

Calibration of the audio High-End System in Cars

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High-End audio systems for cars are active systems, which means that each speaker is driven by its own amplifier. The input signals to the amplifier from CD, Radio, and MP3 etc. pass through a DSP (Digital Signal Processor) that includes a programmable digital Crossover for bandwidth restriction, also an equaliser for frequency compensation and also phase shifters and delays, to optimise the phase or time behaviour of all speakers in relation to each other, before hand.

The correct adjustment of the DSP with its complex functions demands appropriate measurement technology, only then make individual preference corrections real sense. The goal of master-mind use of DSP is a well balanced and pleasing sound quality of the audio system in general, a high definition of details, convincing stage imaging and auditory spaciousness and also minimum distortion such as that, that results from over powering the single speakers for instance.

1. Delay

Delay also means in this context delay-time or running-time. This function is the advantage of DSP crossovers to passive or discrete crossovers.

What the advantage is shall be explained by the reproduction of guitar sound. A guitarist plucks the string of an acoustic guitar. The first thing you hear is the metallic high frequency plucking noise. Then quickly following, the ground tone, resonate frequency of the string. The frequencies of the ground tones of an acoustic guitar lie in the bass/mid-range region. These frequencies cause the full sound of a guitar, because the string swings the body and the air in the body. This is time dependant because the body and the air have to swing into steady state. The sound creation has a time dependant behaviour, which is typical for the sound of a guitar.

In a typical car the high frequencies are produced by a tweeter in the A-Column, the mid and low frequencies by a speaker in the foot room. There is a difference in the distance of the mid-bass to the listener and that of the tweeter. Because of the speed of sound the music from the mid-bass speaker takes longer to reach the ear than that of the tweeter, as the distance is longer. By the reproduction of guitar sound this means that first you hear the high frequency noise of the plucking of the string from the Tweeter and only after a pause the low-mid frequencies sound of the ground tones. The listener hears a chopped up guitar sound. This is an unnatural sound and listening to music is no more a pleasure.

To reproduce the original sound the difference in distance or running-time difference has to be compensated. With DSP the forward running noise of the tweeter is delayed to the measure of the running-time difference. The resulting is the correct reproduction of the music, so that listening to music becomes an experience.

In P1. The delay function is described with help of the measurement technology. This is meant for understanding, but is not needed for setting up the DSP crossover.

2. The frequency crossovers

For a car audio crossover following values have been well-proven:

1st Order Crossover 6dB/Octave and 2nd Order Crossover 12dB/Octave.

Crossovers 3rd Order 18dB/Octave are needed for maximum power applications, for instance public address.

Crossovers with 24dB/Octave are from days gone by.

Crossovers 1st Order 6dB/Octave:

These crossovers allow an in-time reproduction of the audio system. The single Speakers have to have special properties, broad bandwidth frequency response and load capacity for frequencies beyond the reproduction range of the crossover.

Speakers are driven in-phase, same polling or 0° Phase shift.

Crossovers 2nd Order 12dB/Octave with Q=0.7.

For high sound levels, the choice of speakers is simple.

Speakers driven cross polled or 180° Phase shift.

The Phase shift of 180° is also described as phase reversal.

Schematic for:

2-way: Bass/Mid 0°, Tweeter 180°

3-way: Bass 0°, Mid-range 180°, Tweeter 0°

4-way: Sub 0°, Bass 180°, Mid-range 0°, Tweeter 180°

3. Connection plan for the DSP outputs



Pict. 3.1

Picture of the DSP rear side with outputs.

In the article the measurement calibration of a 3-way speaker system will be described.

The Outputs of the DSP will be connected as following:

CH 1 = Tweeter/left, CH 2 = Tweeter/right

CH 3 = Mid-range/left CH 4 = Mid-range/right

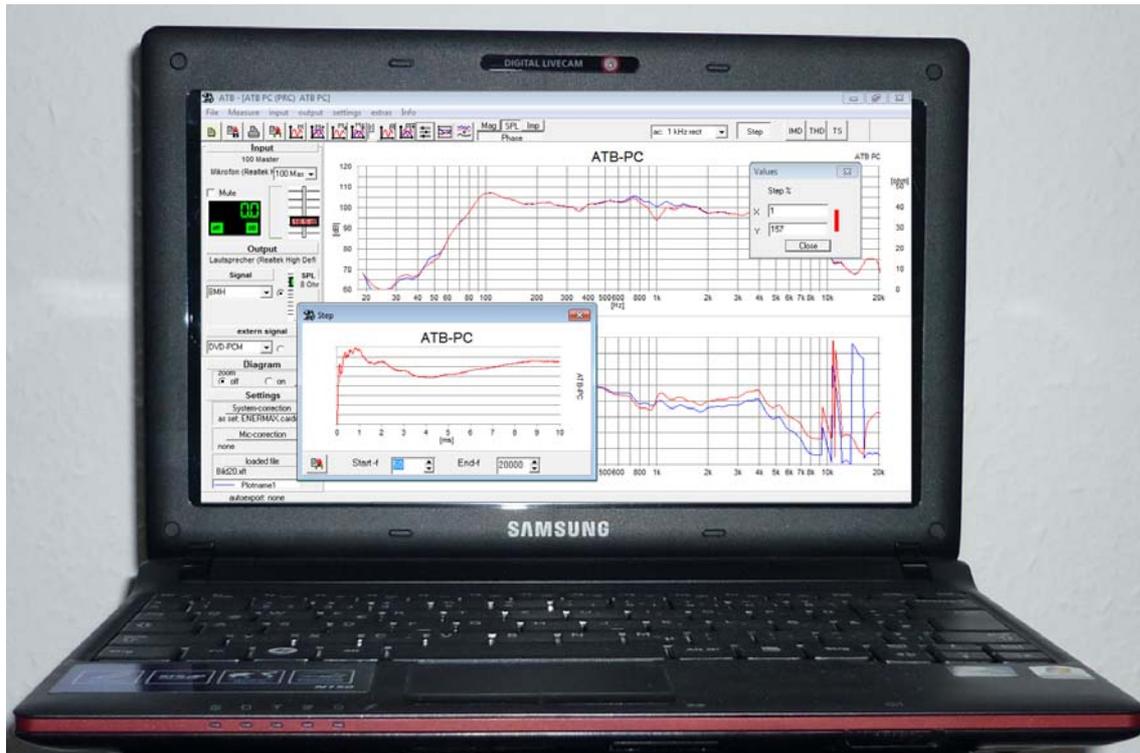
CH 5 = Bass/left CH 6 = Bass/right

CH 7 and 8 are left free, as an option they can be used with an amplifier for a pair of passive 2-way speakers in the back of the car.

The outputs of the DSP are connected to the amplifiers and the amplifiers are connected to the speakers. The choice of amplifier depends on the power displacement of the bass/mid speakers. For a homogeneous sound mid-range and tweeter should be driven by identical amplifiers.

The settings for sound level are made using the potentiometer on the amplifiers. The inputs of the DSP should be set to full level.

4. Running the software



Pict. 4.1

Picture 4.1 shows the netbook with ATB PC Pro software.

To set up the DSP Crossover you need a PC with the DSP programming software and ATB PC Pro software for the measurements. The Mosconi software and ATB run on Windows XP and 7. The demands on calculation capacity are low so that a netbook with ATOM processor is adequate for the task.

The Masconi DSP is used as an example for the measurement technology. Newer DSP's of other companies have the same functions so that the instructions can be used in connection with other hardware.

For the optimal setting and so for optimal sound, a precision measurement system is a total must-have! Measurement systems that only allow the measurement of frequency response has to be very precise. When the system doesn't allow phase and step measurement the user has to rely on the precision of the frequency response measurements. Software that uses pink or white noise as a measurement signal were developed for noise measurement. The measurement curve changes constantly so that the measurement is undefined and too unprecise. They are not suited for setting up high quality DSP systems.

Only when the DSP is correctly set up transparency, heard through ability of even quiet signals, the in space imaging and voluminous presentation of ensembles and instruments will gain in a stunning manner, the sound impression is finer and more realistic. And that is what ATB can do for just any sound system.

If you abstain from that, hearing music in the car usually becomes stress and nerve raking instead of being a pleasure.

For noise level measurements a microphone is included in the ATB package.



This is connected directly to the sound card input of your PC. The microphone has a very linear frequency response and therefore does not need to be calibrated.

The Software is run after installation.



The Mosconi program runs in the background and you switch between the programs by minimising or maximising the ATB program.

You switch to the Masconi program and choose DSP in the device menu. After that the DSP is assigned to one of the USB outputs. Setting up the connection takes a few seconds and is verified by following message:

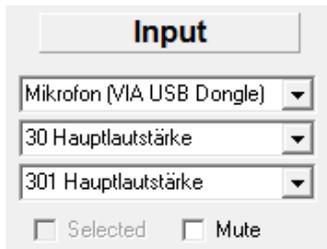
Status: verbunden mit COM4, Firmware 1,4



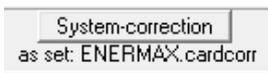
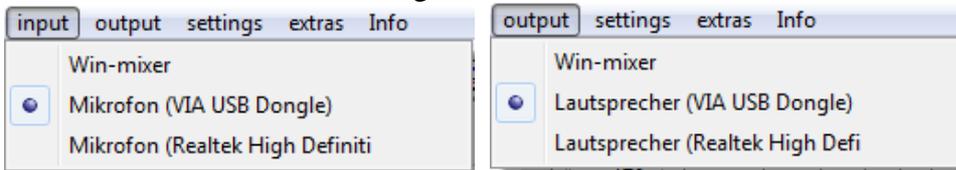
This window appears.



Here you choose “Copy Data to Console” Preset1 . With a non programmed device the basic settings are written into Preset1 and the device is ready for use. For the first measurements following settings are to be made in the ATB program:
ATB PC sound board settings



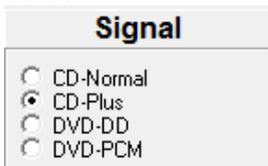
ATB PC Pro sound board settings



System correction, for impedance and phase is needed.

With the ATB PC the test adapter is connected with the output of the CD, DVD or PC to the microphone input of the measurement PC sound board. You play either the Auto-Test CD or the WAV measurement file from the installation CD ROM.

With the ATB PC Pro the test-box is connected using the phone jack- cinch adapter to the head-speaker output and to the microphone input using the cinch plug, of the sound board.



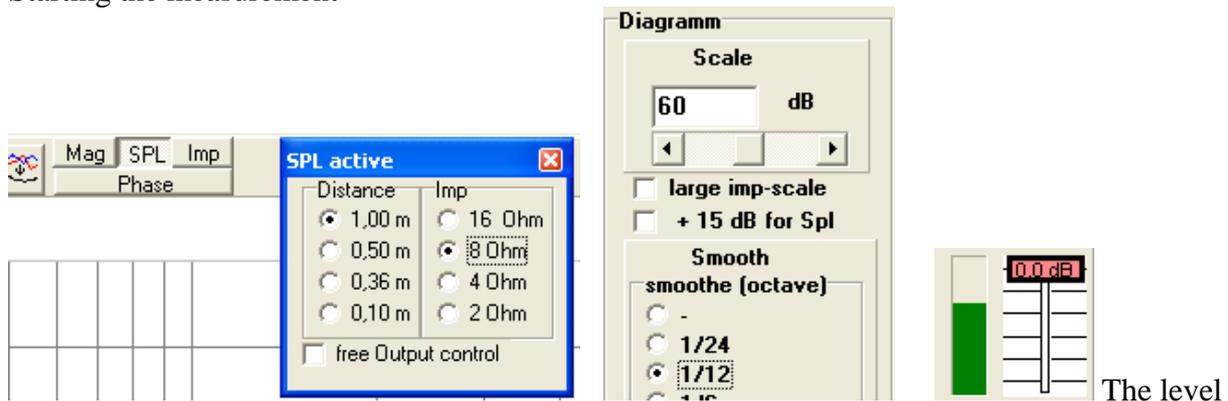
Always use BMH=CD-Plus signal. A WAV File is to

be found on the installation CD-ROM.



gives the assignment Curve to dB Scale

Starting the measurement



indicator should be showing a green bar after the measurement.

With the ATB PC program the measurement signal comes from playing Track 1 of the Auto-Test CD in the cars CD player. But you can also copy the WAV file from installation CD ROM to a USB stick and play that on your car audio system.

With the ATB PC Pro you don't need a player, if the head-speaker output of the sound board is connected to the DSP input.

With the ATB PC or ATB PC Pro you carry out the simple set up measurement described in the first part of the instructions.

For professional set up measurements you need the ATB PC Pro program which is identical to ATB501PC, ATB701PC and ATB801PC programs.

5. Equalising the single speakers

The DSP offers you the possibility to linearise the speakers using the filter functions or equalisation of the separate channels. But this should only be done by construction dependant dips or humps in the speaker's frequency response. These can be compensated. Other deviations such as rising response to higher frequencies due to radiation resistance, are accounted for when filtering, using the crossover. This also applies for the never linear frequency response due to placement or listening room. Any manipulation of the frequency response will also affect the Phase and Impulse behaviour.

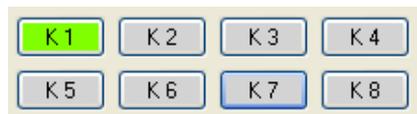
Experience shows that equalising is not necessary when using high quality speakers. Equalising is described in the professional chapter. For demonstration a low quality speaker is used that does not belong in a high quality system.

