

1 Case study: Secondary Microphone Calibration by substitution method

This chapter briefly describes the steps necessary to perform a microphone calibration based on the comparison of an unknown microphone to a reference microphone with well-known sensitivity and frequency response. Basically, this calibration is simply done by dividing the frequency response of a loudspeaker measured first with the reference microphone and then with the MUT (microphone under test) at exactly the same position. To obtain a sufficient signal-to-noise ratio at all frequencies of interest, it is required that the loudspeaker emits a reasonably flat test signal. Very often, the loudspeaker will not fulfill this requirement, especially in the range below, say 100 Hz. In this region, environmental noise also becomes a problem, as the insulation of the test chamber's walls decrease and hence, background noise increases. So it is a good idea to use a pre-emphasized test signal that not only compensates the transmission characteristics of the loudspeaker, but even gives an additional boost in the critical low frequency region. This will improve the measurement certainty considerably.

Although this case study deals specifically with microphone calibration, the technique to create a custom-tailored sweep signal described here is applicable in many more cases. So reading the document could be interesting for anyone who pretends to perform acoustical transfer function measurements.

The microphone calibration requires the three following steps:

Measuring the loudspeaker with the reference microphone. Based on the obtained frequency response, creating an appropriate pre-emphasized excitation signal, be that an MLS, a 2^n noise signal or a sweep.

Using the new stimulus to measure the loudspeaker again with the reference mic. and store the result to disk

Replacing the reference microphone with the MUT and performing the measurement again, using the previous measurement as reference. This will yield the desired frequency response of the MUT, relative to the reference microphone.

The following detailed description of the proceedings required to perform the calibration are intended as an exemplary case study and assumes three things:

- 1) A two-way speaker (using both amplifier outputs of the measuring system as an active crossover) is used to establish the sound field for the microphones. Just to make the affair a bit more sophisticated.

- 2) A fine custom tailored 2-channel sweep with constant envelope shall be used as the excitation signal.
- 3) A Monkey Forest (TM) (C) Version newer then May 2000 is available. Only this latest release will feature all the required tools to make the creation of a pre-emphasized stimulus a real easy task. If this version did not come pre-installed with your operating system (look for the monkey icon on the task bar), visit one of the many international MF-Web servers, pick up one of the free Monkey Forest bonus disks attached to all mayor weekly magazines, or simply contact your nearest Monkey Forest dealer around the next corner.
- 4) You are able to navigate in a DOS environment. If resizing windows and shifting them around on the screen by mouse is all you ever managed to do with you PC, you will have a hard time following all the steps described later, requiring some “finger acrobatics”..
- 5) You already read the “Building excitation signals” chapter, as it contains in-depth information about the BEX menu. Not everything will be repeated here.

1.1 Building an excitation signal to measure the loudspeaker

So let’s start with the first step, the measurement of the loudspeaker. Its frequency response will be used afterwards to create a new excitation signal that shall make the response essentially flat. First, an appropriate excitation signal has to be selected for this measurement itself. If it is not yet available on disk, we can easily create one. To do this, please do the following:

In the frequency domain, enter into the *AD/DA, Frequency response* menu. This menu offers the easiest way to measure the response of the loudspeaker if absolute sensitivity (SPL in 1 meter distance when loudspeaker is fed with 1 W) is not of interest. But of course, you could also use the “LS sensitivity” measurement menu. The remaining steps are substantially the same.

Press ‘R’ to enter into the “**Reference and more**” menu. In this submenu, press “N” (“**Build new sig**”) to get to the *Build Excitation Signal* menu. You have now entered the main playground for excitation signal kiddys.

Enter into the “**Setup**” menu (upper line). You are already feeling dizzy from climbing down this submenu of the submenu of the submenu? Don’t worry, from this point, we will climb up again. But before, you have to do this:

From the preset list on the left, select “**Loudspeaker/Microphone**”. This will

establish reasonable default values in the Excitation Signal menu, including the choice of a sweep with constant envelope and 20 dB bass boost as the excitation signal.

Back in the “Build Excitation Signal” Menu, choose an **FFT degree** if the one you desire is different from 14 (giving an FFT length of 16384 and thus a frequency resolution of $44100 / 16384 = 2.69$ Hz, assuming 44.1 kHz sample rate).

Change the three last characters of the prefix of the “**Signal file name**” from “NEW” to something more meaningful. Based on personal experience, you might also want to change some of the adjustable parameters in the menu (window, start/stop margin, etc.). Feel free to do so. For a first trial, the default parameters will do. So press the <Enter> Key to actually create the sweep signal.

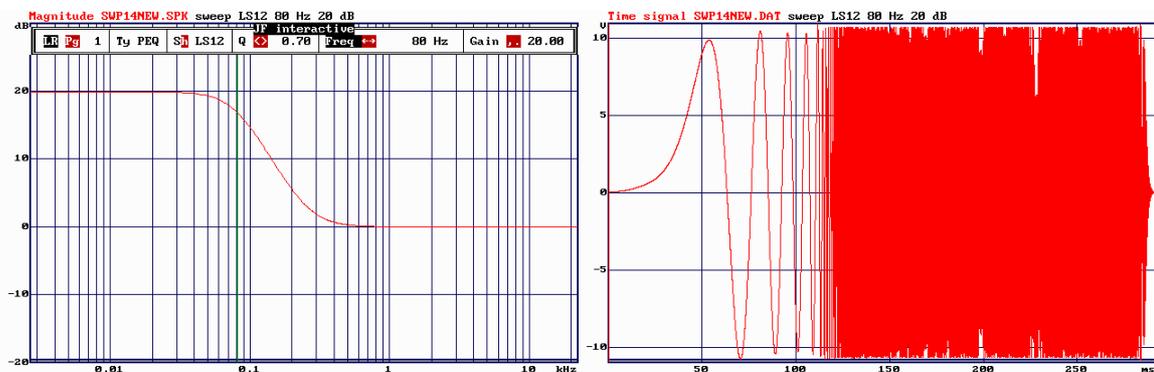


Figure 1: Constant envelope sweep from default set “loudspeaker/microphone” selected for measuring the source loudspeaker

1.2 Measuring the loudspeaker

Now, the new excitation sweep is stored on disk and also automatically selected as the active measurement stimulus. Before performing the loudspeaker measurement, we first have to pass through a reference measurement:

Back in the “**reference and more**” menu, press the “B” key to enter into the **basic settings** menu. Check if everything is ok here (correct hardware, indicated by a , desired sampling rate, AD input = the one where the microphones is connected). Set AD fullscale and DA fullscale to 1,0954 V when using the ITADDA16+Robo combination.

Press <ESC> or <Enter> to return to the “reference and more” menu. Set the “**DA level**” to -3 dB.

Now we are ready to perform the very important reference measurement. Generally, a reference measurement yields an equalization spectrum with which all subsequent measured spectra are multiplied. This is done in order to compensate at the same time the

spectral distribution of the excitation signal and the transmission characteristics of the measurement system itself. Thus, the equalization spectrum is simply obtained by letting the system measure itself and then inverting the resulting spectrum. To do so, the output of the measurement has to be connected to the input. If the ITADDA16 9,5" Robo box is your measurement frontend, this connection will be done automatically during the reference measurement. The reference measurement can be started by entering into the "Reference and more" menu and pressing <Ctrl> <Enter>. Check if the level of the signal was right by watching the level bar in the lower right corner. If the bar turns pink, an overload has occurred. If the bar leaves a lot of blue space on the right side, then the level was weak and the S/N ratio can be expected to be poor. Use the "AD fullscale" in the "Basic settings" menu to correct this. Also observe if the obtained reference signal is looking more or less like the inverse of the excitation signal, with no noise components visible. If the spectrum consists of a rugged, noisy trace, something went wrong in the reference measurement and you have to check out what it was. When the reference measurement is complete, the equalization spectrum has been stored to disk and can be used for the actual LS measurement:

Be sure that the reference microphone is connected correctly to the measurement system. Connect the woofer of the 2-way speaker to the left output of the measurement amplifier. Turn the volume to the 5 o'clock position.

