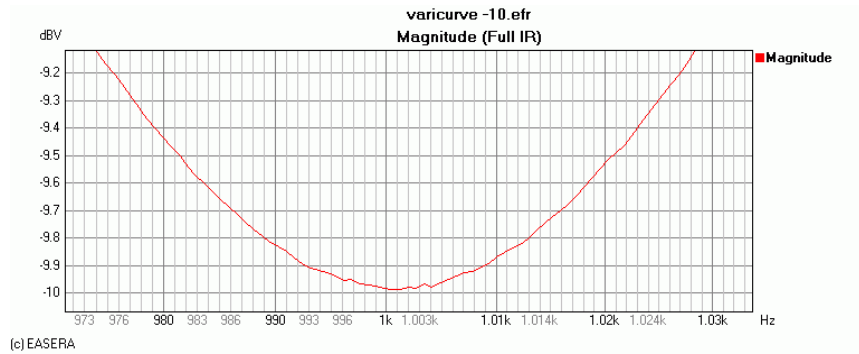
The harmonic distortion components occurring in a measurement with an EQ do not differ from those occurring in a measurement without an EQ, so we do not show a diagram here.

### 1 kHz full cut

The measurements have so far been carried out with a neutral EQ setting. Now we set a filter of -10 dB with a frequency of 1 kHz and a bandwidth of 0.5 octave:

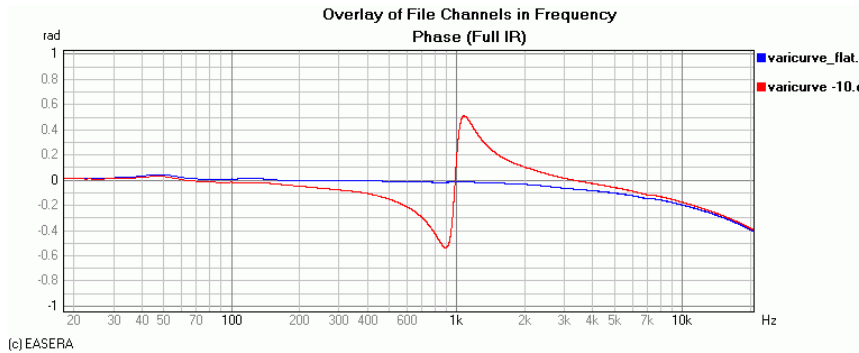


Now we want to check how accurately the center frequency is selected. To do this we zoom tightly around the notch:

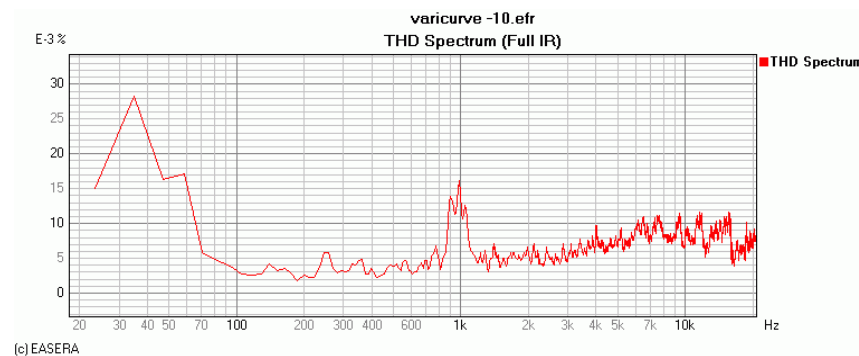


This is excellent for an analog (even though digitally controlled) parametric equalizer.

The next glance is to the phase response shown here as an overlay for comparison with the phase response of the linear setting. We see that an equalizer – and this is not just specific to this model – influences the phase response over a large part of the frequency range.



Now let us still have a look at the distortion factors:

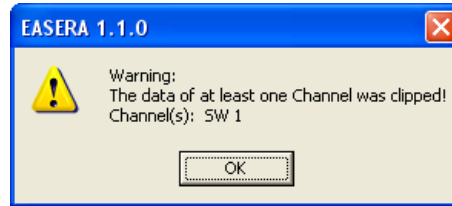


First we must be aware that the scale graduation of the Y-axis must be divided by 1000. The maximum harmonic distortion factor now indicated amounts to approx. 0.028%. For the measured values below 100 Hz caution is advised anyway. Harmonic distortion factors are multiples of the fundamental frequency. With a fundamental frequency of 1 kHz, for instance, H3 is approx. 3 kHz. If the highest harmonic distortion factor is at 35 Hz, the fundamental frequency for H2 would already be below 20 Hz. Since the linearity of soundcards is limited outside the audio range, one cannot define with certainty which portion of such a value should be attributed to the soundcard and which portion to the device under test.