6. Tweeter frequency response

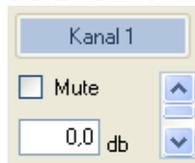
First of all we set up the crossover for the tweeter.

Following settings are made in the DSP program:

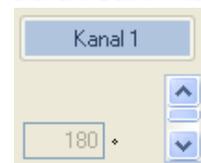
Main menu on/of



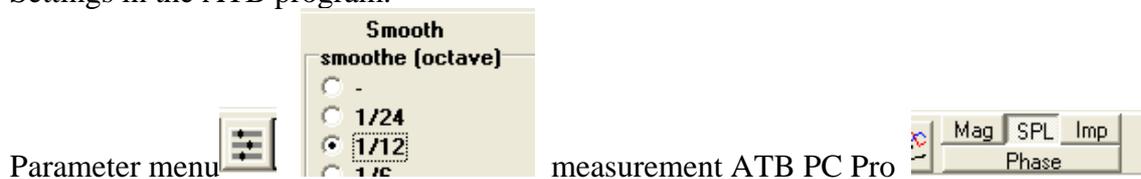
Menu level



Menu Phase shift



Settings in the ATB program.



Microphone settings



The playing medium for the measurement signal is

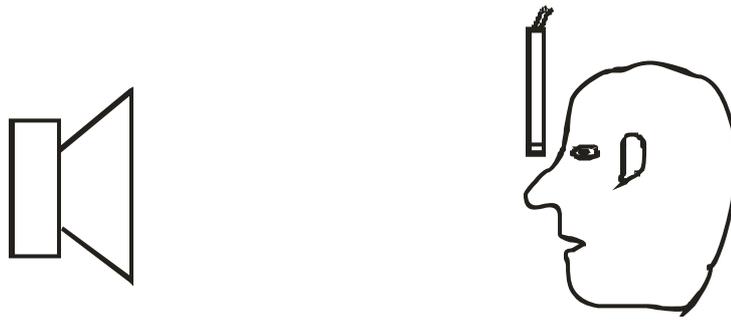
started with  and the measurement started with .

After the measurement the playing medium is stopped with . With ATB PC Pro starting the player is not necessary when the headphone output of the sound board is connected to the input of the DSP.

In the menu Phase Shift of the DSP program channel 1, tweeter, is set to 180°. This 180° is set back to 0° after setting up the crossover for the tweeter/mid-range, in accordance with our scheme.

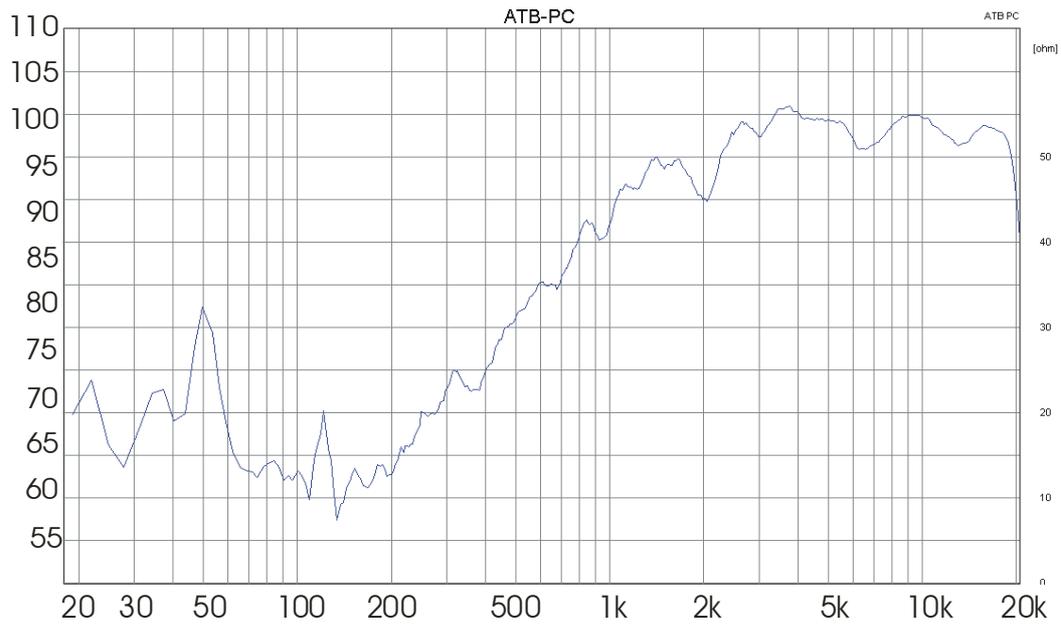
For smoothing the measurement curve 1/12 octave is chosen in the ATB parameter menu. With this setting no information is smoothed away, the curves are not falsified.

The set ups begin with the tweeter on the driver's side. This is switched on with Channel 1, CH 1. The frequency response is measured with M and the microphone position of



the driver's head.

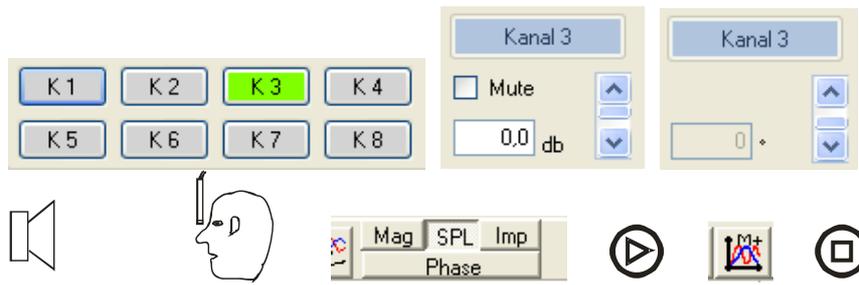
The picture shows the microphone in the position of the driver's head. For sound relevant measurements the driver's seat should be occupied. The microphone is held in front of your nose facing downwards.



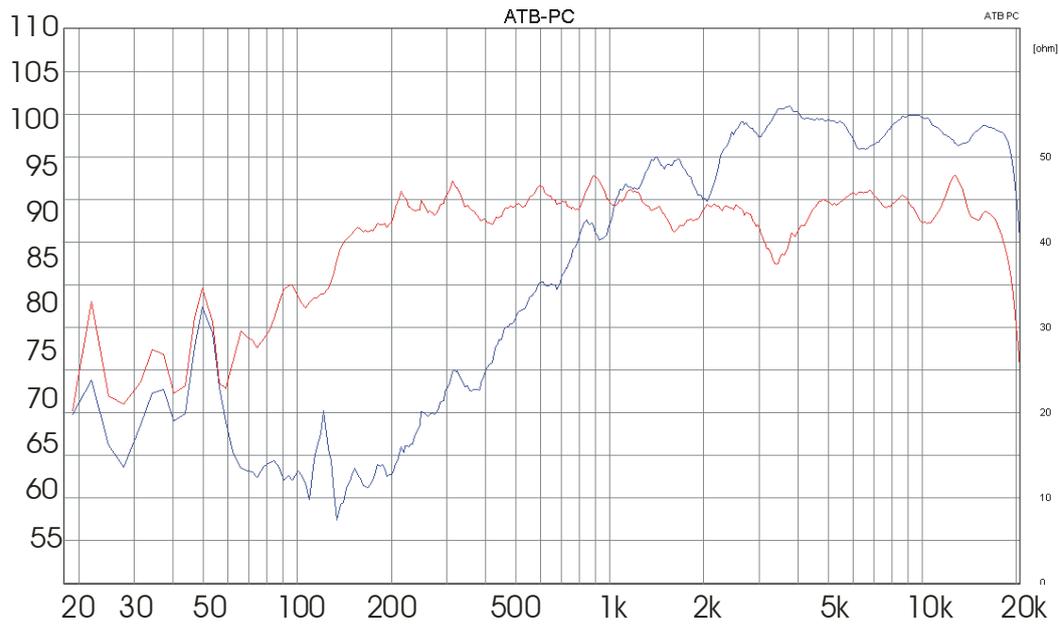
Pict. 6.1

The frequency response of the Tweeter.

7. Frequency response of the mid-range speaker.



Channel 3 is switched on alone, frequency response with M+ measured in head position.

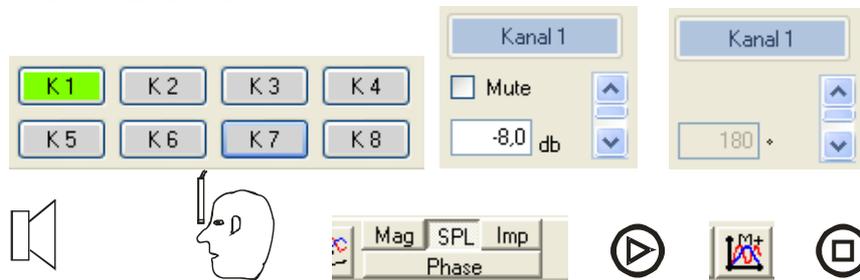


Pict. 7.1

The red curve shows the frequency response of the Mid-Range Speaker.

8. Adjusting the sound level of the tweeter and mid-range speakers

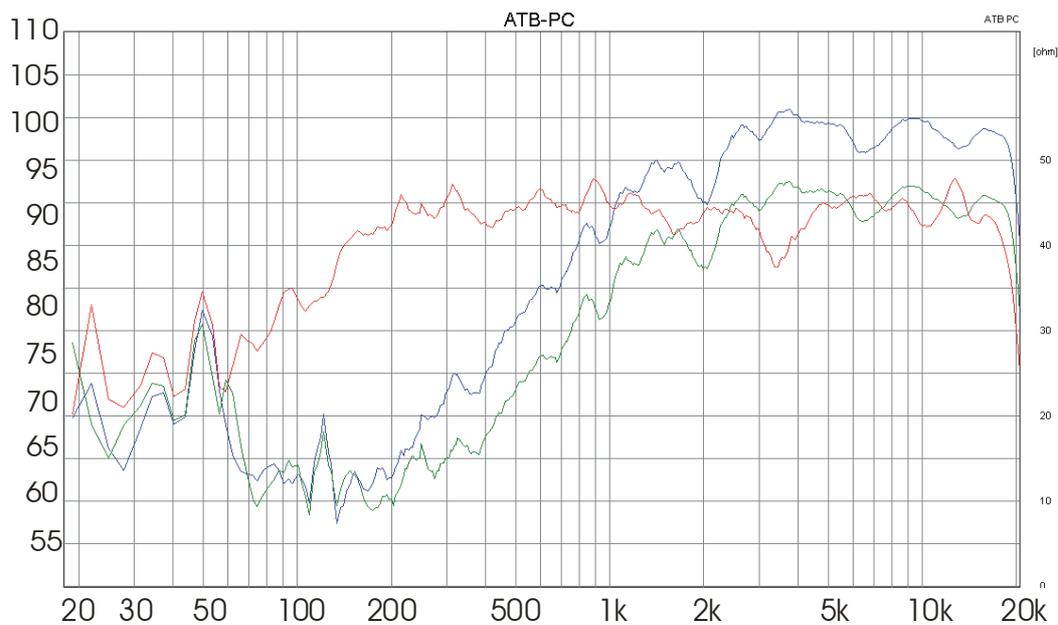
The frequency responses in pict. 7.1 show that the Tweeter has a higher sound level than the mid-range speaker. For the crossover settings both speakers are adjusted to the same sound level.



The tweeter is driven by itself and made 8dB quieter, setting -8dB in the sound level menu.

If you are using a simple DSP with digital sound level settings, it is better to adjust the sound level using the volume potentiometer of the amplifier.

Measurement takes place with M+.

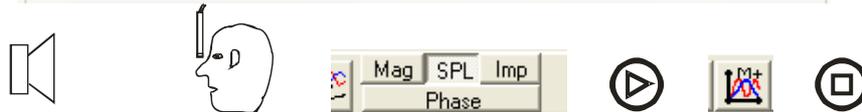
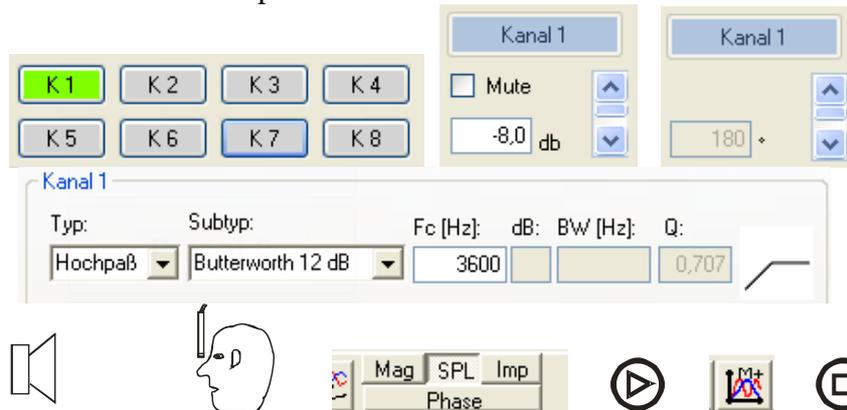


Pict. 8.1

The picture shows the adjusted sound level of the speaker, green curve.