In the "**Frequency response**" menu, press <Enter> to start the first measurement. Check the level of the loudspeaker signal by simply listening for audible distortion. If distortion is audible or you think your loudspeaker will not sustain this level in the long term, reduce volume by turning the "DA level" in the frequency response menu down. Now check the level of the input signal by looking at the level bar in the lower right corner. If it is too low, raise the external preamplifier gain, or, if the mic is connected directly to the ITADDA16, lower the "AD fullscale".

When the level relations are fine, you can optionally set a window, especially when you operate in a non-anecoic environment, to get rid of reflections and noise. To do so, switch over to the *impulse response* domain by pressing <Ctrl U>.

In the impulse domain, press the "E" key to see the entire impulse response.

Now press the "M" key. The active cursor will jump to the peak of the IR. This will be the middle of the symmetrical window to be applied later.

Press the "=" key to drag the passive cursor (the one that's red) to the same position.

Now move the active cursor to the position where you want the right edge of your

window. This will usually be a point just before the first major disturbing reflection.

Press the “W” key. A window titled “**the real windows**” (and it’s true) will appear, with the start and length values of the window preset according to the cursor positions. Don’t worry if the start value is negative: This means that the window start will be folded back to the end of the IR period. In other words: The window starts near the end of the IR, transits over the 0 s boundary and stops somewhere at the beginning. Remember that the IR is periodic, so “start” and “end” of the period are actually the same point. But be sure that “Slope : symmetrical” has been selected. If you do not want this, you can also select a right-half window. The cursors will then delineate the whole window region from top to the right edge, which means that their position is the same as in the symmetrical case. Remember that choosing a window too narrow will falsificate the response at the low end.

Change back to the magnitude response by pressing <Ctrl M>. Perform the measurement another time (the window is activated automatically when selecting it in the impulse response domain) and store the result to disk (by pressing the “S” key and typing a meaningful name). Now disconnect the woofer and connect the tweeter to the right output. Before performing the new measurement, contemplate if the tweeter will handle the output power delivered by the frontend. If in doubt, lower the “DA Level” in the “Frequency response” menu. This change will have no effect on the level of the resulting measured spectrum. Start the measurement by typing <Enter> and also store this result on disk. Remember not to change the position neither of the microphone nor of the speaker, as the delay and phase relation of the two measurements matters for establishing the right delay between the 2 channels of the sweep.

1.3 Joining the 2 ways to form a 2-channel spectrum

Now the measurement of the two ways of the loudspeaker is complete. What we would like to create now is an excitation signal that equalizes the dips and peaks in the frequency response. Remember that the method of pre-equalizing is not mandatory for the comparison measurement, but reduces to a great extent the uncertainty at frequency regions where the loudspeaker’s response is poor. Furthermore, we would like to introduce an active crossover function, with the woofer emitting a sweep up to the crossover frequency, from where on the tweeter will be fed with the rest of the sweep. Monkey Forest is able to take into account the phase and delay difference of the two ways at the crossover point. One of the two channels will be shifted in such a fashion that when transiting the crossover region, the two signals will arrive in-phase at the microphone position. This way, a sound pressure drop at the x-over frequency is avoided.

To prepare the construction of the excitation signal, we first have to load the two measurements previously made into one single spectrum with two channels, which of course is an easy task for the experienced Monkey Forest user. For the bloody beginner, here is how it works:

In the spectral domain, enter into the *File* menu by typing <Alt F>. Select the woofer file by moving the highlighting selection bar up or down, using the up/down cursor keys. When you hit the desired measurement file with the selection bar, press the <Enter> key and the file will be loaded as the new spectrum.

To load the tweeter file as the second channel, do NOT enter into the file menu again (an error all beginners commit, sigh...). Instead of this, enter into the *Edit / Read file* menu by typing <Alt E>, R. Select new channel, and in the file menu appearing, locate the selection bar to the measurement of the tweeter and hit the <Enter> key.

Store the new 2-channel file (simply strike ‘S’) using a new name.

We have now created the “target” for our equalization game. In the example presented here to illustrate the process (Figure 2), it’s a 2-way coaxial speaker with a very irregular frequency response, ailing from deep dips in the high frequency region and a far-too-early roll-off at low frequencies. But we will see that it’s even possible to use such a misplaced transducer by using the powerful weapon “constant envelope sweep with speaker equalization and crossover”.

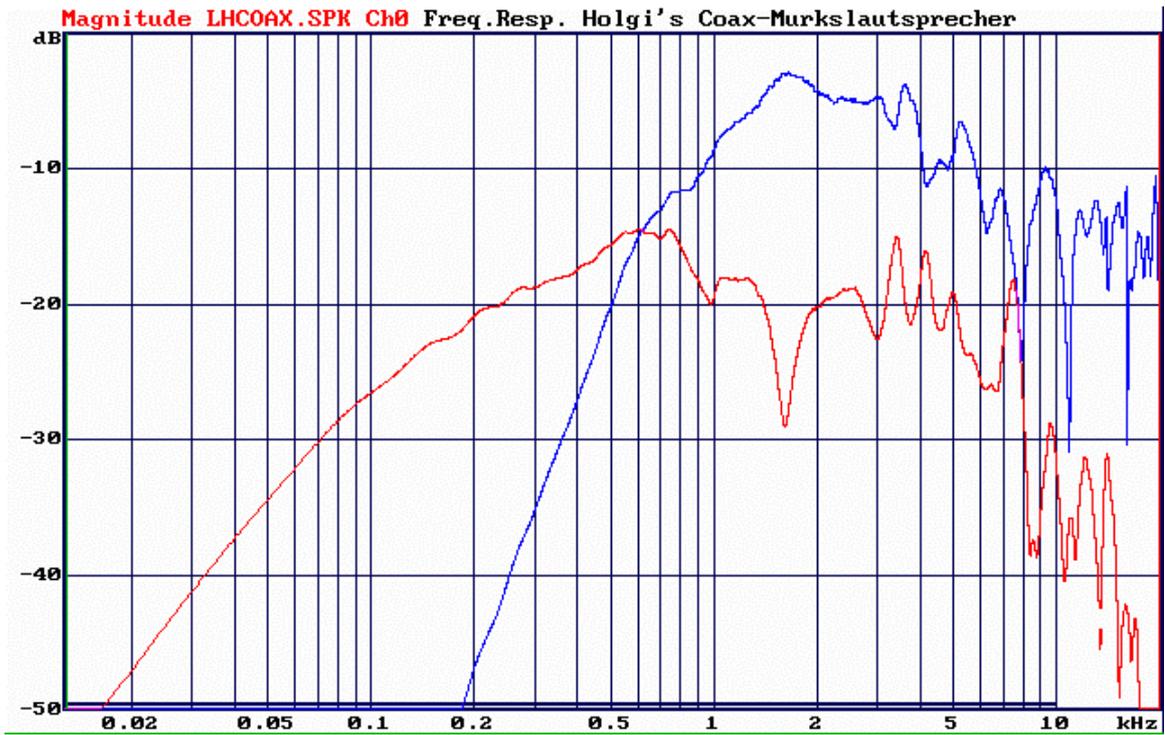


Figure 2: Frequency response of a mediocre 2-way coaxial speaker used as source for the microphone calibration

1.4 Building the excitation signal

Now we are ready to create the pre-emphasized two-channel-sweep with crossover functionality. To solemnly initiate this process, enter into the *Alt AD/DA, Excitation signal* menu. Don't feel afraid of the pile of entries appearing. We will handle them line by line.