A further rise of the harmonic distortion factor can be seen at 1 kHz. This is, however, not due to the filter distorting heavily at this point, but to the fact that the harmonic distortion factor is derived from the distortion products divided by the fundamental signal. If the fundamental signal is lowered by 10 dB, the unchanged distortion products result in a higher harmonic distortion factor.

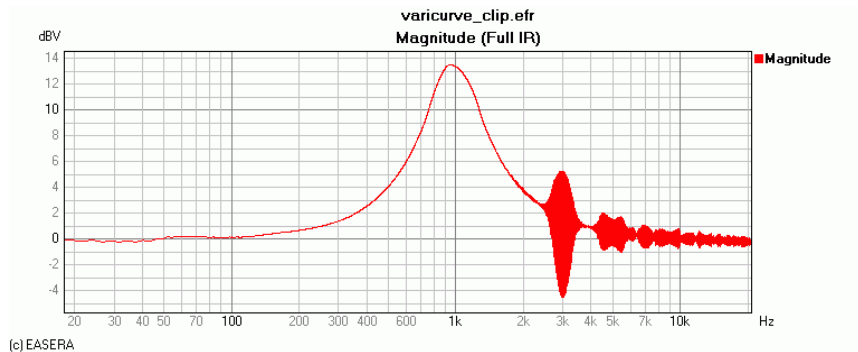
### 1 kHz full boost

For the next measurement we boost 1 kHz the maximum possible, which is +15 dB with the Varicurve. EASERA now warns us that we have run into clipping of the input:

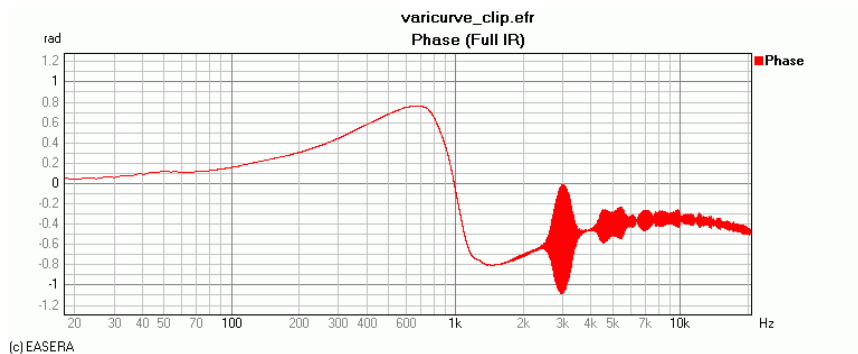


What we have measured now is not so much the behavior of the equalizer, but rather the behavior of a clipping and no longer linear AD converter. Nevertheless we shall now have a closer look at it. If it is not the AD converter, but another component of the transmission chain that is clipping, the diagrams would look similar, but EASERA could not issue any warning, because it is not the AD converter that is clipping.

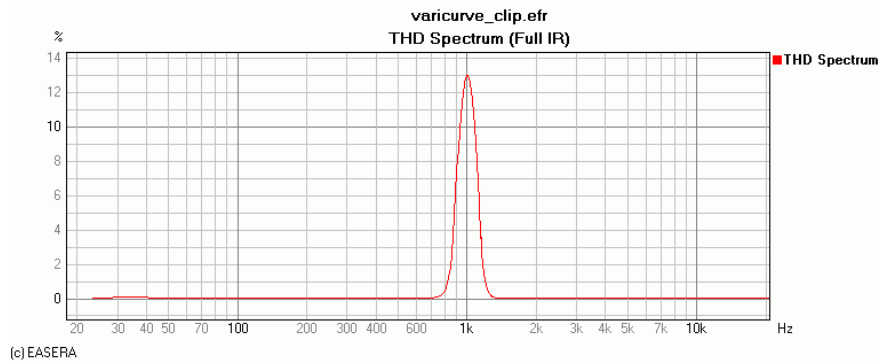
In such a case the user must recognize by the appearance of the diagrams that the signal has been clipped and EASERA is not able to produce an acceptable transfer function, because the measured device is working outside of the linear range.



The phase response also shows the characteristic patterns:



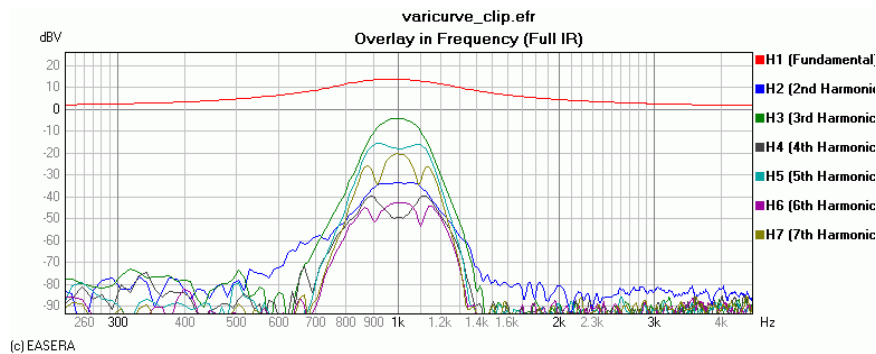
And the harmonic distortion factor rises sharply at 1 kHz:



At first glance it may be astonishing that the harmonic distortion factor is so high particularly at 1 kHz and not at those frequencies at which the artifacts are visible in the frequency and phase responses. Let us realize once more the facts in their context: If the signal gets clipped at a frequency, harmonics are formed, especially the odd-numbered ones. Thus an overmodulation at 1 kHz generates harmonics at 3 kHz (H3), 5 kHz (H5), and 7 kHz (H7) and so on. (When the signal gets distorted asymmetrically – e.g. because the positive supply voltage of a circuit is higher than the negative one –, there are also even-numbered harmonics produced, i.e. H2, H4, H6...)

We have here a narrow-band rise at 1 kHz. This originates the harmonics so that the harmonic distortion factor increases massively here and only here. The harmonics which we can discern in the frequency and phase responses, however, are not at 1kHz, but at the odd-number multiples of it, i.e. at 3 kHz, 5 kHz and so on.

In the spectra this would look like the following:



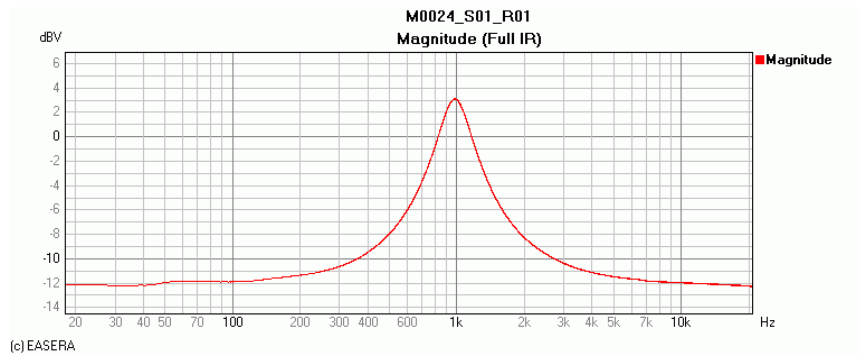
As can be seen pretty well, the odd-numbered distortions are definitely stronger than the even-numbered ones. This is why there are no nonlinearity effects found at 2 kHz in the frequency response.

The lesson we draw from this is that if strange patterns appear in the frequency and phase responses, these are presumably caused by clipping in the signal chain.

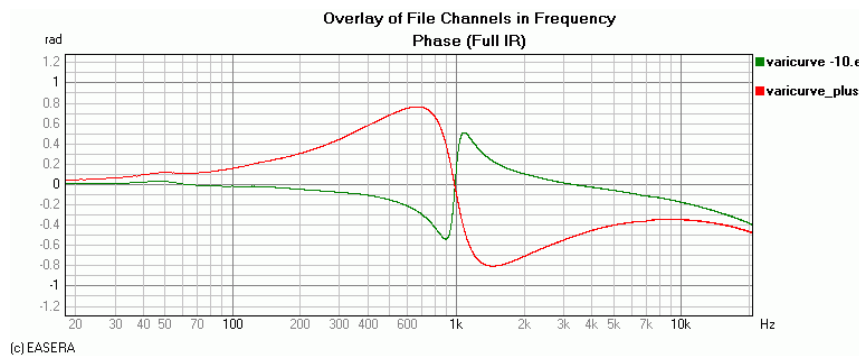
## Gain at -12 dB

Fortunately, most equalizers have a gain fader that makes it possible to avoid such clipping. The control range reaches for the Varicurve only goes down to -12 dB which is, however, sufficient for avoiding clipping in this case.

In the frequency response one notes that for the frequencies not influenced by the filter the level is at  $-12$  dBV. But where a boost of 15 dB was applied, it is at  $+3$  dBV.

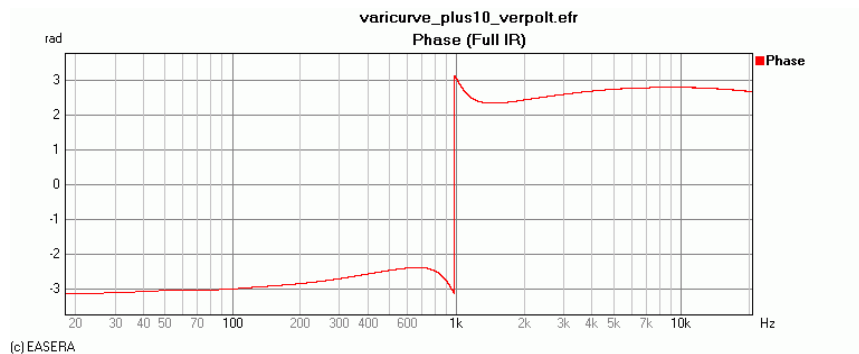


Let us now investigate the corresponding phase response in comparison with the 10 dB cut:



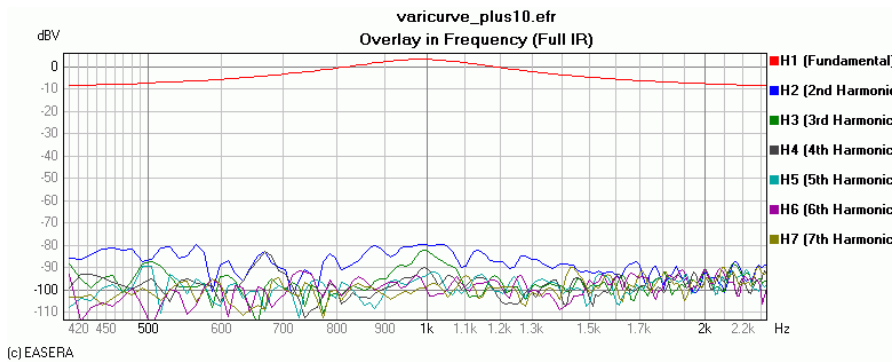
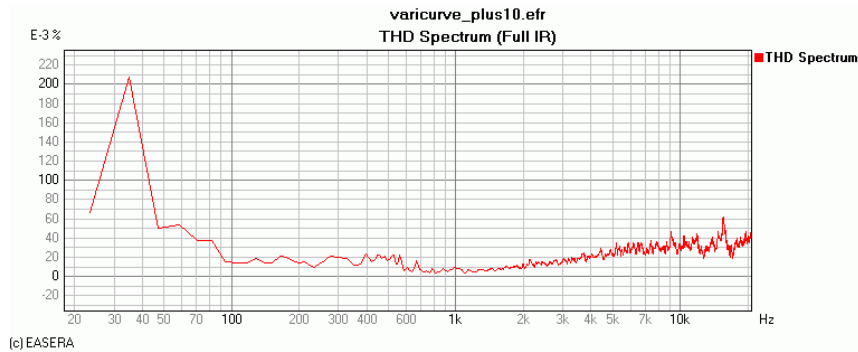
From the qualitative behavior of the phase response one can then determine whether a boost or a cut is shown: With a boost the phase response crosses the zero line at the mid-band frequency from top downward and with a cut from bottom upward.

This characteristic is also maintained with a polarity inversion, the only difference being that the zero line is no longer involved, but the  $180^\circ$ -line (i.e. 3.14 rad) is crossed instead.



If we once more compare the phase responses with boost and cut, we note that the boost influences the phase response to a greater extent. However, this is only due to the fact that we boosted by 15 dB and cut by only 10 dB. We conclude: the greater the boost or cut of a filter, the greater the phase response will be influenced.

And finally a glance at the harmonic distortion factors: A boost of 15 dB naturally does not reduce the harmonic components. The spectrum shows that H3 rises e.g. by about 10 dB, whereas the fundamental wave rises by 15 dB (which is precisely the boost we made) so that the ratio between distortion and fundamental wave becomes more favorable and the harmonic distortion factor is even slightly reduced.



# Lesson 16: System Calibration



## Preliminary Remarks

The greater part of measurements in room acoustics is independent of the absolute level value – the reverberation time does not change, for instance, if the signal decays from 80 dB instead of from 110 dB. Calibration of the system is, therefore, of secondary importance and one may perform measurements in EASERA instantly without previous calibration.

In case one wants to carry out special measurements, e.g. to calculate the STI with masking or to measure a loudspeaker, the absolute levels are important indeed and calibration of the input and output cannot be avoided. A complete calibration may take several minutes, but it must be carried out no more than once for a given hardware configuration.

### What is a calibration?

EASERA receives its data from the soundcard as digital values (e.g. from -32768 to 32767). Without calibration it would not be clear which digital value corresponds to which voltage or to which sound pressure at the input. The same applies to the output.

To calibrate an input, a constant signal is applied where the magnitude is known or was measured by the user and is entered in EASERA which then determines the digital value of this signal and assigns an actual physical quantity to it. The physical quantity of all other digital values can then be derived from it. (For those interested: EASERA stores the voltage quantity for *full scale* i.e. the full digital level).

To calibrate the output EASERA emits a constant signal and the voltage is measured by the user and entered in EASERA.

EASERA is able to half calibrate itself if the input is connected to the output. If the input is already calibrated, one can calibrate the output by it, if the output is already calibrated, one can calibrate the input by it. Such autocalibrations are performed avoiding the rounding and reading inaccuracies of external measurement devices and are therefore very precise.