9. Crossover for tweeter

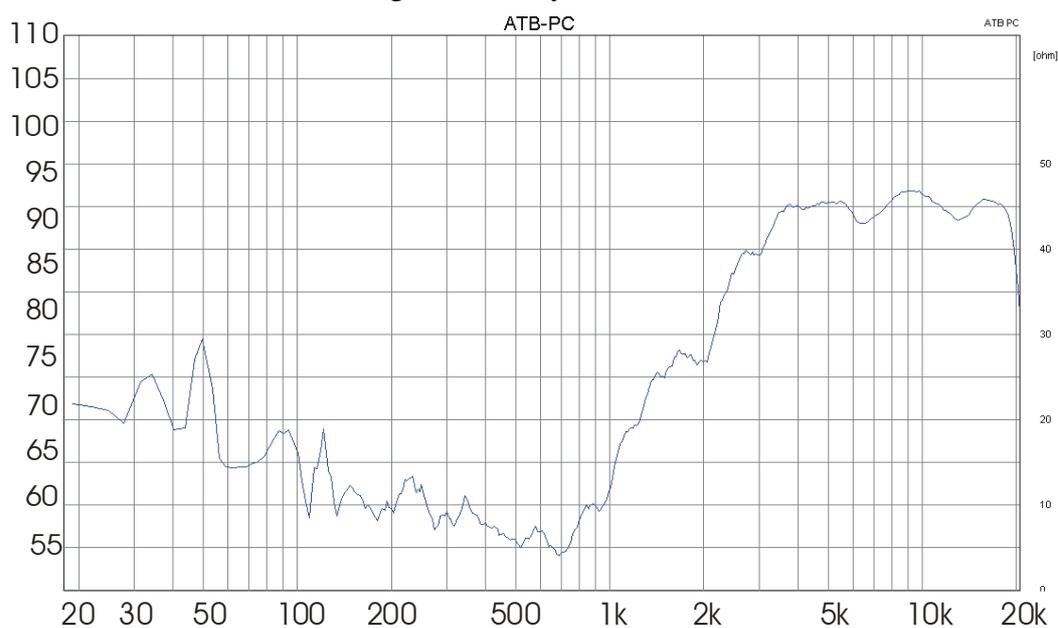
First of all we set up the crossover for the Tweeter.



The Butterworth 12dB crossover was already introduced in the chapter Delay. To set it up we need a cut off frequency. This is determined by the Tweeter construction and where it is situated. Most of the time dome tweeters are used. These have a good radiation behaviour and can, for instance when built into the A-Column, radiate freely. To utilise that advantage the cut off frequency should be as close to the mid-range as possible. But the cut off frequency is restricted by the power handling capabilities of the Tweeter. The lower the frequency, the larger the hub of the dome of the Tweeter. The hub of the speaker is only in an area, determined by the height of the pole plate and length of the driver coil, linear. If the dome exceeds this area Total Harmonic Distortion results. This can be measured by THD in ATB PC Pro. The rise of THD should lie at one octave below the cut off frequency.

The thermic power displacement can be taken from the technical description of the speaker. In the technical description the power displacement for a certain crossover is noted, for instance, 80W, 12dB and 3.6kHz crossover. That would be the frequency you choose for Fc.

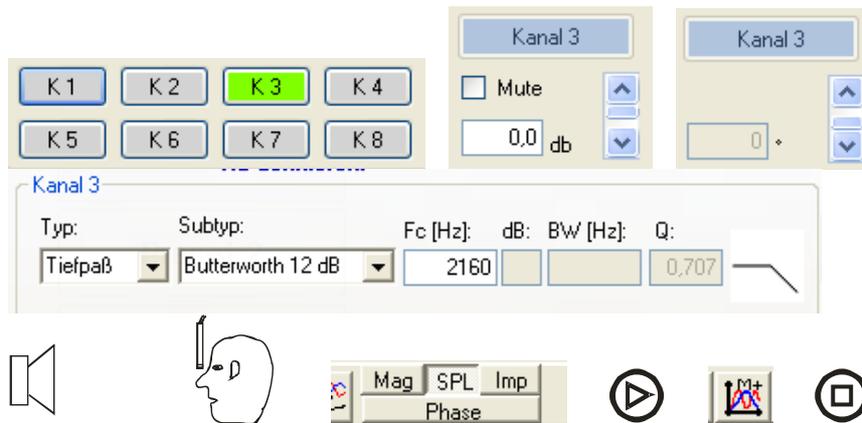
In the DSP menu active crossover/filter you choose for Channel 1+2 high-pass. Settings are: Type = high-pass, Subtype = Butterworth 12dB and Fc = 3600Hz. Measurements are made in the driver head region and only Channel 1 is switched on.



Pict 9.1

The measurement verifies a fall at 3.6kHz of -3dB

10. Crossover for mid-range speaker



The crossover for the mid-range is set up as such that the combination of tweeter /midrange results in a balanced frequency response. As the crossover for the Tweeter has been done the mid-range is made to fit. So that the falling slopes of both frequency responses are symmetrical and add to result to a flat line, Butterworth 12dB/Octave is also activated in the DSP menu active crossover/filter for the mid-range crossover. So that the frequency response of the mid-range to tweeter is balanced the slopes of the frequency responses have to cross at -6dB. The reference point of 0dB is the average sound level of the speaker.

It would be nice and simple if Fc, the cut-off frequency of the tweeter, could also be quoted as the cut-off frequency of the mid-range. But this is not possible, because then the curves would cross at -3dB at Fc causing an over rise in the frequency response.

In the DSP menu active crossover/filter the stated cut-off frequency is for -3dB, according to science the definition of filters are following:

For Butterworth crossovers:

Cut off frequency x 0.77 = Threshold frequency Low-pass

Cut off frequency x 1.3 = Threshold frequency High-pass

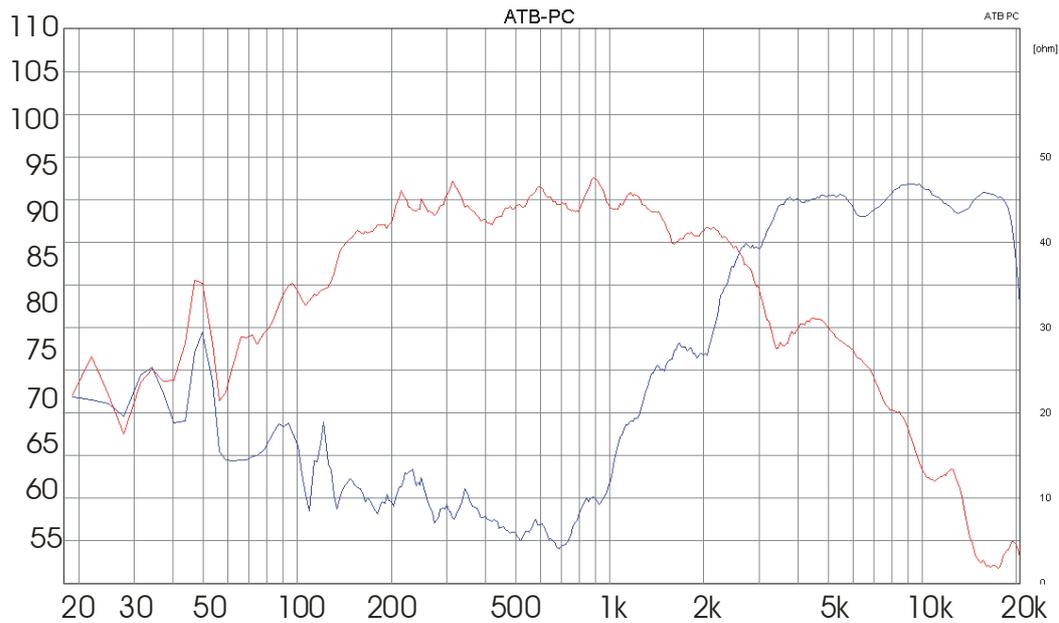
For the choice of the Fc for the mid-range following formula will help:

$Fc_{\text{High-pass}} \times 0.77/1.3 = Fc_{\text{Low-pass}}$

$Fc_{\text{Low-pass}} = 3600\text{Hz} \times 0.592 = 2132\text{Hz} \sim 2160\text{Hz}$

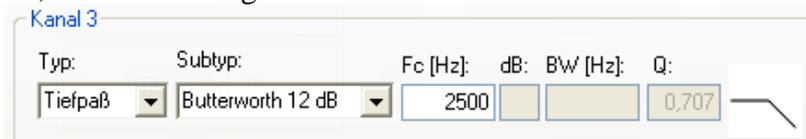
The ATB measurement confirms the calculation.

In the DSP menu active crossover/filter Low-pass is chosen for Channel 3+4, for the Mid-range. Type = Low-pass, Sub-type = Butterworth 12dB at 2160Hz.

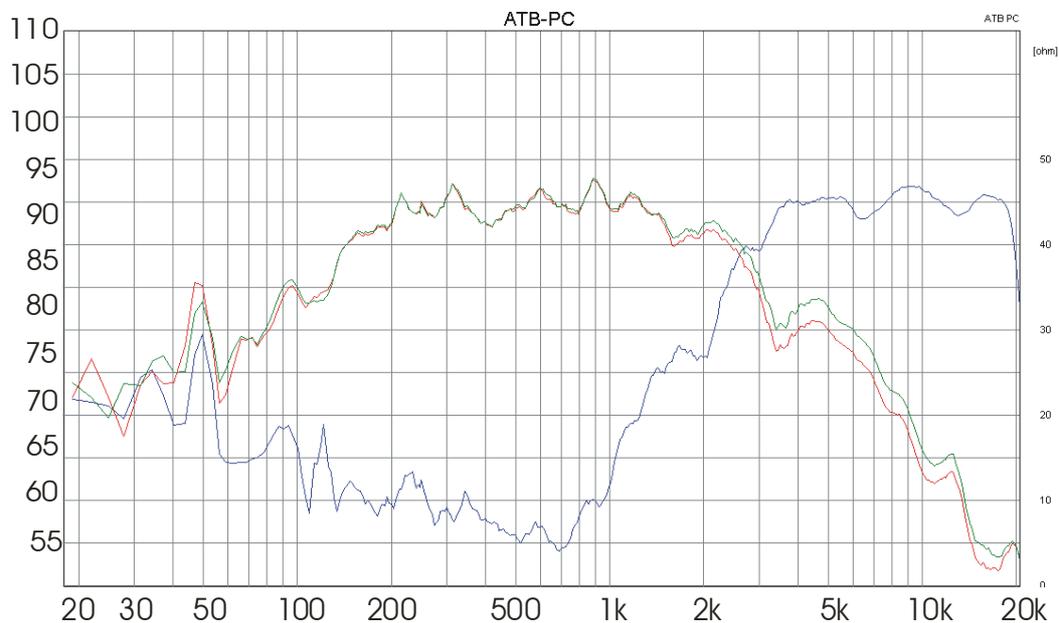


Pict. 10.1

The measurement with +M in head area shows that the curves cross at -6dB. But as both speakers do not have an ideal frequency response at crossover, the cut-off frequency, F_c , of the Mid-range is made to fit and set at 2500Hz.



The measurement is carried out with

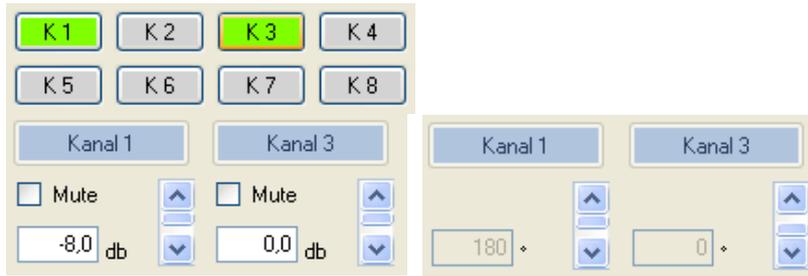


Pict. 10.2

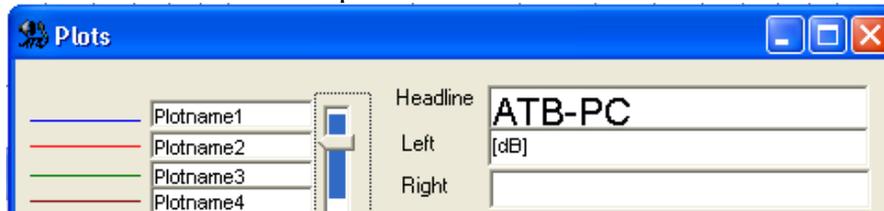
The picture shows the crossover point of mid-range and tweeter.

11. Transition tweeter and mid-range speaker

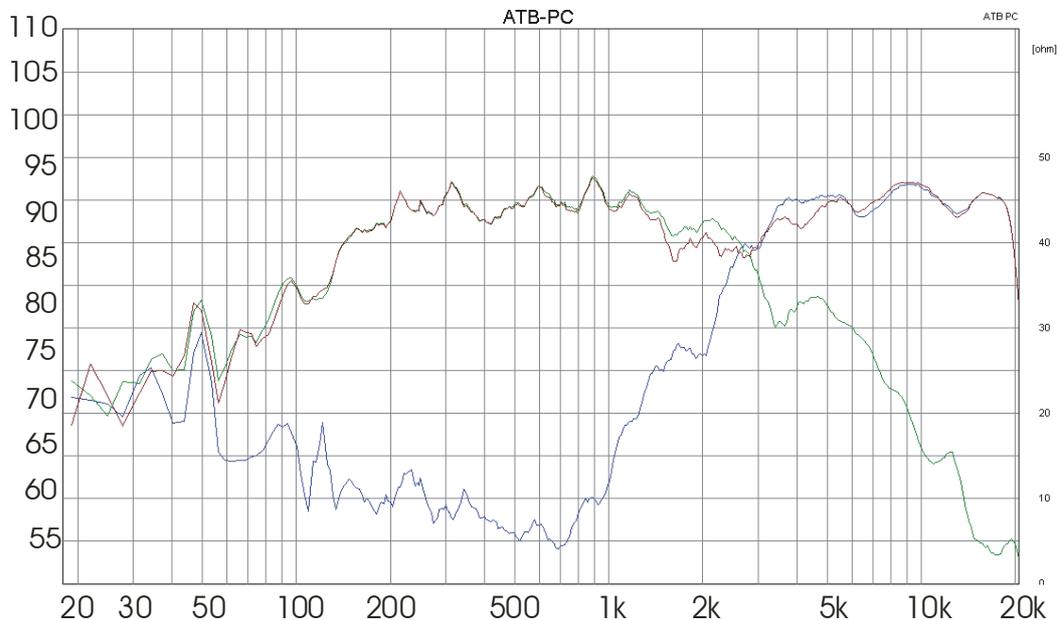
For the simultaneous measurement of mid-range and tweeter the tweeter is also switched on.