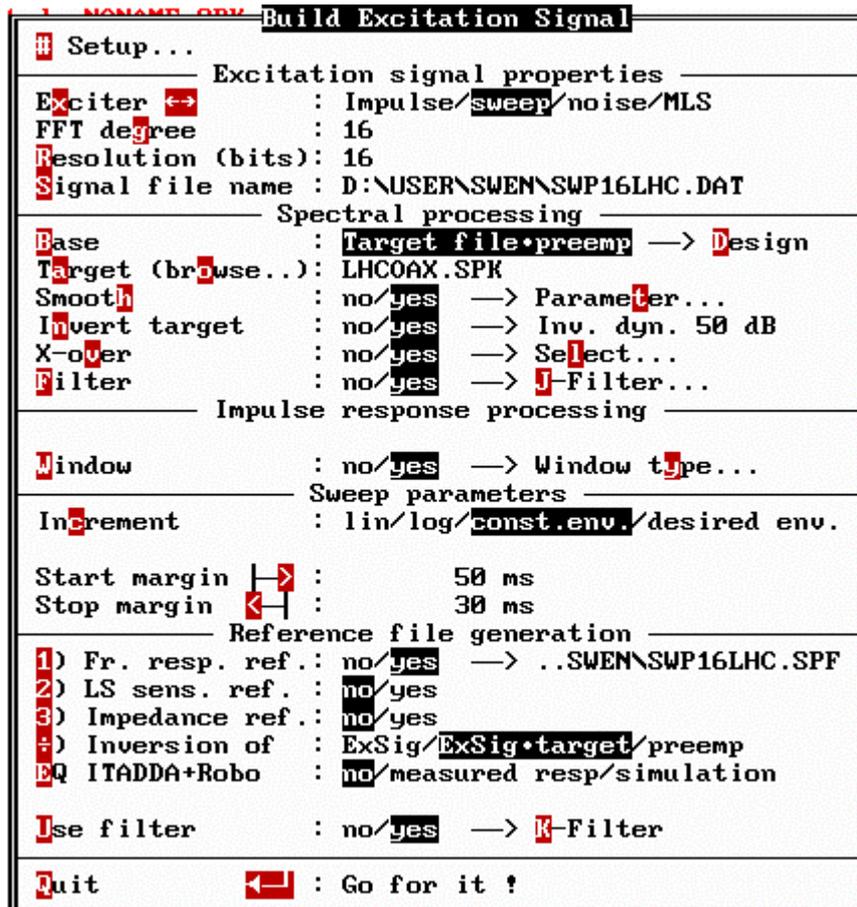


Figure 3: BEX menu with switches set for building a stereo sweep with constant envelope to equalize the 2-way speaker LHCOAX.

1.4.1 Excitation signal properties

In the “Excitation signal properties” section, the “exciter” has already been set to “sweep”. The Resolution normally is 16 bits and there’s little need to change it.

The “FFT degree” is a first important thing to consider thoroughly. As you know, the length of the excitation signal will be $2^{\text{FFT degree}}$ samples. Although the default FFT degree for loudspeaker/microphone measurements is 14, it is beneficent to use a degree that is somewhat higher. The reason for this is less to achieve a higher frequency resolution, but simply to pack more energy into the excitation signal and hence improve

the signal/noise ratio of the measurement. Another reason is that the spectral details of the target spectrum (our loudspeaker to be equalized) can be tracked more faithfully and with less overshoot in the time domain when creating a longer signal. On the other hand, extending the length clearly also extends the measurement time. So you have to compromise according to your needs. If the highest possible measurement certainty is your goal and you don't mind that emitting the sweep and calculating the subsequent FFTs will take many seconds, choose a high degree of up to, say 19 (corresponds to a sweep length of nearly 12 seconds at 44.1 kHz sampling rate). But if you are a poor student not really keen on perfect results and an old 486/33 is all they gave you to perform this task, nobody can blame you for using a shorter length. Restricted RAM installed in your computer might even force you to do so. By the way, the degree of the loudspeaker measurement you performed before does not necessarily have to correspond to the degree chosen here: The target response will be transformed to the time domain and extended with zeros if the excitation signal is longer than the impulse response. In the illustration example, we chose a degree of 16 which, under normal conditions, should yield reasonable measurement certainty while at the same time keeping the measurement length under 2 seconds on a modern machine.

Don't bother about the "**Signal file name**": It will be created automatically by choices to be made now in the "Spectral Processing" menu.

1.4.2 Spectral processing

Position the "**Base**" switch to "**Target file • preemp**". This simply means that we want to include a file (the loudspeaker measurement) in the target, and we would further like to give an additional pre-emphasis to improve the S/N ratio at low frequencies.

Select the **target file** (the 2-channel measurement of the speaker) by either entering its name or browsing for it if you forgot the exact name or location.

Generally, when equalizing loudspeakers, it's a good idea to **smooth** their response. Especially in our illustration case, the loudspeaker is so misbehaved that it's mandatory to do so. Otherwise, the resulting excitation signal would contain large energy peaks at the frequencies where the tweeter has deep dips. As these dips have their roots in cone break-up or destructive interference, feeding energy particularly at these frequencies would be counterproductive. Instead of trying to obtain sound pressure at frequencies where the speaker is trammled by its physics, we will simply accept that our measurement will lack signal-to-noise ratio in these very narrow frequency strips. We will later mask them by smoothing. The smoothing selected here to avoid peaks should be logarithmic and may range between, say 1/24 and 1/3 octave.

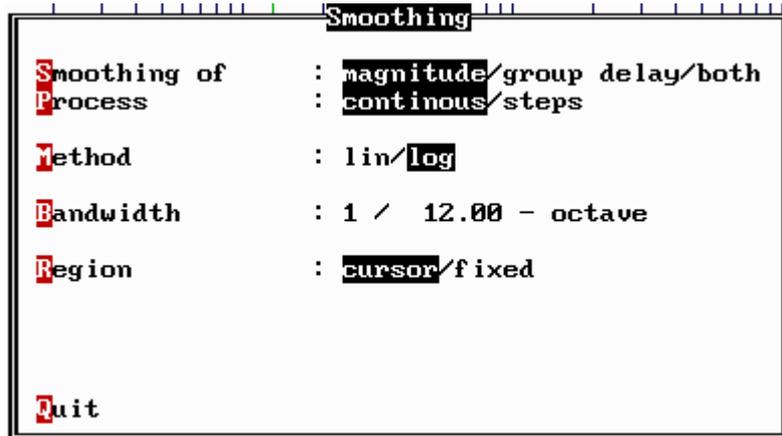


Figure 4: Parameter Select in the smooth “Parameter” submenu.

The “**invert target**” switch, designed especially to deal with loudspeaker spectra to be equalized, has to be set to “yes”, of course. As inverting results in very high magnitudes outside the transmission range of the speaker, the dynamic range of this operation is restricted to 50 dB. However, the inverted spectrum should still be confined by means of J-Filter.

But first, let’s adjust the crossover point. Before entering into the “**X-over select**” submenu, you might want to check the original 2-way frequency response for a good crossover region. You could then step into the submenu and enter this crossover frequency as a starting value.

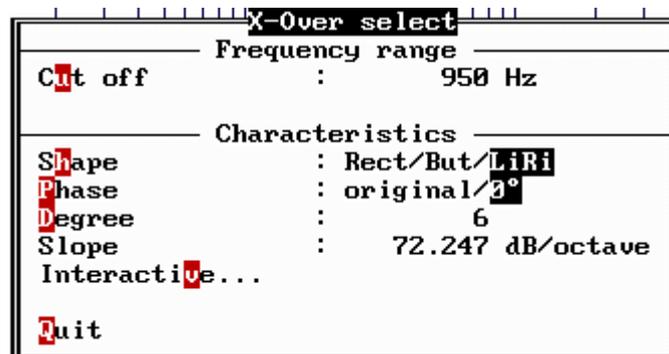


Figure 5: X-Over select submenu

To do the fine adjustment, enter into the “**interactive**” mode by striking the “V” key. The target spectrum will appear with all the steps activated in the “spectral processing” menu already involved. While all the other parameters (J-Filter, Preemp design, smoothing etc.) are applied, but not accessible here, you can modify the crossover parameters freely. The active cursor will be positioned at the crossover frequency. Use the left, right cursor keys to change the x-over point. If you press the left shift key at the same time, the steps will be finer. If, in contrast, you press the Ctrl key, you can move rapidly through the spectrum. You can also directly enter a value by simply typing it into the

keyboard. The value will be placed in the parameter least recently changed (highlighted with inverted letters in the control bar). The “<” and “>” keys can be used to adjust the selectivity of the crossover filter by changing the filter order. 10 is the maximum. The “gain” is functional, but it makes no sense to change it from 0 dB. Many of the commands you are used to for controlling the display are still functional. For example, use the Cursor Up/down keys to shift the whole image. If you press Shift at the same time, the dynamic range will be changed. Use “+” to zoom in, “-“ to zoom out. When you have finished adjusting the crossover frequency, simply strike <enter> or <ESC> to return to the BEX main menu.