To calibrate a microphone it is exposed to a defined sound level (there exist suitable devices called sound level calibrators or pistonphones). The sound level of this device is entered in EASERA so that the input signal can be assigned to a sound level to derive all other values from it.

### Calibrating in View & Calc

Some users may be irritated by the fact that on the *View & Calc* page there are different values shown than on the *Live* page. This is because on the *Live* page the actual applied input quantities are shown (Volt or Pascal or a corresponding level in dB), whereas on the *View & Calc* page we see a transfer function normalized to an output voltage of 1 V.

This means that a response function is shown that under the assumption of a linear system behavior would result if an output voltage of 1 V was used. If such an output voltage of 1 V is used, there is no difference from the *Live* page. But if the output voltage were only 0.1 V, all levels would be 20 dB higher. (We remark in passing that the normalization refers to 1 V peak and is thus independent of the signal.)

## Calibrating the mixers

Some soundcards use Windows mixers which enable the sensitivity of the soundcard to be varied using faders or gain controls. An alteration of this mixer setting for the input or the output causes the corresponding calibration to become invalid. (The same applies to an alteration of the gain of an external amplifier).

But in EASERA there also exists the possibility to calibrate the Windows mixers by determining which fader position corresponds to which gain. Alterations of the mixer settings can then not only be recognized by EASERA, but also assessed and deducted from the result. With calibrated Windows mixers the calibration in EASERA also remains intact if the settings of these mixers are changed.

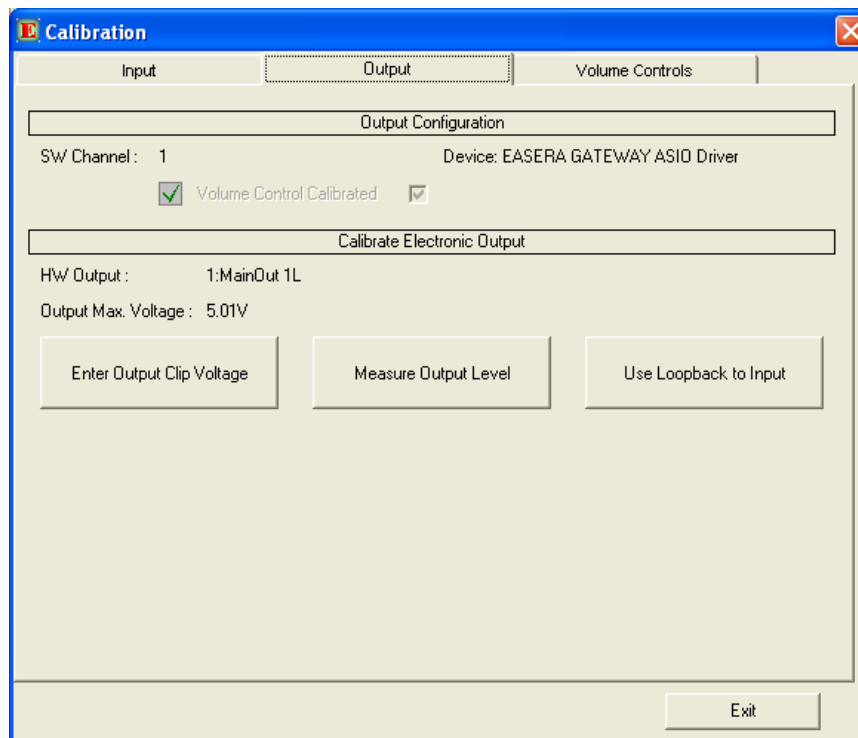
## Complete system calibration

EASERA is able to partly calibrate itself. As soon as an output is calibrated one may perform a calibration of the input using a loopback – a connection between output and input. Vice versa it is also possible to calibrate an output with a calibrated input.

Usually a voltmeter is more readily available than a signal generator producing a defined output signal. Typically it is easiest to first calibrate an output and we are now going to proceed to do that.

## Calibrating the output

To do this we select the *Measure* page and go to the *Select Measurement Setup* area. Click on one of the two *Calibrate* buttons to open the *Calibration* window:



We are now interested in the *Output* tab. On principle we have three possibilities to calibrate the output:

- Under *Enter Output Clip Voltage* we enter the maximum output voltage, if it is known. This voltage is sometimes specified by the manufacturer in the documentation. If the soundcard is equipped with faders, this specification normally refers to gain controls set to their maximum position.
- The output voltage can be measured using *Measure Output Level* and then entered.
- If the input is already calibrated, it is possible to use it to calibrate the output using *Use Loopback to Input*.