For a better overview the plot menu is switched on with



The red curve of the old filter setting is deleted with



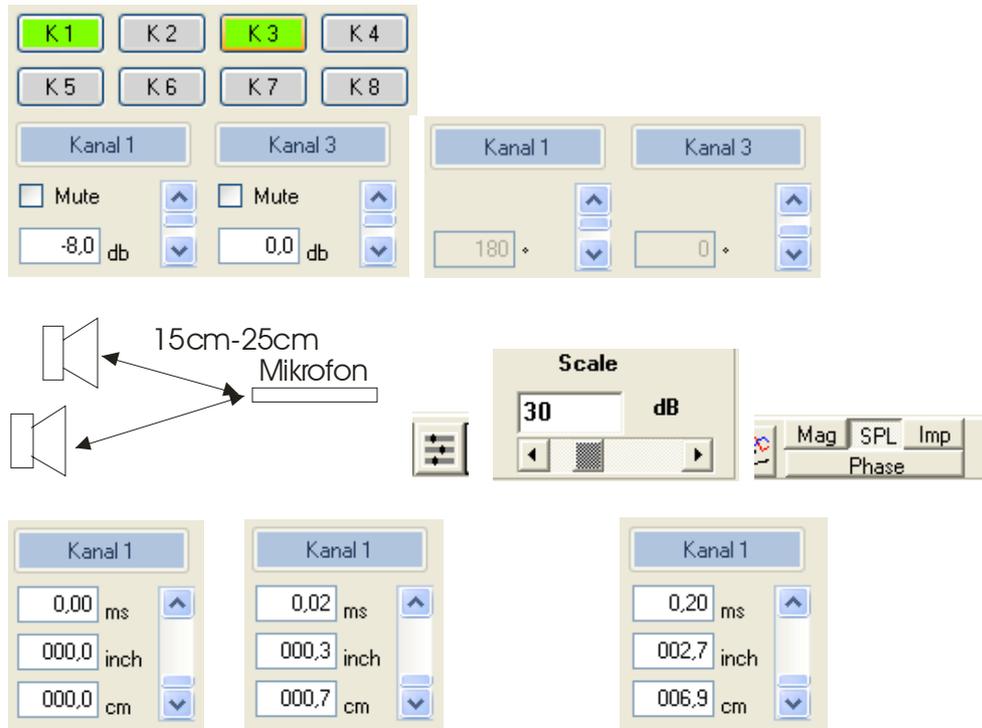
Pict. 11.1

The picture shows a big drop in the frequency response at the crossover frequency, brown curve. This shows that the acoustic phase of the two speakers have a phase shift of 180° so that there sound is deleted. Next the delay of the speakers is adjusted so that the frequency response has no drop.

12. Setting up the delay of the tweeter

The delay allows you to adjust the timing of the speaker. With the delay the running distance and as such the running-time to the listener can be compensated.

First of all you measure the distance from the speaker to the listener. The speaker with the shortest distance to the listener is delayed. In our user instructions it's the tweeter, but if the mid-range is closer that will be delayed. The to-do when calculating the delay is identical.



Multiple measurements are started with   and carried on with .

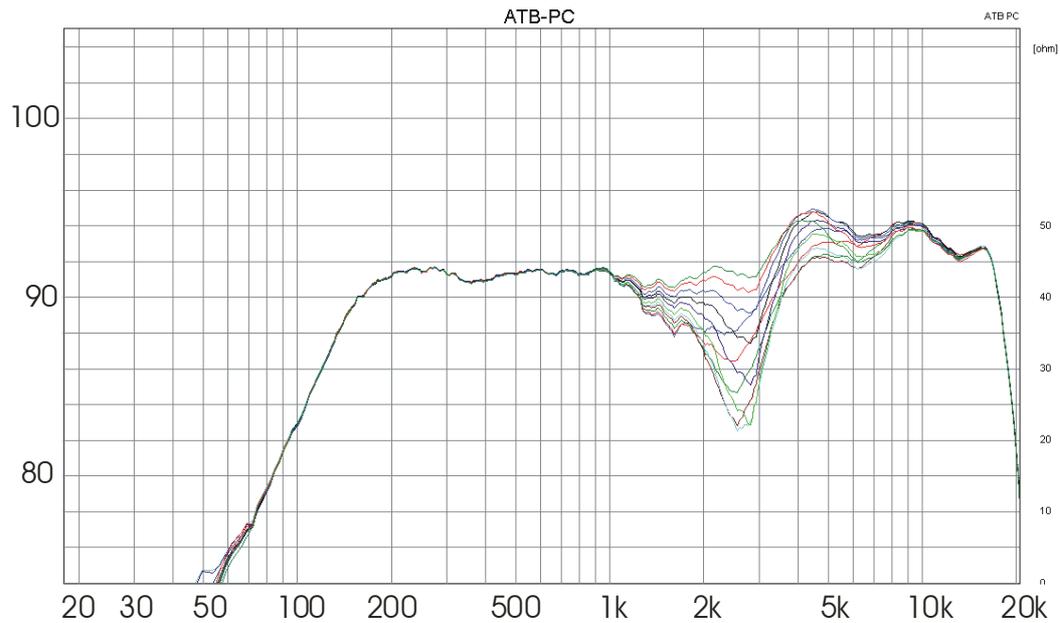
After that the player is stopped with .

Mid-range and tweeter are measured at the same time and switched on with CH 1 and CH 3. The measurements are carried out in near field as the direct sound is of interest. When measuring in the head area the influencing part of reflections is so big that time dependant measurements like delay, acoustic phase and step are not possible. The microphone is positioned between the speakers at a distance of 15 to 25 cm. It lies in the midpoint between both speakers to listener head position.

Using the M function the measurement is started. The delay is set to 0.02 for CH 1 in the menu delay. The following measurements are started with M+, and when the frequency response is balanced no new curve is measured and the player stopped.

The ideal value is in the area of the running difference of both speakers to the microphone.

The values of the difference found by the ruler and by measurement are not the same, but should not differ too much. If the running distance is larger, the measurement should be started with a higher value for the delay.



Pict 12.1

For better demonstration the scale was set to 30dB on axis.

The picture shows the curves for 0.02ms steps. The maximum drop at the cut off frequency is at 0.1ms. After that the curve becomes more and more balanced. The green curve shows the optimal setting at 0.2ms. The balanced frequency response allows you to assume that the signals are coming in at the right time as far as the mid-range and tweeter go. A delay of 0.2ms means a distance of 6,9cm. Using a ruler the difference is 3.6Cm which just goes to show you can't measure the running time with a ruler. This difference of 6.9cm is not really big as a car allows you to build in the speakers close to each other.

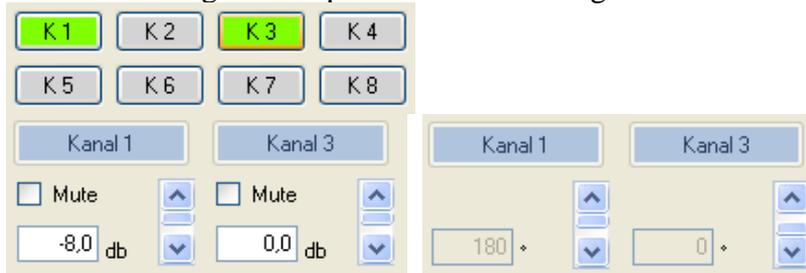
13. Listening test for the delay settings

The delay setting would have no meaning if it was not clearly to be heard.

For the listening test you choose track 4, Pink noise, on the CD player.

Important: Pink noise is an analogue generated signal. The frequencies in the signal are placed at random, it is not periodic. This makes it comfortable to listen to. Pink noise is not suitable for measurements. The measurement signal Cdplus is a calculated signal and is periodic. Through the calculation the signal contains exactly defined frequencies those are placed evenly. When evaluating the measurement signal a period that contains all frequencies evaluated. Contrary to measurements done with white or pink noise the measurements are precise and replicable.

For the listening test the previous DSP setting are used.



You start the CD player with  and the change the delay in the DSP menu.

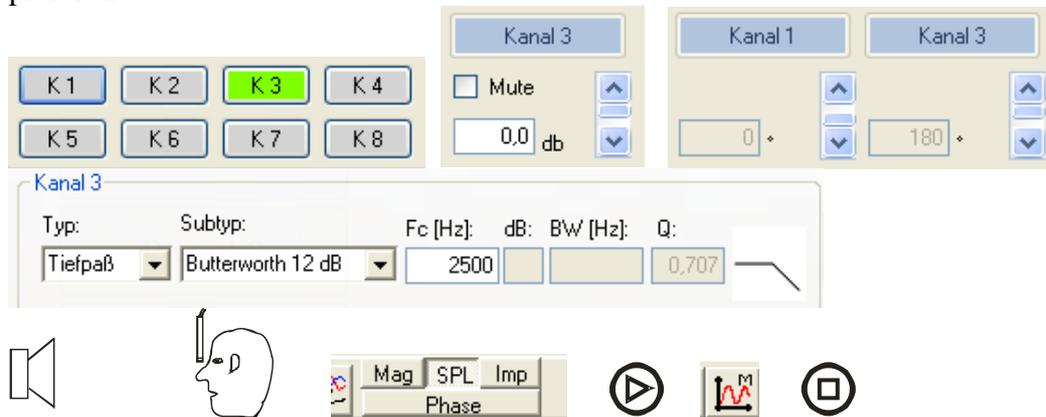


With a time delay of 0ms a hard noise is to be heard. The high frequencies are dominant and the tweeter is easily located. As the delay increases the tweeter becomes less dominant and you hear the noise also from the mid-range speaker. With the right delay time the noise is located between both speakers and is well balanced. The result is a new speaker that doesn't change the timing within the music signal.

This hearing impression also relates to other sitting positions in the car. This shows that the sound of speakers with time alignment is less dependent on the listening position.

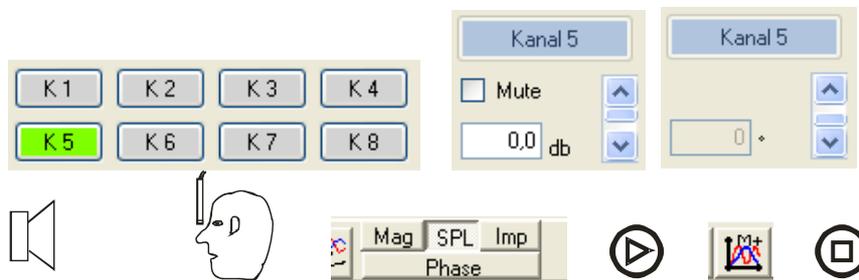
14. Sound level adjustment between mid-range and bass speaker

In the next step you set up the crossover between mid-range and bass speaker. The frequency response of mid-range and tweeter are measured and the sound level of the Bass is adjusted to fit. The adjustment is the basis for setting up the crossover. Once the crossover is set up the sound level of the bass can be adjusted to suit listening preferences.

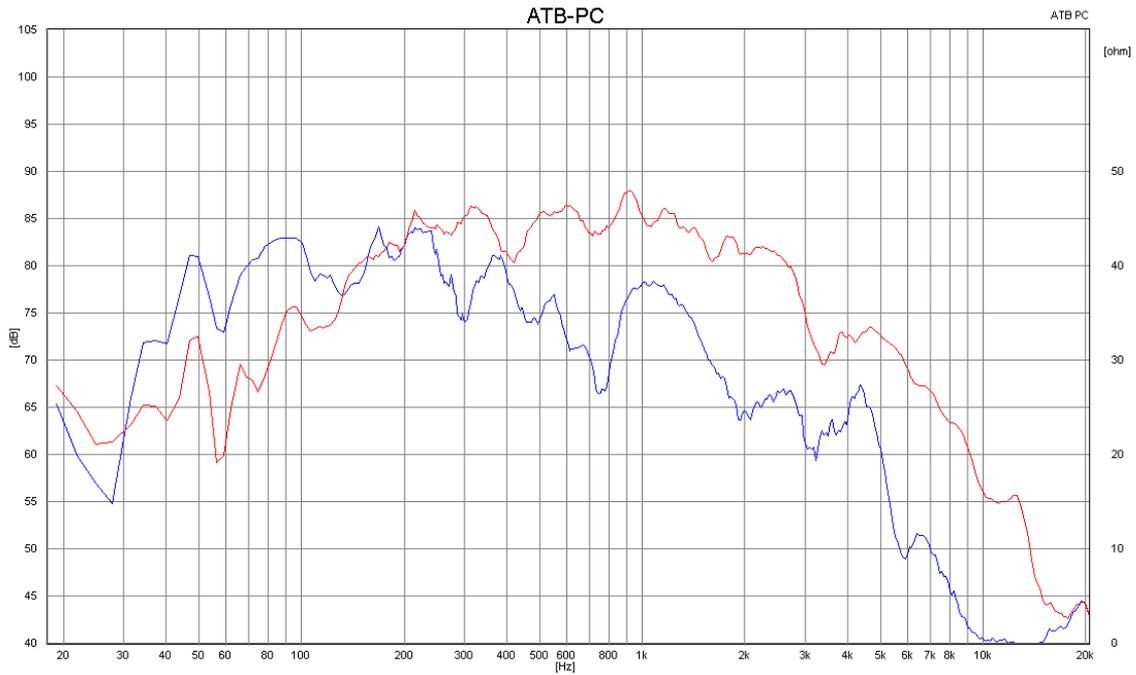


For the first measurement of the mid-range the measurement is started with M and Channel 3 is switched on.