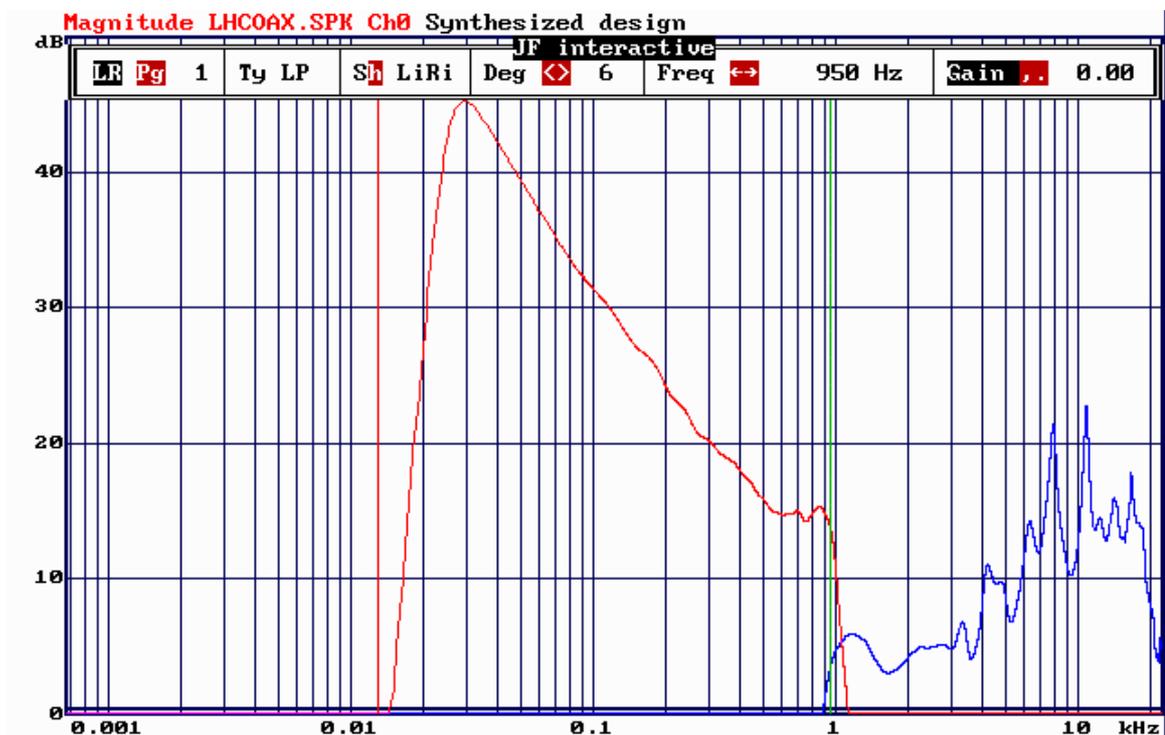


Figure 6: Screenshot from the interactive x-over adjustment procedure.