We now want to measure the output voltage and so click on *Measure Output Level*:

Click on *Start* to start emitting the signal (a pure sine wave) and click once more on this button to stop the sound. Enter the voltage measured meanwhile in the *Enter Voltage (RMS) measured at Output* text box. Please keep in mind that EASERA expects a RMS value. Then click on *Apply* to accept the value entered. EASERA then informs you via a message box about which clip voltage was calculated using that value.

## Calibrating the input

Now we want to calibrate the input. To do this we make sure that the phantom power (if available) is switched off and then connect the output to the input. Then we change to the *Input* tab.

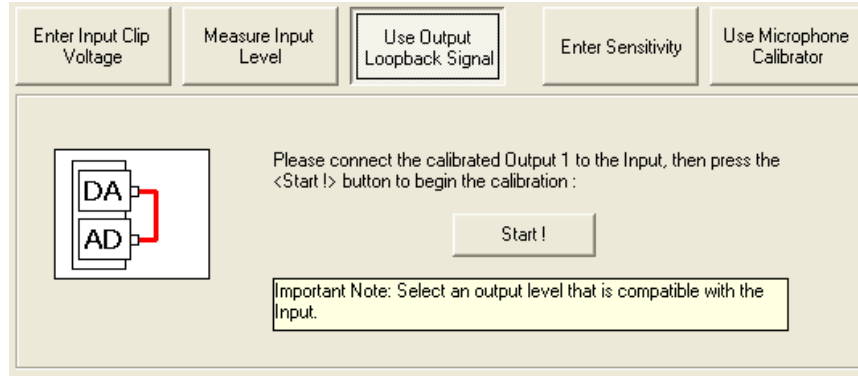
On the one hand it is possible to calibrate the input electrically and on the other hand the microphone used can be acoustically calibrated.

### **Electrical calibration**

We start with the electrical calibration. Here we also have three options:

- From the manufacturer's specification one can take the clip voltage of the input and then enter it into the *Enter Input Clip Voltage* text box.
- If a known input signal is available (the voltage of the signal generator must be measured in this case) the system can be calibrated using *Measure Input Level*.
- If the output is already calibrated it is possible to use the *Use Output Loopback Signal* button.

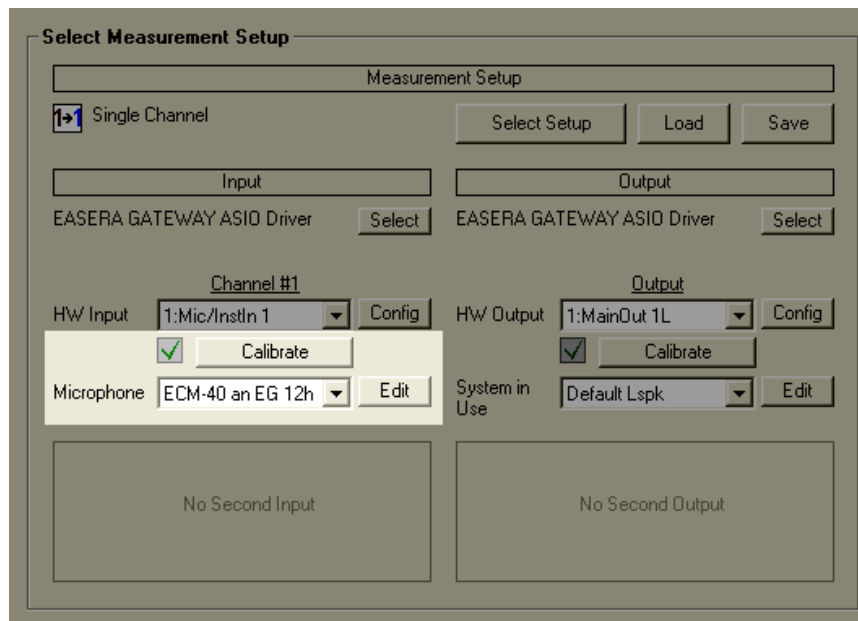
We have just calibrated the output and now employ the loopback method:



After connecting the input to the output you begin the calibration by clicking on the *Start* button and then wait a few seconds (until the hour glass disappears again). The input is now electrically calibrated.

### **Acoustical calibration**

Usually one works with several different measuring microphones to which individual calibrations can be assigned. We want to install a new microphone and first close the *Calibration* dialog box. Under *Select Measurement Setup* there is a *Microphone* section where we click on the *Edit* button.



Click on the *New* button first to create a new microphone, then immediately enter the *Name* into the corresponding text box. *Manufacturer* and *Directionality Characteristics* are optional details. Finally we confirm these entries with *Apply Changes* and close the dialog with *Ok*.

Afterwards we once more open the *Calibration* dialog box. To calibrate a microphone there are two options possible:

- If a sound level calibrator is available we choose *Use Microphone Calibrator*.
- If this is not the case we enter the sensitivity specified by the manufacturer in the *Enter Sensitivity* text box.