In the menu for Phase shift Channel 3 is set to 180° and Channel 1 to 0° . The settings in menu active crossover/filter remain as they are. The measurement is done with the microphone in head position. The measurement started with M.

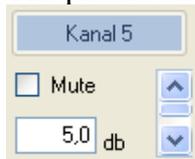


For the measurement of the Bass speaker Channel 3 is switched off and Channel 5 switched on to measure the Bass on its own, by clicking +M.

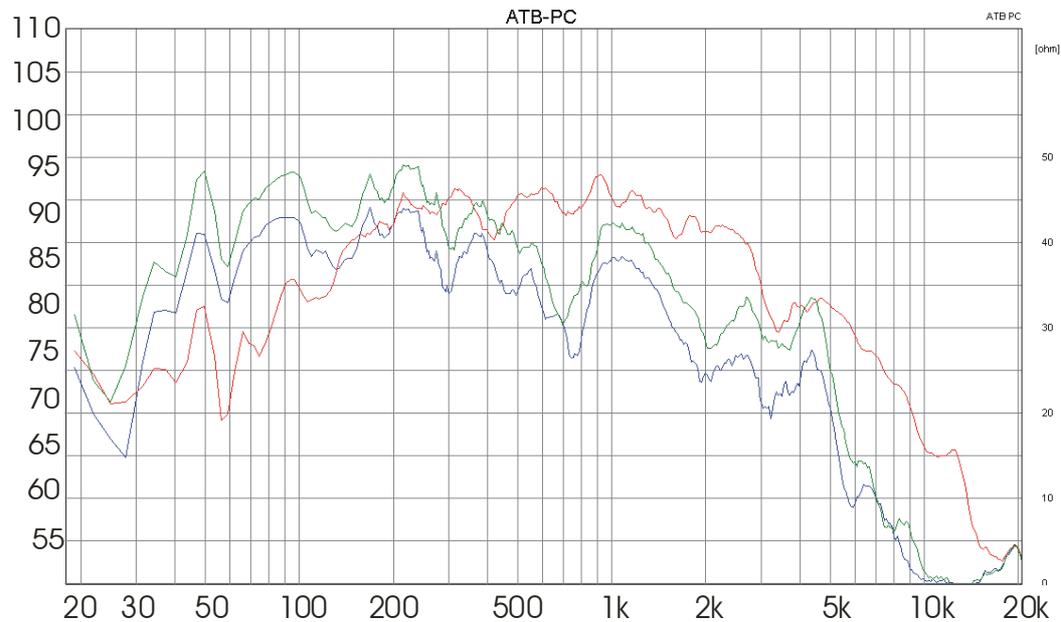
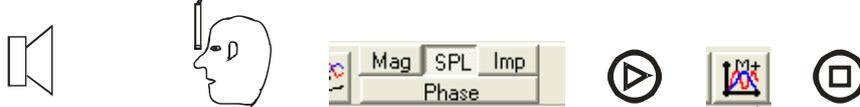


Pict. 14.1

The picture shows that the Bass, blue curve, is 5dB too quiet.



In the DSP menu sound level the Bass is set 5dB louder.

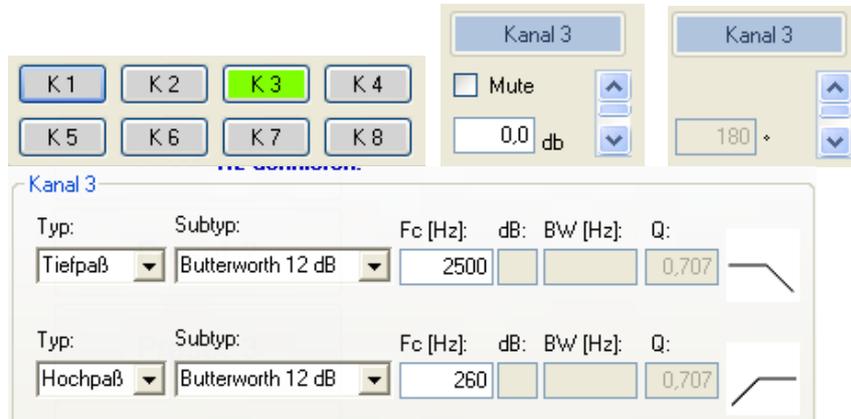


Pict. 14.2

The picture shows the 5dB louder adjusted Bass, green curve.

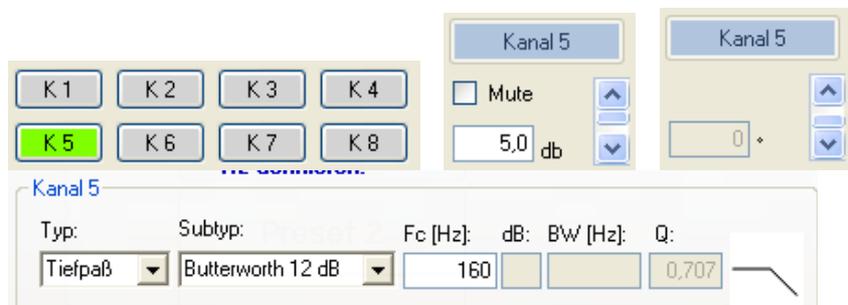
15. Crossover for mid-range and bass speakers

After adjusting the sound level the crossover for the Mid-range is set up. We use the Well-known Butterworth 12dB crossover. To choose the cut off frequency we have a look at the frequency response of the Mid-range, red curve in picture 15.2. From 200Hz onwards the frequency response is quite even. As the Mid-range can also handle bass we can choose 200Hz as the cut-off frequency. F_c is calculated with $1.3 \times 200\text{Hz} = 260\text{Hz}$.

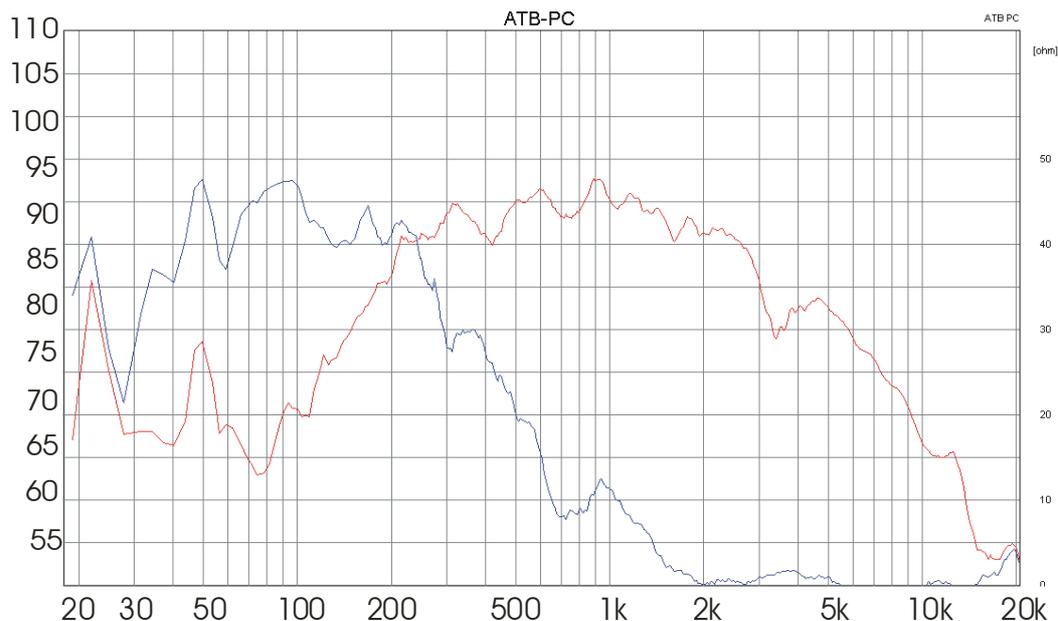


In the menu active crossover/filter Channel 3+4 is set to: Type = High-pass, Sub type = Butterworth 12dB and $F_c = 260\text{Hz}$.

For reasons of symmetry we choose Butterworth 12dB/Octave for the Bass speaker. As for the Mid-range/Tweeter crossover we calculate the F_c with:
 $F_c \text{ Mid-range} \times 0.6 = 156\text{Hz}$, rounded to 160Hz.



For the Bass we set up Channel 5+6 to Type = Low-pass, Sub type = Butterworth 12dB and $F_c = 160\text{Hz}$. The measurement is done with the microphone in head position and started with +M.



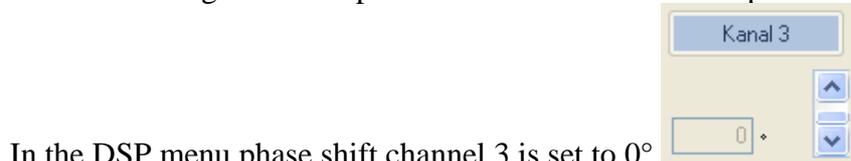
Pict. 15.1

The picture shows the frequency response of the bass and mid-range. The crossover frequency is at around 200Hz and the geometrical addition of the slopes suggest an even frequency response.

The question if the speakers compliment each other on transition depends on the acoustical phase of the speakers. As phase measurements are not possible using ATB PC the frequency response will have to indicate that. To locate the right phase shift step by step both speakers are measured with a phase shift of 180°.

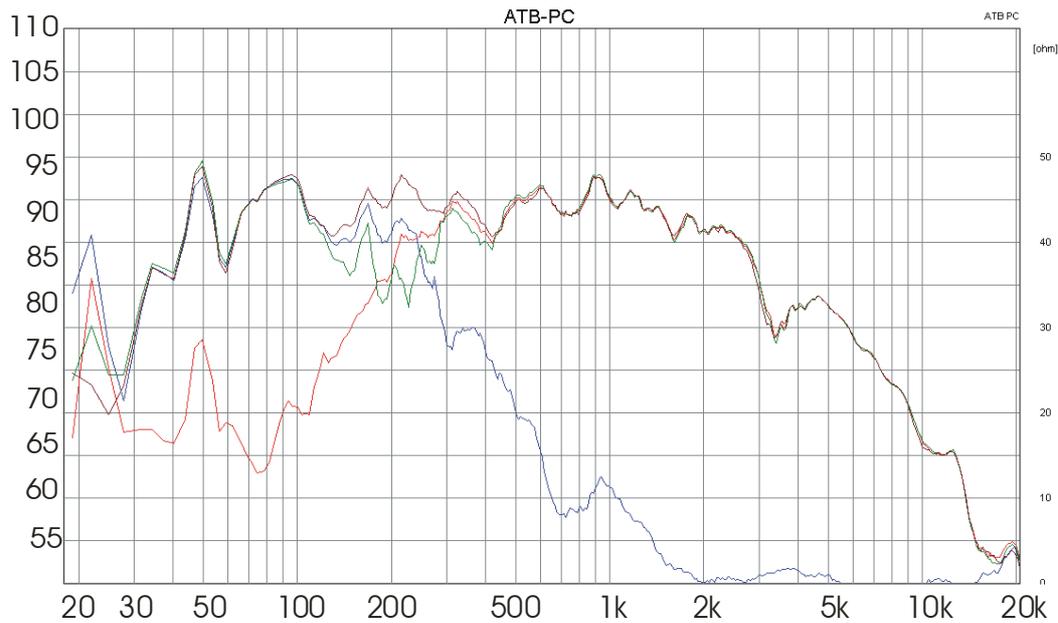


In the following test both speakers are measured without phase shift.



In the DSP menu phase shift channel 3 is set to 0°





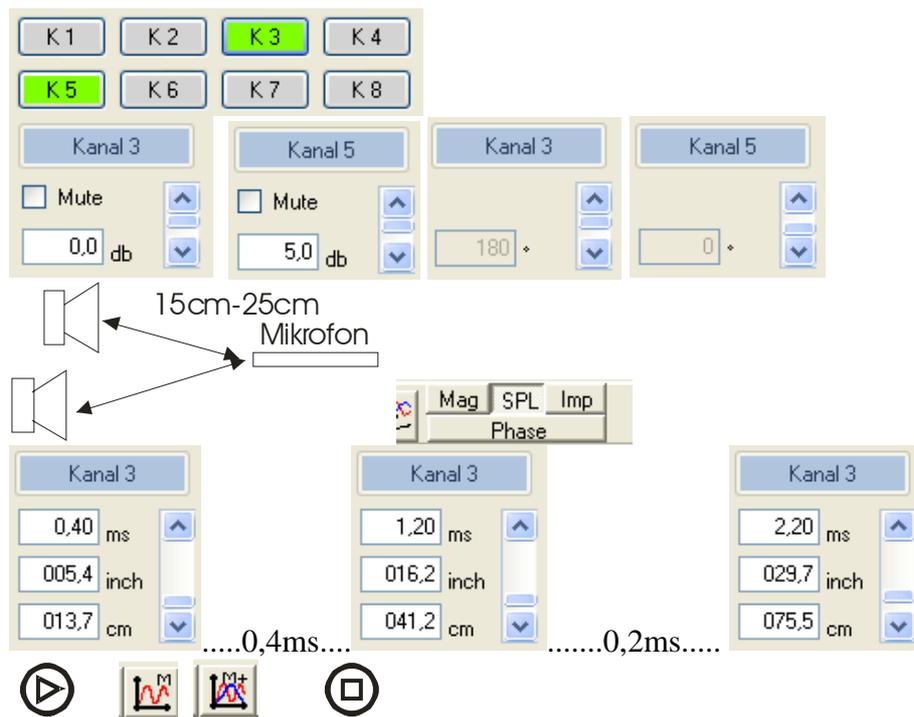
Pict. 15.2

The picture shows the combined frequency response of bass and mid-range. The green curve shows the response when the speakers are cross poled, the phase shift is 180° . When the phase shift is set back in the phase shift menu of the DSP the measurement shows a balanced frequency response. But as shown in capital Delay, a correct reproduction is only possible with a Butterworth 12dB crossover when the speakers have a phase shift of 180° . So that both speakers radiate with correct timing, the running-time due to the difference of distance to the listener is compensated. With the delay function the mid-range is delayed. This is important for an even frequency response. An even frequency response is achieved when both speakers are in phase. This is accomplished with a phase shift of 180° and correct delay.