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Design preemphasis curve
----- Tilt -----
Slope      :      0 dB/octave
Lower cut off :      20 Hz
Upper cut off :      20 kHz
----- Filter characteristics -----
Filter No Pg↑↓ :      1 of 3
Shape Ctrl ↔ : Bell/LoSh6/LoSh12/HiSh6/HiSh12
Quality     :      0.70
Gain       :      5 dB
PEQ frequency :      200 Hz
----- Interactive design -----
InteractiVe...
View preemp only : no/yes
Quit

```

Figure 7: Pre-emphasis selected for coax speaker

Before confining the band edges, let's adjust the pre-emphasis now. To do this, hit the "D" key to access the "Design pre-emphasis curve" menu.

The "interactive" tool offers two visualization modes here: You can select to see the emphasis spectrum all alone ("view preemp only" = yes) or to see the product of target file with all subsequent processing steps (smooth, invert, x-over, J-Filter) multiplied by the emphasis curve. In the case of our exemplary rotten coax-chassis, there's already so little response at low frequencies that only a very moderate emphasis of 5 dB is chosen here. If your speaker has a good response below 100 Hz, you can use 20 or more dB of emphasis.

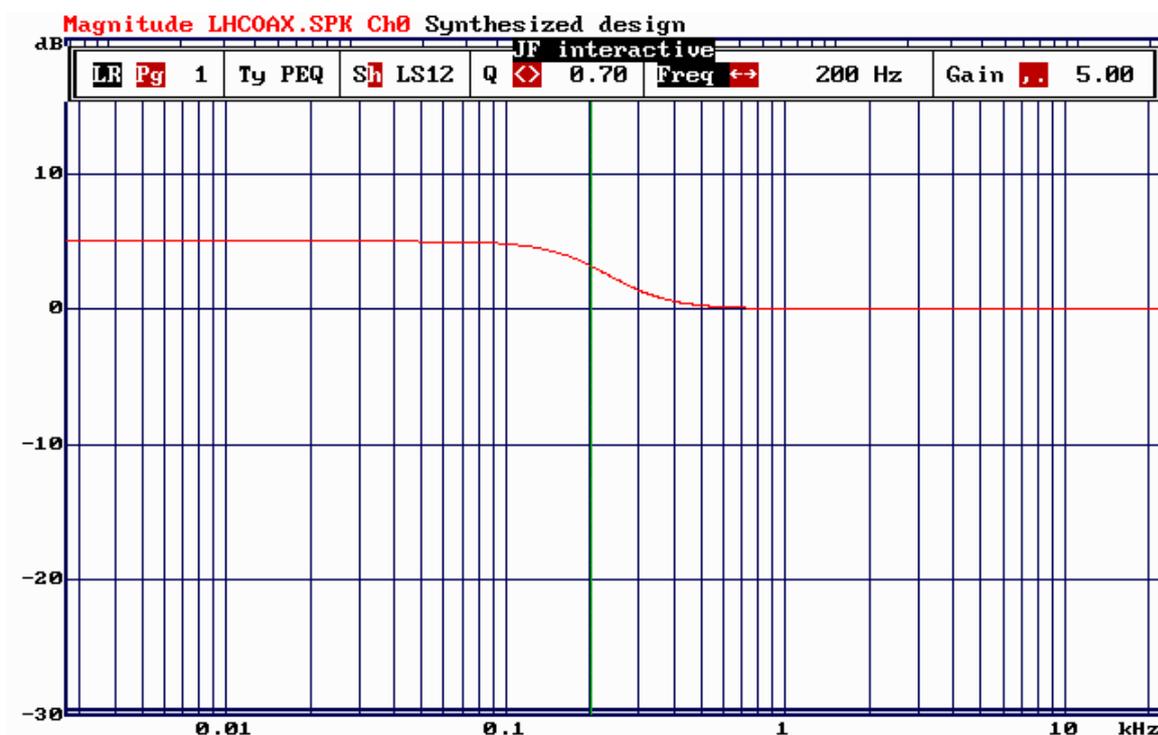


Figure 8: Moderate pre-emphasis of 5 dB for coax speaker.

And now it's time to confine the inverted target spectrum with the J-Filter band-pass. Actually, in the preceding pictures, an appropriate band-pass to inhibit excessive out-of-band level already has been applied. You can change the order of the adjustments or iterate in several passes until you find a satisfying combination of emphasis, cut-off frequencies and crossover point.

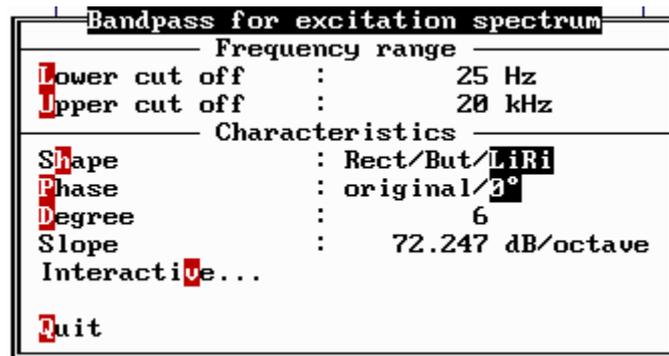


Figure 9: J-Filter submenu for confining the target frequency response

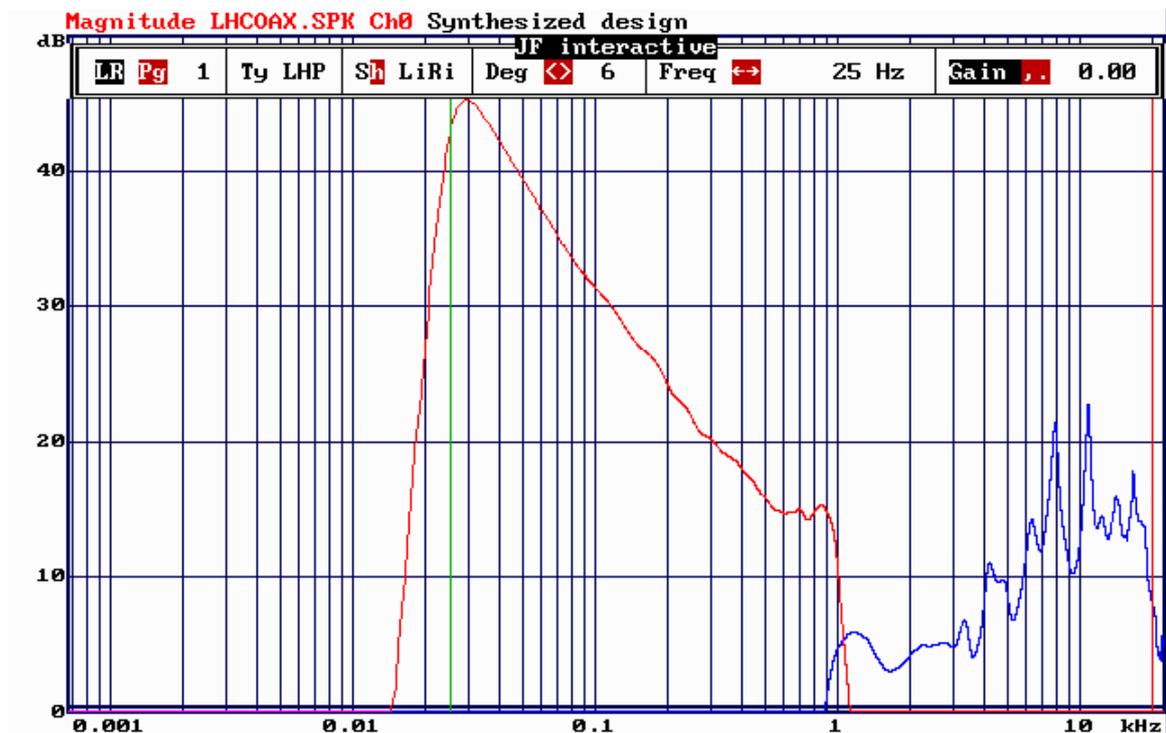


Figure 10: J-Filter band-pass to confine inverted target function

1.4.3 Impulse response processing

Before any excitation signal is created, the target's spectrum is first transformed to the time domain to impose a window. This windowing acts like a smoothing of constant absolute bandwidth. So on a logarithmic frequency scale, it affects the resolution of low frequencies first. The optimal window length depends on the lower excitation signal's cut off frequency. The lower it is, the larger the window has to be to avoid too much smoothing. Take your time to iterate several times the excitation signal erecting process and observe the influence of the window. In this example, 75 ms proved to be a good value. The window will be positioned automatically (you do not have to locate it to the peak position by yourself).

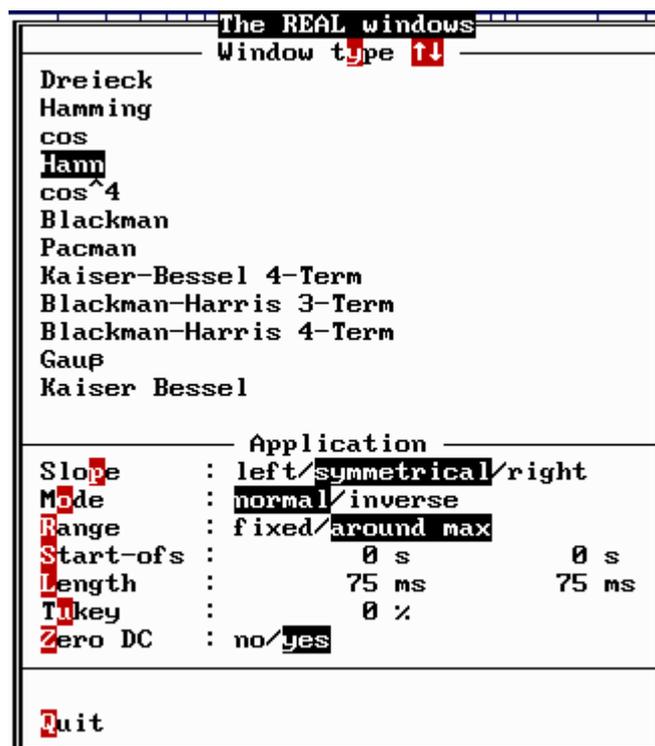


Figure 11: Window sub menu to confine the impulse response of the target spectrum

1.4.4 Sweep parameters

The **increment** of the sweep refers to the group delay course. The constant envelope setting will yield a time signal with an amplitude almost independent of time. However, two crucial parameters have to be adjusted here to produce beautiful results:

The **start margin** defines the of the group delay of the lowest frequency. It's necessary to start with a value $> 0s$. See the "Building excitation signals" chapter for further explanation. Unfortunately, it would be very difficult to set this entry automatically according to the excitation signal properties. So this task has to be performed by you, iterating a couple of times the building process and checking both the time signal and the resulting spectrum. If the spectrum's low end is very different from what you expected, try either increasing the FFT degree, the window size in the impulse response processing section or this start margin value. If, on the other hand, the time signal seems to evolve late on the time axis, reduce the value to allow more energy to be packed in the measurement interval.