We want to use a calibrator here. To do this we connect the measuring microphone with the input (usually phantom power is now necessary) and click on the corresponding button:

Select or enter the sound level of the calibrator in the *Enter Level of Calibrator Signal* box and then connect the calibrator to the microphone and switch the calibrator on. After that click on *Start*, wait until the signal is stable and click on *Stop*. EASERA then informs you about the microphone sensitivity and the clip voltage and asks you whether these values are to be accepted. Complete the microphone calibration by confirming the new values.

At this juncture we would like to point out that the calibrations are, of course, to be carried out with those input and output gains which will be used for the subsequent measurements. But if one uses a calibrator with a high output level (e.g. 114 dB SPL), one can often avoid clipping at the input only by decreasing the input sensitivity. The resulting high input clip voltage, however, is then possibly unfavorable for the following measurement.

This problem can be dealt with by various methods. One way is to define the input sensitivity enhancement required for the measurement. With the EASERA Gateway it is possible to switch on a fixed additional gain of +12 dB. (This gain should then be entered for the input channel involved as *External*: +12 dB.) Another way is if the Windows input mixer has already been calibrated, then this calibration also remains intact if one increases the gain via this path.

If both options are not viable, it is also possible to use the calibrator to determine the microphone sensitivity and to determine the clip voltages occurring with the measuring gains by other methods, e.g. using a voltmeter or a signal generator. You can also use the loopback from the output to the input and compare the levels on the *Live* page using a sine wave stimulus before and after adjusting the control, then enter this difference for the input channel involved as a positive value in the *External* text box.

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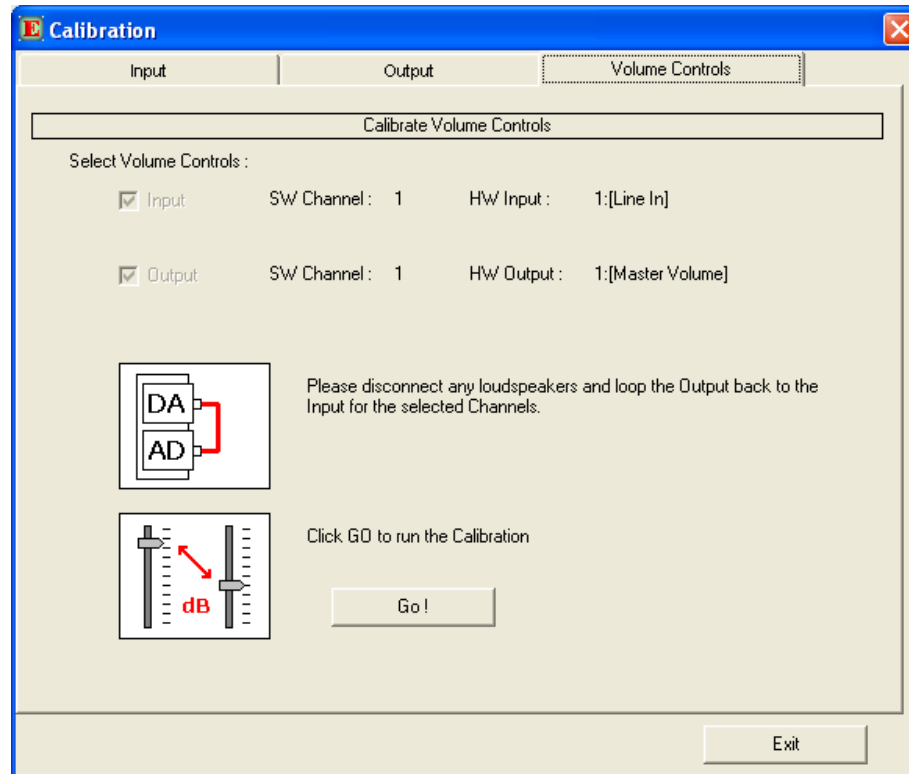
## Calibration of the Windows mixers

The EASERA GATEWAY and the AUBION X.8 are equipped with an ASIO driver recommended for the measurements. For this driver there are no internal windows mixers available. But if one uses for whatever reason the Wave or Direct Sound driver, the volume for the output can be adjusted via the associated Windows mixer. But EASERA also cooperates, of course, with all other Windows soundcards which usually have input and output mixers, i.e. faders for the sensitivity of the inputs and outputs, at least for the Wave and Direct Sound drivers.

The simple calibration of the system, as performed above, can be maintained by EASERA only until an input or output sensitivity (clipping voltage) is changed from outside. Such a change can be caused, for instance, by the gain controls of the device or by adjustment of the mixer faders.

In case of adjustment of the external faders or preamplifiers it is always necessary to enter the corresponding gain, if known, under *External*. If this gain is not sufficiently well known (e.g. with continuous faders), the associated clipping voltage must be determined anew.