16. Determination of the delay for the mid-range speaker.

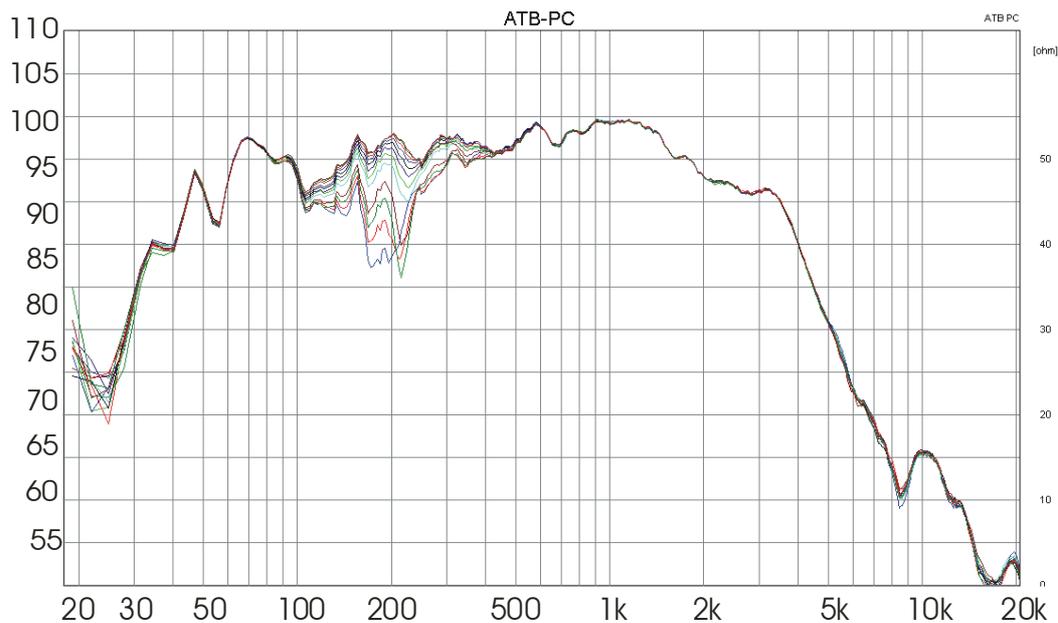
The delay is of great importance for the impulse response when designing the crossover for bass to mid-range. With the delay the time behaviour of the speakers is trimmed as such that the impulses add to each other. This results in a more natural sound. The pressure of the Bass is felt much stronger. The usual lifting of the bass in cars can be reduced.

For the determination of the delay for the bass we also measure the distance between speakers and listener with a ruler. The speaker with the shorter distance is the one delayed. If the mid-range is delayed to fit the tweeter, the delay is set back to 0. The correct value for the delay is searched for with test measurements.



To set up the delay near-field measurements are made. Here the microphone is situated between bass and mid-range at a distance of 20 to 25cm to the mid-range. The phase shift is set to 180° for the mid-range and 0° for the bass.

For the first measurement the time-shift is set to 0 for channel 3, in the DSP delay menu and measured with M. In the following measurements the time constant is raised in 0.4ms steps and measured with +M. As the difference between the curves gets smaller the time constant is raised in 0.2ms steps and measured with +M. The delay is set properly when the frequency response becomes even.



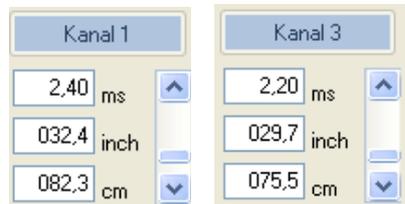
Pict. 16.1

The biggest breakdown in frequency response, blue curve, at the crossover frequency occurs when the delay is 0. In the following measurements the delay was made greater by 0.4ms steps and the breakdown gets less and less. The higher curves are closer together where by the curve with 2,2ms delay is the most even one.

If the mid-range has been delayed because of the tweeter, you prove if the value found for the Bass is larger or smaller.

The value is larger:

For the final DSP crossover set up following settings are made in menu delay:



This delay value of 2.2ms is set for the mid-range, Channel 3. For the tweeter channel 1, the delay is set to the 2.2ms + 0.2ms = 2.4ms. The 0.2ms is the adjustment between mid-range and tweeter.

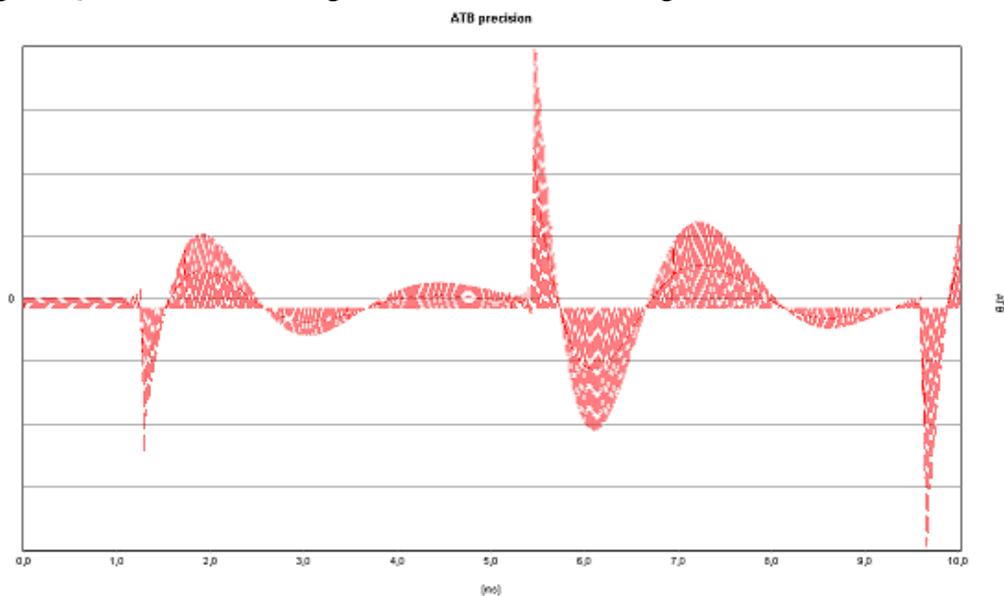
The value is smaller:

The delay value for the adjustment between mid-range and tweeter is set in the delay menu of the DSP.

The bass is set value of the adjustment for mid-range to tweeter minus the value from bass to mid-range.

17. The settings for the bass, subwoofer speaker

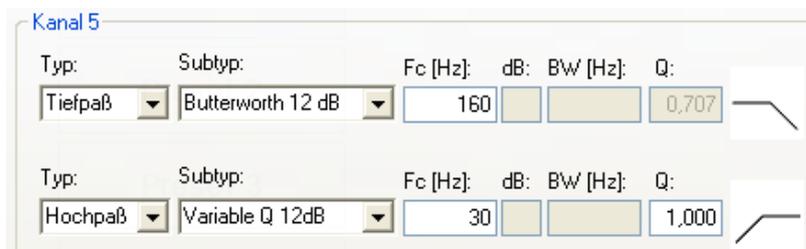
Usually the bass is boosted for listening behaviours in cars. The best impulse behaviour is achieved when boosting the bass by turning up the volume on the bass-amplifier. Here you have to consider the power displacement of the Bass as not to over-load it. The lower the frequency, the greater the hub of the speaker, which has its limits. To protect the Bass there is the high-pass function, or subsonic filter. Especially when boosting the Bass the subsonic filter should be used. As any filter influences the impulse behaviour, the filter can cause an over-swing, which in some cases is wanted as it can result in a more voluminous Bass reproduction. The Bass volume or over-swing is a function of the order, 6dB/Oct, 12dB/Oct, and the Q-factor of the filter. Filters with high orders and higher Q-factors result in higher volume or over-swing.



Pict. 17.1

The picture shows a subsonic high-pass 4th. order. This was set up in the DSP for demonstration purposes. The test signal is the square-wave described in chapter P1. You can see that the first impulse which is responsible for the dynamic of the stroke is hardly present. The signal consists of the over-swinging of the filter. The order and Q-factor of the filter should not be chosen too high.

The cut-off frequency of the filter depends on how big the speaker is, but should be higher than 20Hz. In the DSP menu active crossover/filter the subsonic filter is set up as high-pass.

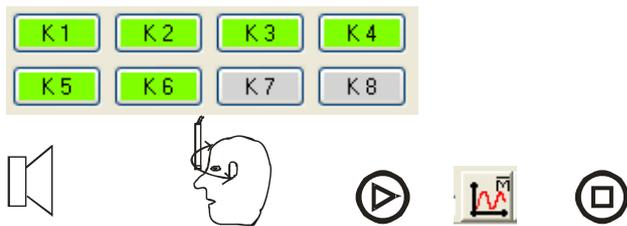


18. The settings for the passenger side

To produce a stable stereo and stage image both sides should be set to the same values. To do this the DSP has a copy function. As a rule the DSP is set up by the driver. Then the opposite side is usually made louder to compensate the difference in distance. In the case of a left steering wheel car channels 2, 4 and 6. But you can also compensate the distance by delaying the side closer (channels 1, 3 and 6). To do this all values on the one side are increased to compensate the running-distance.

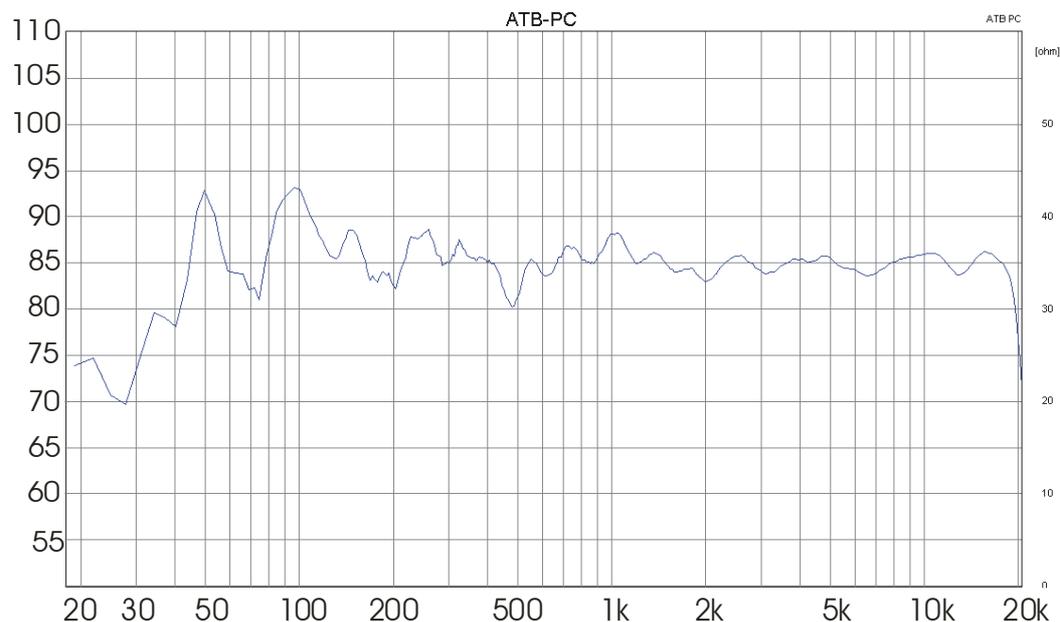
As far as these settings go the opinions differ. Here we need the experience of the technicians.

19. Setting up the equaliser



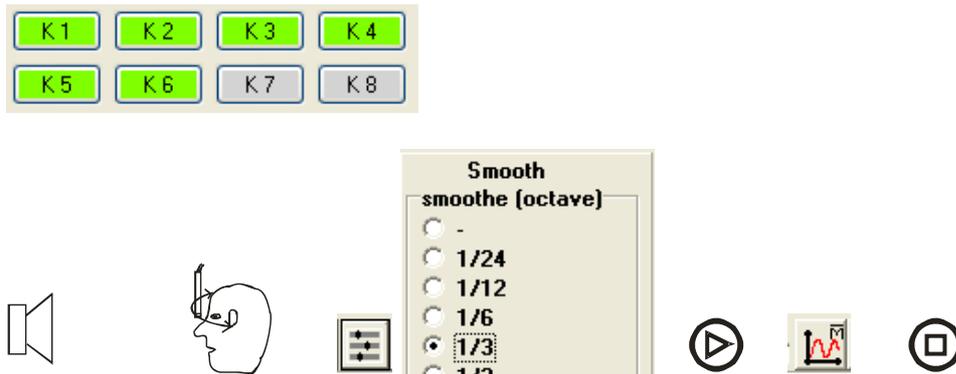
The equaliser is considered the great advantage of DSP. But as equalisation affects the time behaviour, impulse behaviour and phase of the speaker should be used carefully. The single measurements are of no use for making settings as the measurements differ strongly according to the microphone position. The heard frequency response, the sound impression is an average of the signals that reach the head area of the listener. And because the listener has, as a rule, two ears there is no correct microphone position for such measurements. The ATB has the measurement function mean measure which measures constantly and averages the results.