The **stop margin** is less critical as it deals with the high frequency end. Practically the only constraint is that the stop margin has to be at least as high as the flying time of the sound plus the loudspeaker's decay at high frequencies (normally negligible) plus the delay introduced by the anti-alias filters. If the chosen value is too small, Monkey Forest will increment the FFT degree by one when measuring. This would result in a

measurement period with approximately only half of the signal being filled with the sweep.

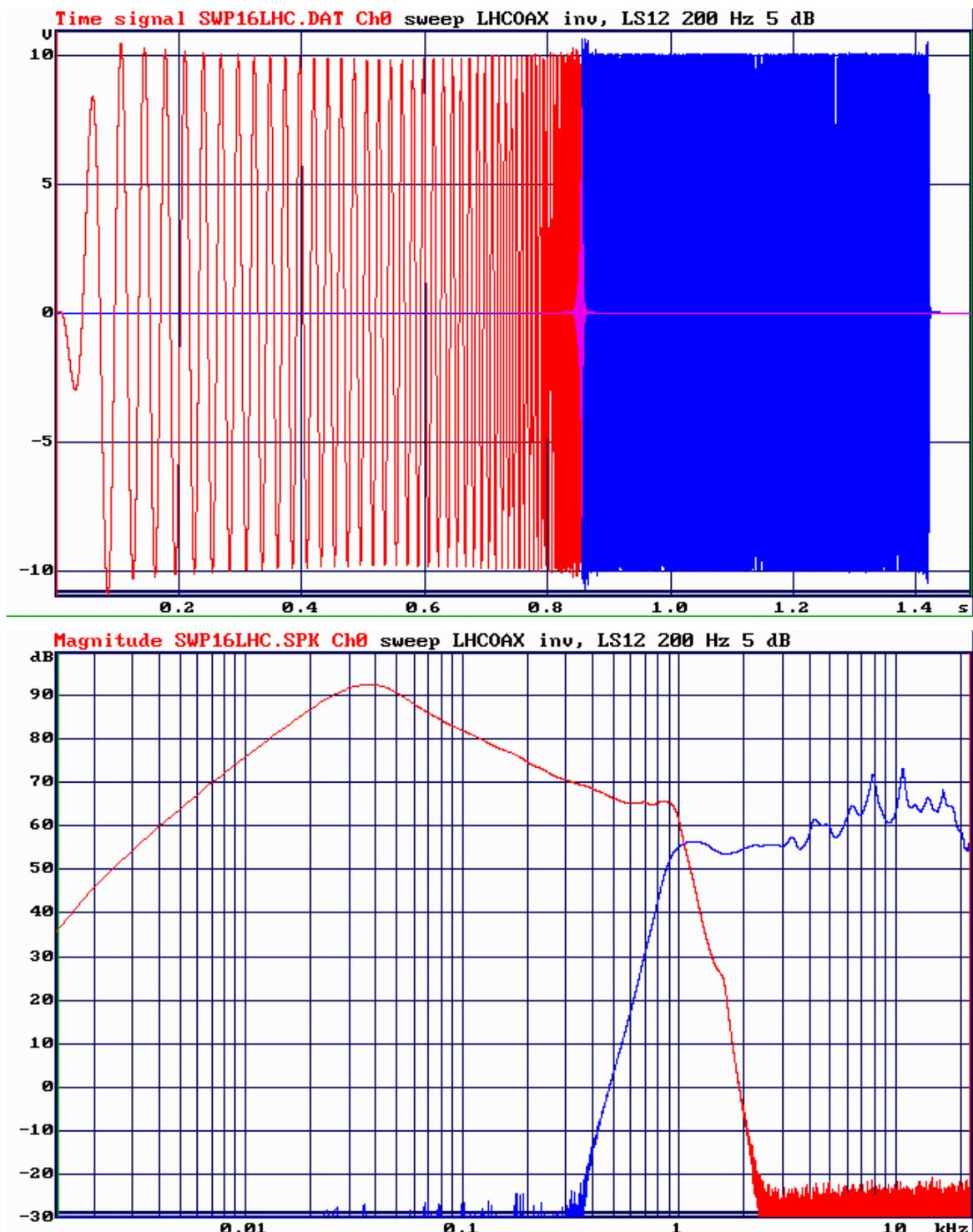


Figure 12: Dual-channel sweep to equalize and crossover the coax-speaker used in the example.

Principally, these are all the settings we need to create the 2-channel sweep with constant envelope, speaker equalization and crossover functionality. The result in Figure 12 shows

low overshoot in the time domain and at the same time high spectral purity without any “dirt effects” in the spectral domain. However, such a good result can only be expected after optimizing the parameters in several iterations. So here we see again: Monkey Forest is a powerful tool, but only freaks can tame it and squeeze some useful results out of it. Nothing is done automatically, everything has to be controlled by the user. But hold on, things get worse, we do not have finished yet.

1.4.5 Reference file generation

The reference file generation is a substitute for the reference measurement that normally should be performed to obtain accurate results. As we are dealing with a relative measurement here, meaning that the response error introduced by the measurement system will be reflected in both the measurement of the reference microphone and the MUT and hence disappear by the division, it is not crucial to create an ultra-precise reference. In the contrary, it is even advantageous to emulate the reference for a reason to be revealed soon.

Set the “**Frequency response reference**” switch to yes. Should already be in this position.

The “**Inversion of**” switch should be positioned to “**preemp**”. This will cause the magnitude of the reference spectrum to be simply the inverse of the emphasis filter, multiplied by the band-pass defined in K-filter. This reference will NOT equalize the deviation introduced by smoothing the target and windowing its impulse response.

It’s also possible to use the “**ExSig • target**” mode, but this would yield a reference file which forces all the remaining dips in the equalized speaker’s frequency response to be absolutely flat (remember that we used smoothing to avoid that the excitation signal equalizes very narrow dips). While at first glance this seems to be very nice, it has a serious drawback. This is a bit difficult to explain, so may be you have to read this paragraph twice. Here we go: We have accepted that the S/N ration in the narrow dips will be low. So it’s better not to correct them by means of the reference file (the latter would exhibit high peaks to compensate for the dips) in order to avoid that frequency magnitude values with high uncertainty are raised to a much higher level. Instead of this, it’s better to leave them untouched and apply smoothing to take advantage of the “smearing” from the neighboring frequency regions where the S/N is sufficient. In other words: By performing the smoothing, the neighboring magnitude values with high S/N will fill the narrow dip with reliable and stable information. And only after this smoothing, the MUT should later be divided by the reference microphone.