In contrast, the internal Windows faders can be calibrated, and then the input and output sensitivity can be adapted to the measuring conditions without the system losing its calibrated state.



To perform the calibration connect the output to the input, click on the *Go* button and have a little patience. The calibration of the mixer settings may take several minutes. The duration depends on how many positions are adjustable at the input and output faders and the quality of the soundcard (With above-average large distortion or noise components, EASERA needs more time.)

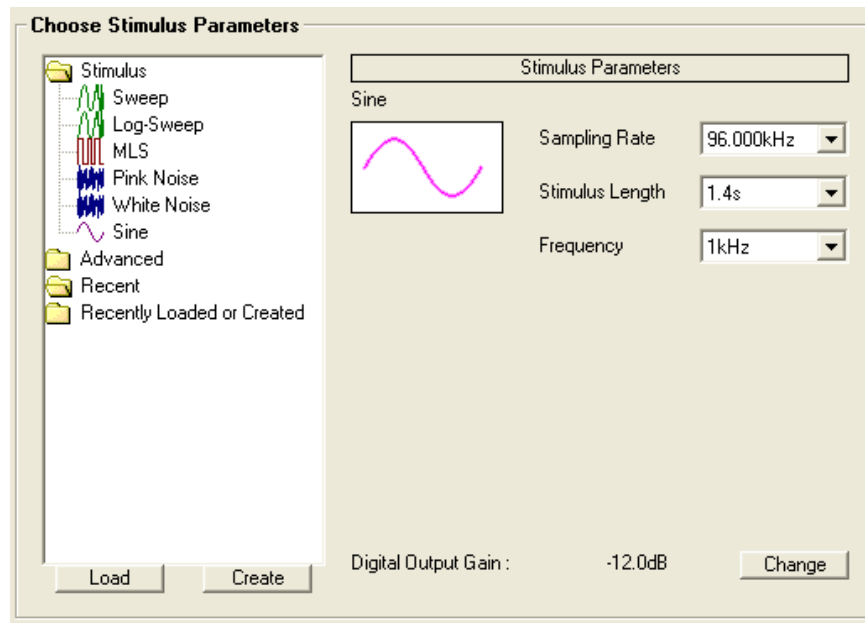
## Calibration to the absolute measuring level

Once the measuring arrangement is calibrated, the transmission properties of the system to be tested can be determined independently of the strength of the measuring signal used. This is possible because EASERA is now fully informed about the data of the whole measuring chain (from the output through to the input) and thus able to normalize the measurement to 1 V.

If the inputs and outputs were electrically calibrated correctly, one may perform a loopback measurement as a check. With the calibration frequency of 1 kHz the frequency response should show a value of 1 V (or 0 dBV).

If these prerequisites are given, we may now establish a relationship between the level readings in the *Live* section and the already normalized transfer functions in *View & Calc*. To do this we will adjust the output in such a way that the transfer function can be accurately repeated using a sinusoidal signal at a defined frequency under *Live*. To achieve this congruence it is necessary for the output voltage to be 1 V (rms). We are now going to make the required adjustments (whereas the original calibration of the output will be lost).

To do this we first go to the *Measure* page and then to *Choose Stimulus Parameters* where we pick a sine wave for the frequency of interest, e.g. 1 kHz. The *Digital Output Gain* should be about  $-12$  dB.



Now we switch on the output signal using the *Play Test Signal* button in the *Adjust Levels* area. With a sufficiently accurate measuring instrument we then measure the signal voltage at the output and adjust the output control so that the voltage indicated is exactly 1 V.

Since the EASERA Gateway has stepped gain controls, it will usually not be possible to adjust the desired voltage accurately. But the deviations resulting from this are usually of the order of inaccuracy of the measuring instrument and therefore negligible.

With this specially adjusted output voltage the levels measured in *Live* conform with the levels of the frequency response in *View & Calc*.

## Deliberate miscalibration

There exist situations in which an output voltage of 1 V is not practical. Some notebooks are for instance not capable of providing this voltage sufficiently distortion-free at the earphone output.

If the transfer functions in *View & Calc* are to be normalized to a different voltage than 1 V, one can accomplish that with a deliberate miscalibration:

The procedure is analogous to a calibration of the output, as was described under *Complete System Calibration*. To do this one emits a signal of e.g.  $-14$  dB and claims – independently of reality – that this was a signal of 1 V. Each time now when a signal of  $-14$  dB is emitted (this can be set under *Choose Stimulus Parameters*), EASERA assumes an output level of 1 V and ignores the necessity of normalization.

Alternatively the system may also have been calibrated absolutely correctly, as described at the beginning. The reference of the transfer functions in *View & Calc* to another output voltage may then be simply established by entering the corresponding gain in the *External Gain* field before the measurement (To do this the correction factor must of course have been measured or be known.).

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