The measurement is started with Mean Measure and the microphone is moved slowly around the area where the head of the listener would be to measure what the listener would hear, with all speakers switched on. Because of the correct setting of the crossover the frequency response is nice and even. Any adjustments should be proven with a hearing test.



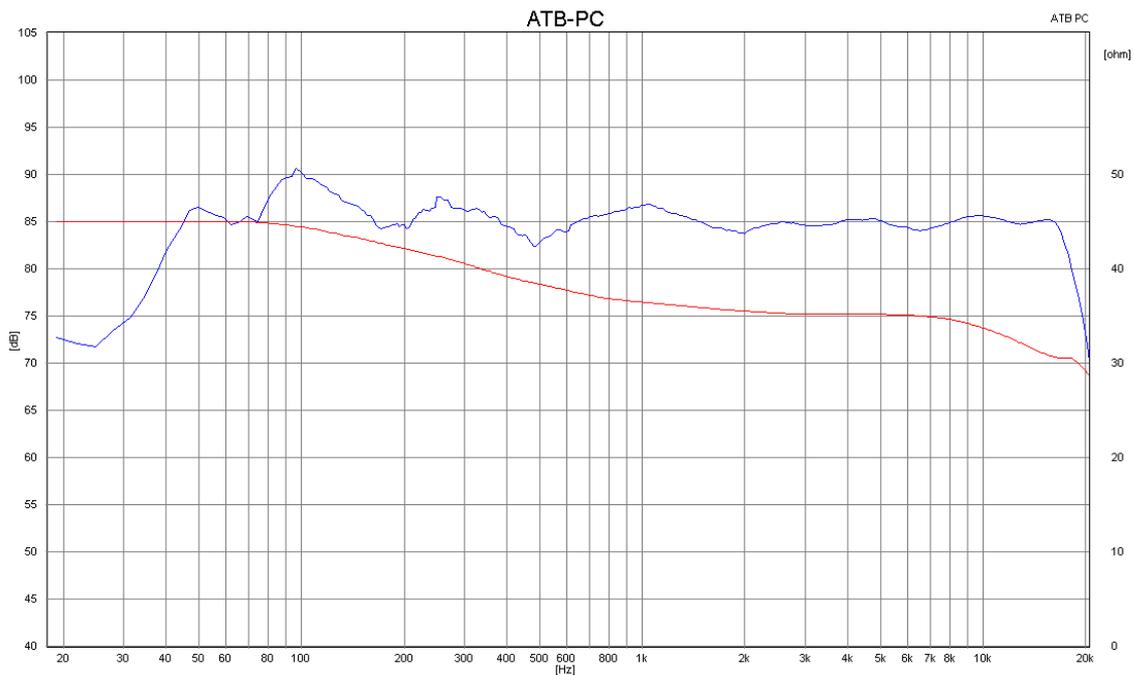
Pict 19.1

20. Adjusting to the Car&HIFI curve



The adjustment to the Car&Hifi is done with Mean Measure, moving the microphone slowly in the area where the listener's head would be. To adapt the curve to 1/3 octave, which is the usual spacing of equaliser filters, you choose 1/3 smooth in the ATB program.

To compare you load the CAR&HIFI curve from the installation CD-ROM. Click the curve in the load menu and load by load selected. The CAR&HIFI curve is then shown.



Pict 20.1

The adjustment to the CAR&HIFI curve should be done by changing the volume on the amplifiers and verified by listening. The CAR&HIFI curve is an average from several different vehicles and can strongly differ from that what you feel is correct for your car. With the equaliser room resonances are suppressed and cause bumps in the frequency response. Room modes cause breakdowns in the frequency response because of signal deletion. You can not compensate for deletion with an equaliser.

21. Considerations when setting up a DSP crossover, compendium

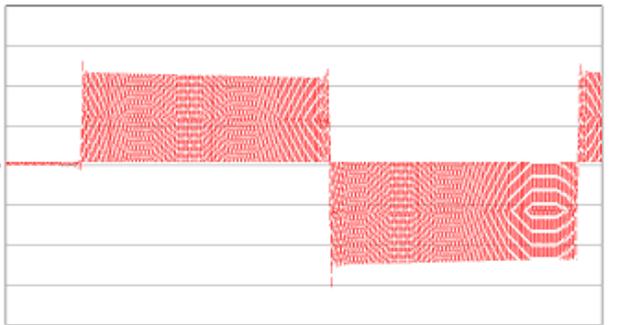
1. Programs that use pink or white noise as a measurement signal were developed for noise evaluation. The measurement curve changes constantly, so that the results are not precise. They are not suitable for setting up a high quality DSP system.
2. A linear frequency response does not mean an optimal time-behaviour and as such natural sound impression. Only when the polling and delay are correct you will get a meaning sound impression.
3. When setting up the filter you have to differ between cut-off frequency and transit-frequency. The transit-frequency is where both curves meet at -6dB. The cut-off frequency is according to science where the filter curve drops to -3dB.
For the Butterworth Filter the following applies:
Transit frequency $\times 0.77 = F_c$ cut-off frequency low-pass.
Transit-frequency $\times 1.3 = F_c$ cut-off frequency high-pass
4. The polarity and phase-shift 180° determines the impulse behaviour. They have to fit into the scheme.
5. The settings of the delay cannot be determined by measuring the distance. Such measurements are not suitable for optimal impulse behaviour because the phase-shift and acoustic C enters of the speakers can not be measured this way.
6. For the optimal settings the acoustical phase measurement is a must. This should be enhanced by the step measurement.

P1. The delay function

The following measurement-technical briefing of the delay function in section P is focussed on professional users.

The following descriptions extend the filter calculations of the loudspeaker development section by time-alignment. The frequency responses show correlate to simulations made with the Boxsim program. In the program the delay is set in menu “Baffle and Position” with “Mutual Enclosure” switched off with position of radiation as distance. The active filters are calculated.

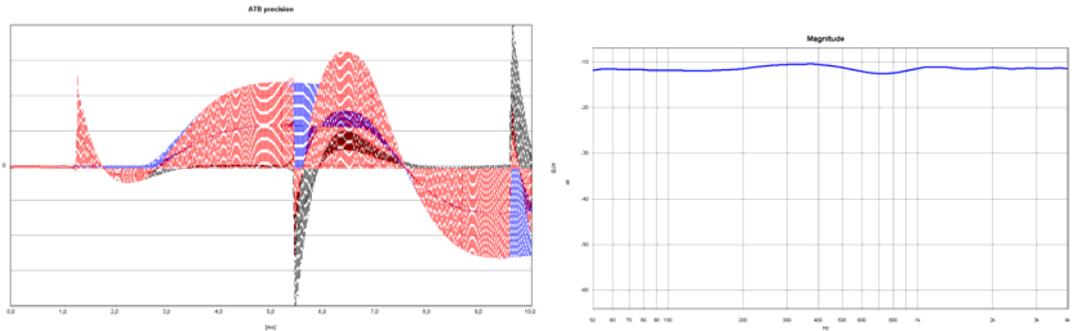
By the description of the time-behaviour a square-wave substitutes the signal from a guitar for the measurements. A square-wave is a combination of single waves just as a music signal also. Each wave has a frequency. The phase-shift and with that the time behaviour is clearly defined. When a square-wave runs through a filter the waves are sorted according to frequency. If a square-wave runs through a high-pass only the higher frequencies are shown. At the same time the filter affects the time-behaviour of the signal. With changes in the square-wave after filtering you can conclude on changes in the time-behaviour of a music signal.



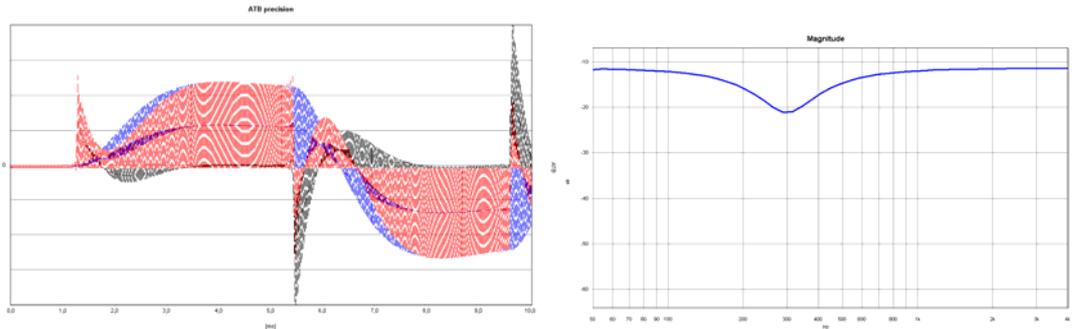
Pict P1.1

The picture shows the oscillogram of a square-wave. The delay at the beginning is due to the running time of the signal through the DSP crossover. The coloured surfaces symbolise the single waves.

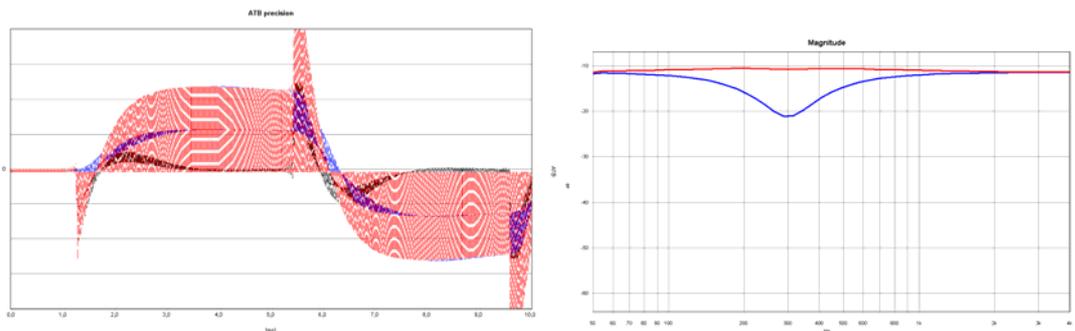
The following oscillograms are measured at the high-pass and low-pass outputs of the DSP. The red sum signal of high and low-pass is measured with an additioned circuit. The amplitude frequency response of the sum signal is shown.



Pict P1.2



Pict P1.3



Pict. P1.4

The oscillograms show the square-wave after filtering with the DSP. The frequency response curve shows the sum signal of high and low-pass. The filters have Butterworth characteristic of 12dB/Oct. The boarder frequency F_c for the low-pass is 154Hz and for the high-pass 260Hz, transit frequency 200Hz.

The first peak with over-swing is the high-pass filtered signal. The curve is black which is only recognisable when it does not belong to the sum signal. The blue signal that follows belongs to the low-pass. The sum of both signals is the red surface. This surface should show the ideal reproduction of the square-wave.

In picture 1.3 the difference in running length is compensated between the mid-range and tweeter using the delay. The tweeter is delayed with the corresponding difference of running length. The signals occur at the same time. The red surface shows the sum signal. This is quite small as the high-pass signal, black, and the low-pass signal are often anti poled which causes deletion.

These parts of the signal sum are non present in the sum signal. The frequency response curve also show the deletion. There is a dip at the transit frequency.

In Picture 1.4 the channel for the Tweeter is cross-poled, 180° phase-shift.

Here the sum is at maximum, as the signals for the high-pass and low-pass do not delete each other. Both signals are also transmitted at the same time. The sum signal greatly resembles the square-wave signal. The linear frequency response, red curve, also shows the optimum setting of the filter. These filters and phase-shift of 180° will be used in the description as settings for the DSP crossover.

The measurement is also to show that delay and phase-shift are two different functions. They cannot replace each other. The commonly found adjustable phase-shifter of active Sub-woofers modifies the phase of the Sub-woofer to Mid-range/Tweeter signal in relation to their steady-state. This will make the frequency response evener. The correct impulse behaviour though can only be achieved by compensation of the running distance with the delay.

Furthermore it shows that after the crossover with high and low-pass function the sum signal does not correspond to the time-behaviour of the input signal. Calculating the filter the signal shows a phase-shift correspondent with an all-pass filter. This is responsible for the change in time-behaviour.

On the other hand the frequency response of the sum signal is linear.

P2. Equalising the single speakers

Equalising is to compensate construction faults of the speakers. These show up in measurements in a measurement room. In a large room you can also carry out near-field measurements.

For the measurement a demo speaker is built into a baffle or enclosure. Measurements are taken in near-field with the microphone at approximately 10cm distance.

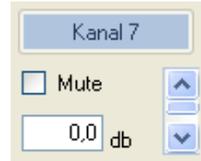
The DSP is powered by a 12V/20W power-supply.

Following settings are to be made in the DSP and ATB programs.