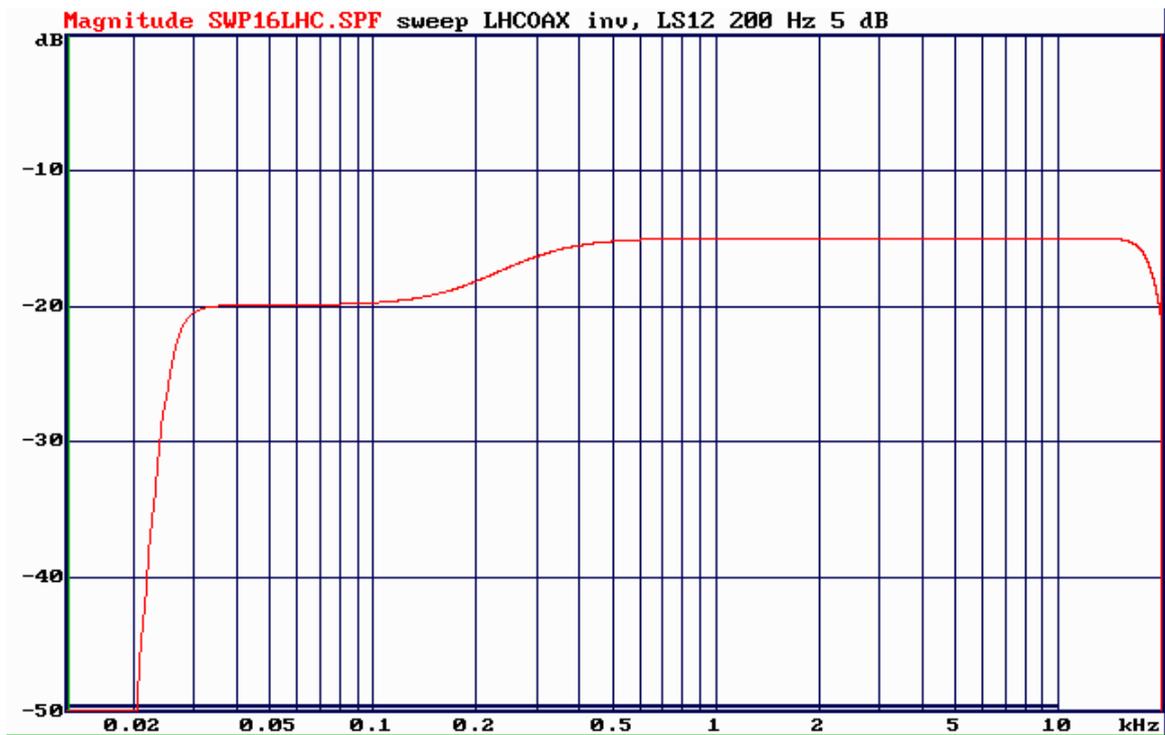


Figure 13: Reference file containing only the inverse emphasis and the confining band-pass

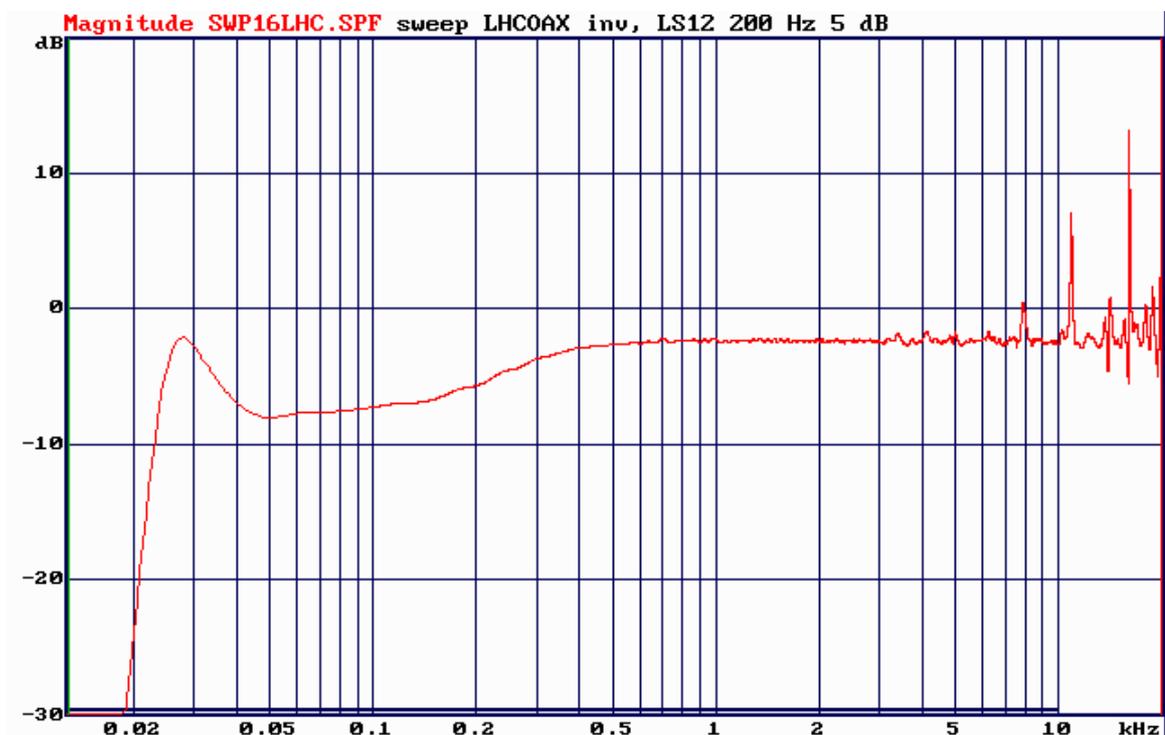


Figure 14: Reference file created by multiplying the excitation signal with the un-inverted target (loudspeaker) response.

The “**Use filter**” should be set to yes to confine the reference spectrum to the transmission range of the equalized loudspeaker. Normally, you do not have to worry about the filter settings in the K-Filter menu as they are copied from the J-Filter menu used to confine the excitation signal’s frequency response.

1.5 The comparison

Once we have passed through the extensive course of creating a well-done excitation signal (fortunately, you have to do this only one time when you chose the source speaker and erected your measurement setup), the comparison itself is fairly easy. First, we will measure the reference microphone with the new excitation signal. To do so, please

Enter into the *Alt AD/DA, frequency response* menu again.

The excitation signal kind, name and length should already have been set automatically by the last BEX session. So principally, after pressing <Enter>, the new sweep should become audible and shortly afterwards a frequency response similar to Figure 15 should appear on the screen. Check the connections and the AD level if it doesn’t.

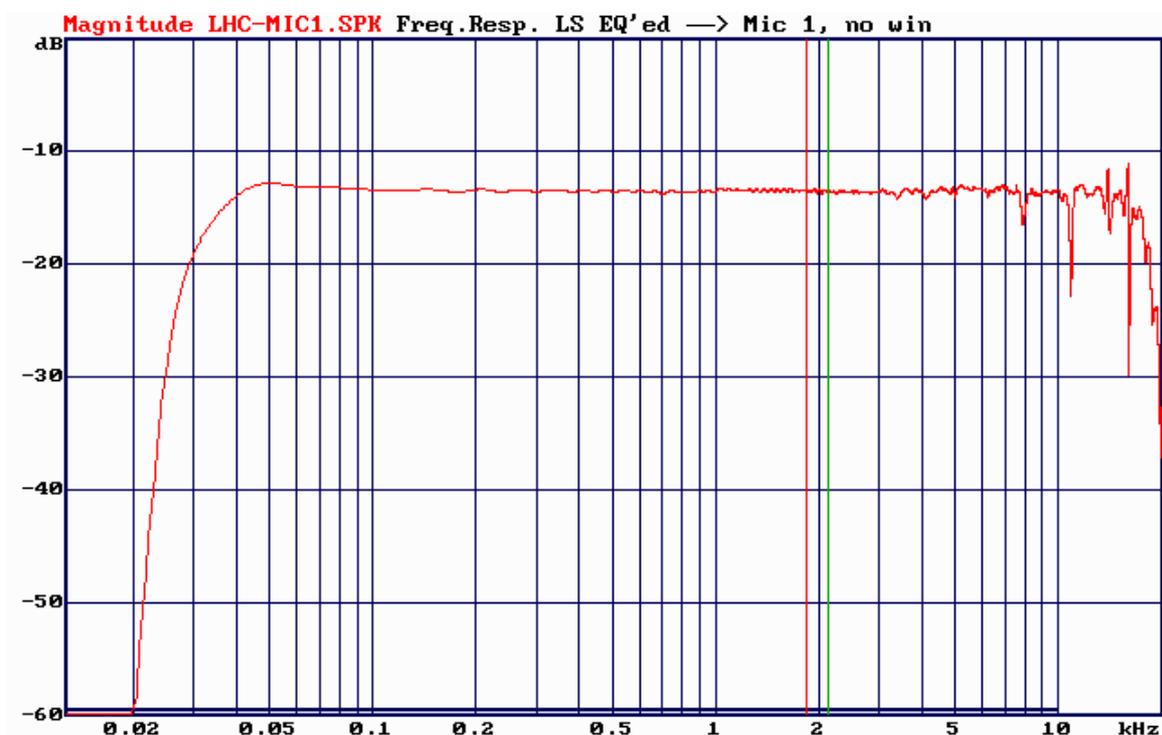


Figure 15: Measurement of the loudspeaker – microphone transfer path using the new excitation signal and reference spectrum

To increase the signal-to-noise ratio considerably and to establish simulated free field conditions if your measurement setup is not installed in an anechoic chamber, it’s a very

good idea to apply a window to the impulse response now. If you want to position the window automatically, switch “**window**” to yes and enter into “**the real windows**” submenu. Select a symmetrical **slope** and set the **range** to “**around max**”. The **length** can be something between 30 and 100 ms. In the case of measurements in a non-anecoic environment, you probably want to check visually where the first disturbing reflections appear. To do this, change to the *impulse response* domain by typing “U”. Do the window positioning as described in the second half of chapter 1.2. Now the measurement could look similar to Figure 16. Note that as usual, windowing primarily affects the resolution in the bass range. Be sure not to make the window too narrow, as this would draw excessive low frequency energy from the impulse response and the corresponding spectrum.

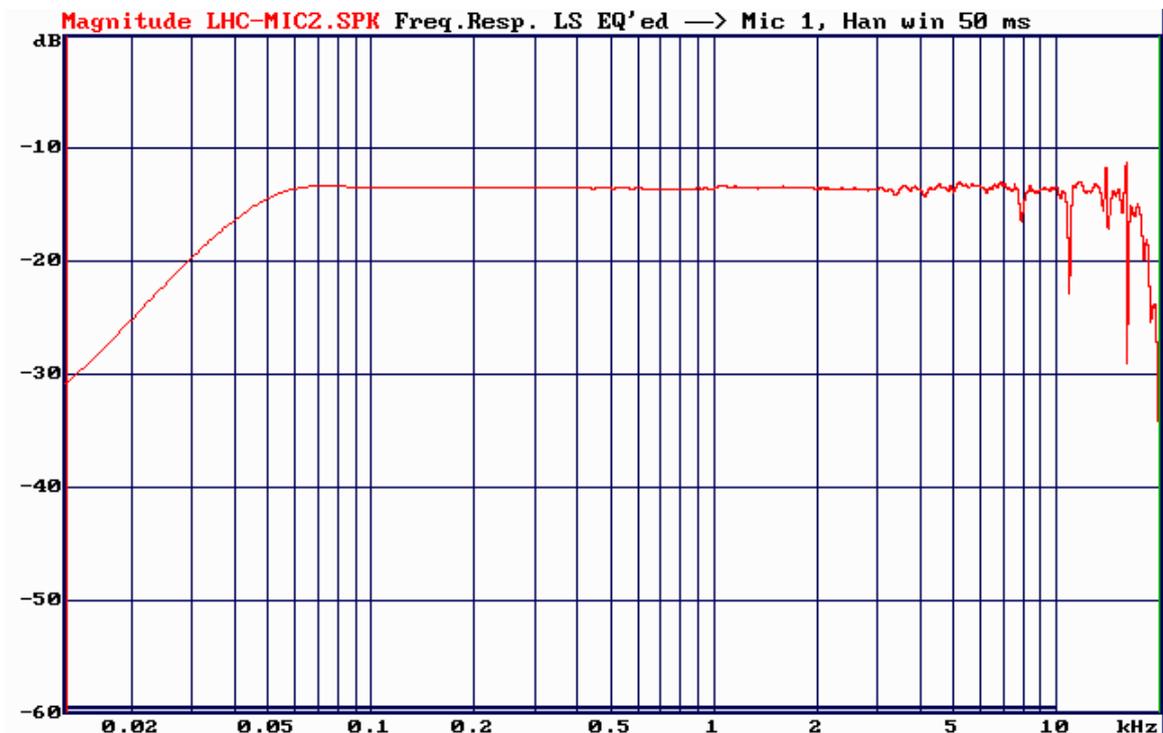


Figure 16: Measurement of the loudspeaker – microphone transfer path using windowing

Now it’s about time to apply the announced smoothing to fill the remaining narrow dips in the high frequency region. In the *Alt AD/DA frequency response* menu, simply set “**Smooth**” to yes and enter into the “**Parameter**” submenu and select a “**log**” smoothing of maybe 1/6 or 1/3 octave. The result should look as depicted in Figure 17. This will be our official LS-to-reference-microphone transfer function. Store it to disk, using a name like “MIC1.SPK” or something alike.

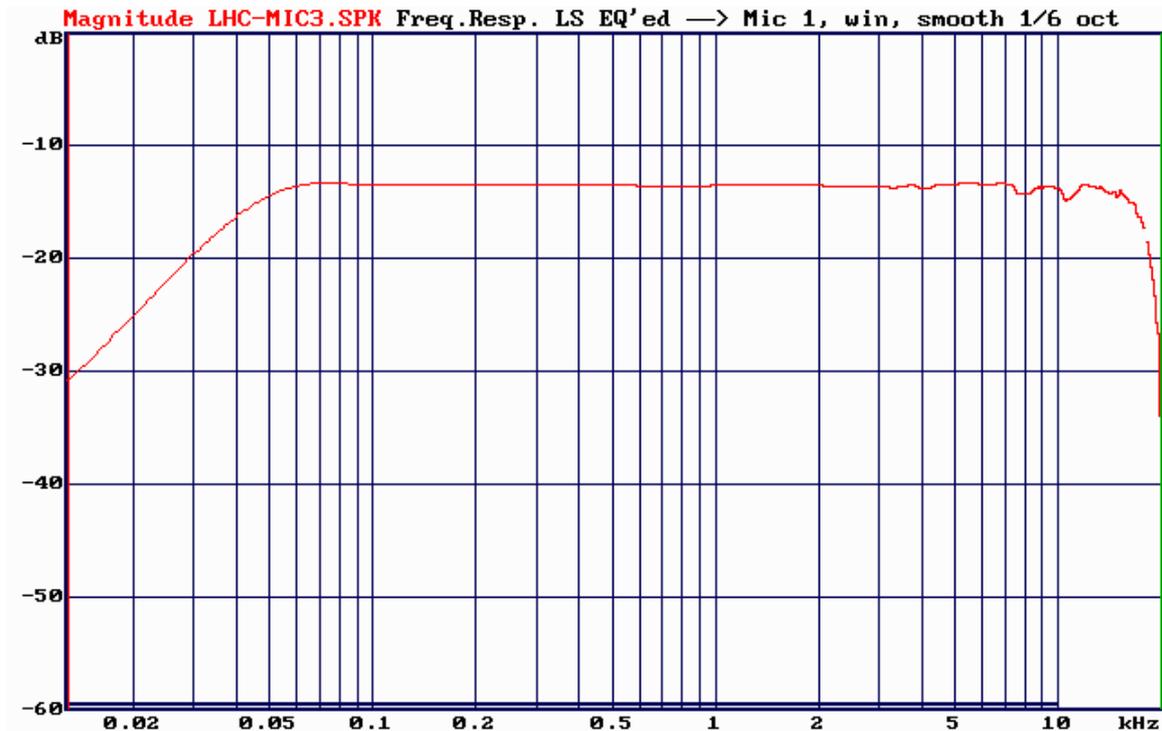


Figure 17: Measurement of the loudspeaker – microphone transfer path using windowing and smoothing.

We will now use this measurement as a secondary reference file. Again in the *Alt AD/DA frequency response* menu, enter into the “**processing**” submenu. Set “**file operation**” to “**yes**” and type the **name** of the file you just stored to disk or **browse** for it. Set the “**operation**” to “**divide through file**”.

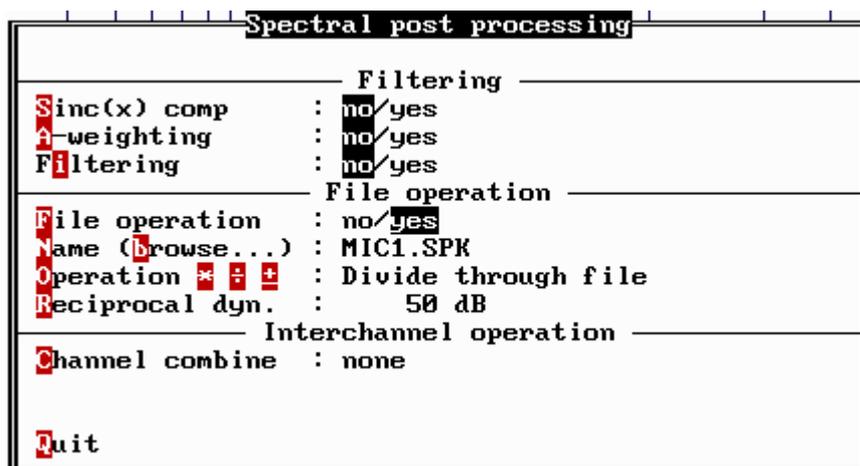


Figure 18: Spectral post processing menu with settings for microphone calibration

Now repeat the measurement. What should appear here is really a straight line at 0 dB (the “**Ref for 0 dB**” switch in the frequency response measurement menu must be set to “**1**” for this purpose).

If everything works fine, the moment has come to replace the reference microphone

by the MUT. Take care to position the MUT at exactly the same position as the reference microphone previously.

Do the measurement again (generally, you can always repeat the last kind of measurement by simply hitting the <Enter> key. This will lead you to the last *AD/DA* menu that you used). Check the input level, adjust it and repeat the measurement if it was not fine. What should appear now is the sensibility curve of the MUT, relative to the reference microphone. None is show here in order not to bring down the suspense! So go check it out by yourself now.

If you want the sensibility curve in $20 \log (\text{Mic.sens. [V/Pa]})$, there's an easy trick to achieve this. Simply load the secondary reference file (MIC1.SPK in our example) and multiply it with the inverse of the reference microphone sensibility. Example: If the reference microphone has a sensitivity of 50 mV/Pa (-26 dB), enter into the *Alt Edit, Multiply* menu (remember to position the cursors to the borders by typing <Shift> E first) and enter a "level" of 26 dB. Type <Enter> and store the multiplied result to disk, using the same old name. From now on, each microphone you measure will have its correct sensibility directly displayed.

You will discover that positioning the MUT as exactly as possible will have a big influence on the result. Even then, the different diffraction patterns of ref mic and MUT will cause ripple on your measured sensitivity curve. You can increase the smoothing bandwidth as long as you do not dissolve details of the microphone's frequency response (in general, as you know, a microphone's response is much smoother than that of a loudspeaker) to partially get rid of the ripple. This will also increase your measurement certainty by averaging out noise components.

Noise problems could occur at the low end of the sensitivity curve because of the restricted transmission range of our equalized source speaker. In many cases, it should be o.k. to replace the trace at very low frequencies with a straight line and pass the entire result through a high pass with the theoretical cut-off characteristics of the microphone (as dictated by the back orifice of the microphone and the coupling capacitor).

This text has been conceived outside a laboratory using barely simulations. So be skeptical about the contents and the amount of truth. If you find any errors or would like to suggest improvements, send a letter (put beautiful stamps, please) to the author swen.muller@gmx.de.

Good night and good luck in your comparisons.