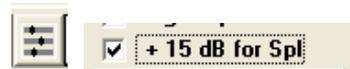
Main menu on/off



Menu Sound-level



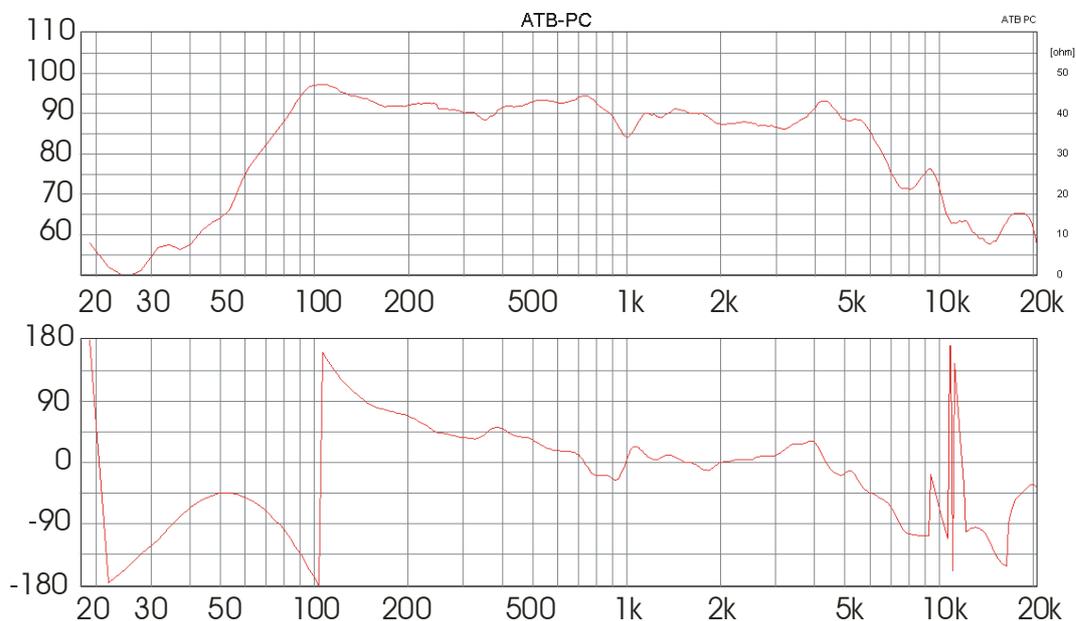
Menu Phase-shift



The loudspeaker is connected to the previously unused channel 7 over the amplifier.

In the ATB PC pro program SPL and Phase-measurement is set up.

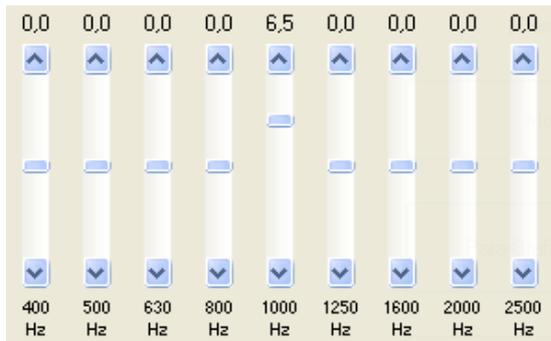
In the parameter menu smooth is set to 1/12 octave. Because of the high sound-level in near field +15dB is chosen for SPL. This measurement setting is taken back when setting up the crossover.



Pict.P2.1

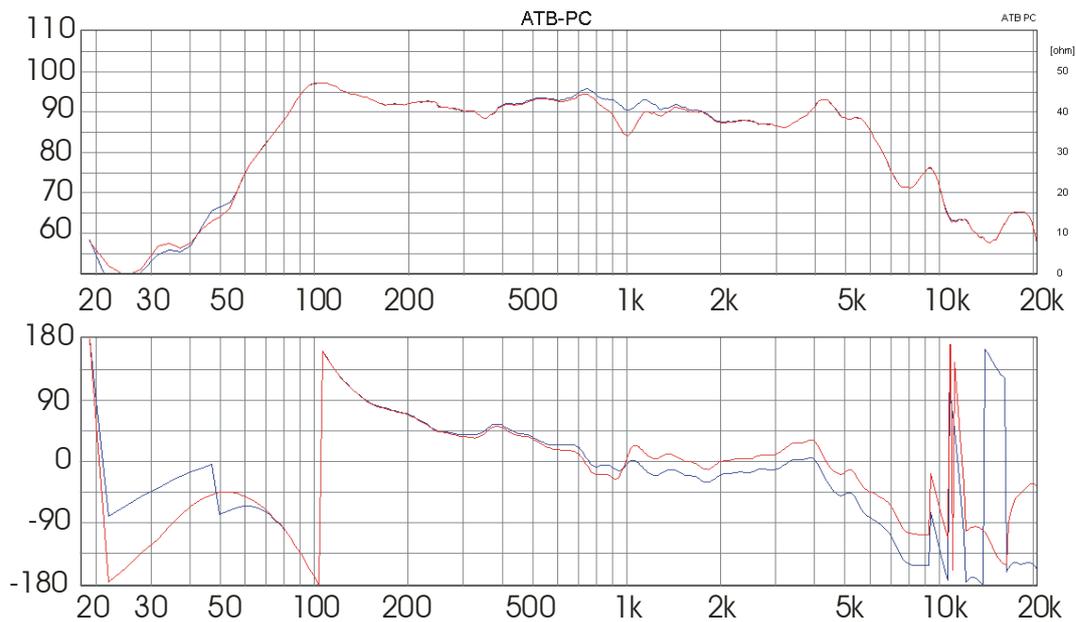
The red curve shows a big break-down in frequency response at 1kHz. At the same time the acoustical phase shows a jump at 1kHz. This is due to a construction fault of the speaker. Just this phase-jump can be heard, as the ear is very sensitive in the mid-range region, when the music loses its time consistency.

This break-down is compensated in the paragrahical EQ menu.



In the DSP menu paragrahical EQ channels 7&8 are set to an increase of 6.5dB at 1kHz.

The next measurement is started with +M .



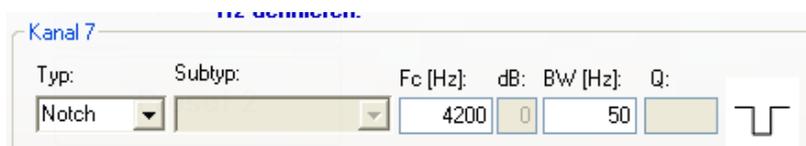
Pict. P2.2

The blue curve shows the frequency response with equaliser. The frequency response and phase is evened out. For the value of the increase an even phase is more important than an even frequency response. The phase delivers the information that the resonance is compensated.

Attention: The large phase jumps from 10kHz onwards are due to the signal being missing for calculation. The signal is too weak from that point on.

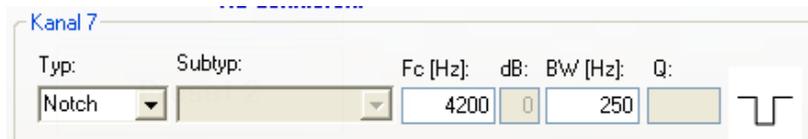
The equaliser settings are pre-set when making measurements in the car.

Furthermore the frequency response shows a peak at 4 kHz. The phase does show a brake point, because of the drag for higher frequencies, but isn't the typical picture of a resonance. To compensate peaks in the frequency response there is the notch filter function in the menu active crossovers/filters.

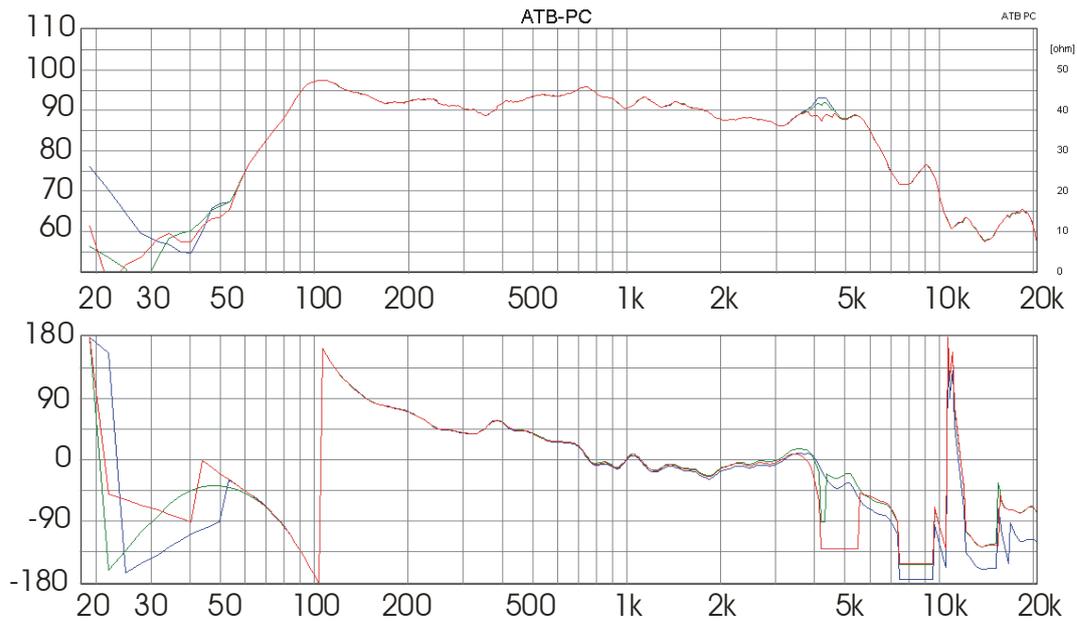


To even the curve, Type=Notch, FC=4200Hz and BW=50Hz is set. BW is the band width, the region where the filter is active.

The next measurement is started with M+ .
 As the peak only flattens a little with a BW of 50Hz, the band width is increased.



The band width is set to 250Hz and the measurement started with M+ .



Pict P2.3

With a band width of 50Hz, green curve, the frequency response hardly changes, but the phase shows a large jump. This makes the sound impression uneven and more in direction of getting on your nerves. The red curve with settings, Type=notch, FC=4200Hz and BW 250Hz, shows a more even frequency response, but very large phase jumps.

As the speaker is driven in combination with a tweeter the peak would be suppressed by the crossover.

P3. Adjusting the delay of the tweeter using ATB PC Pro

Setting up the delay has to do with time-alignment. Several acoustic engineers consider this as being irrelevant. Amongst other things the understanding of time dependant occurrences is blocked by measuring the time-behaviour with the cumulative-decay (Waterfall diagram) method. On the Z-axis of the 3D graph the time is displayed in ms (milliseconds). This is not physically correct, as the time-behaviour is not displayed as in a normal oscilogram, but is the frequency response at different stages in time. At time is zero on the Z-axis the frequency response is displayed as in a normal frequency response measurement. There is no swing-in; no lag caused by crossovers and even a delay is not displayed.

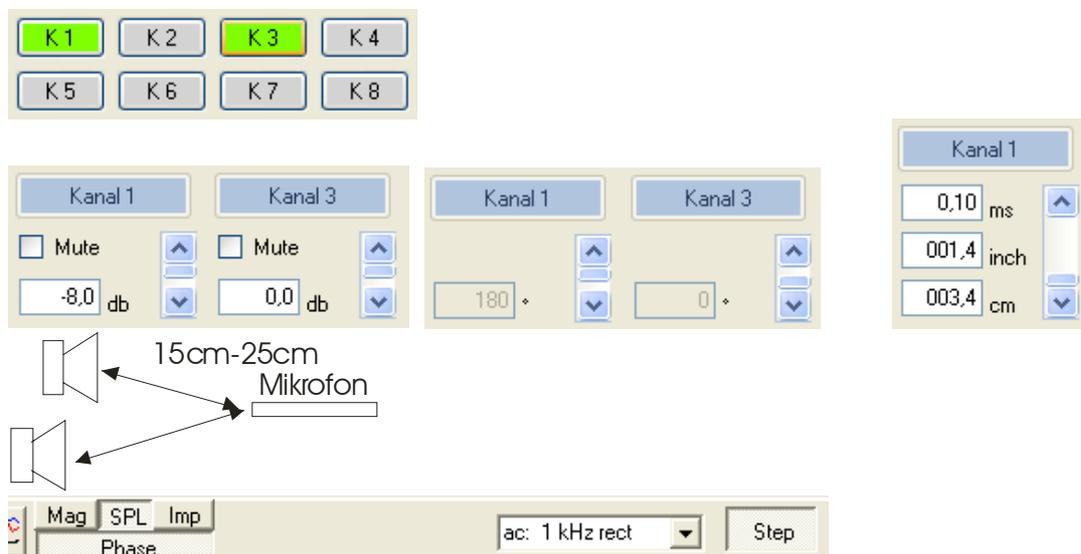
In chapter 12 the setting up of the delay per frequency response measurement was described. There is always danger of inaccuracy here because the frequency response can look even although the delay is set incorrectly. With use of the step and phase measurement function of the ATB PC Pro the correct set up is clearly displayed.

The Step function is an in-time display of the loudspeaker signal. The X-axis shows the time in milliseconds and the Y-axis the amplitude, scaled to maximum height. The ATB program calculates the Step from the amplitude and phase of the signal. Because of that the display is clearer.



With start-f the time axis for the Step is chosen in ms.

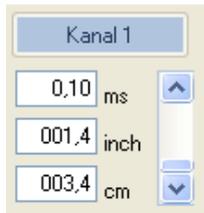
The user should consider that at the range of 1kHz the phase is around 0°. So that phase measurements of tweeters with crossover the signal is too weak and not clear. For the measurement tweeter and mid-range are cross poled. After determination of the delay the settings are set back according to the polling scheme.



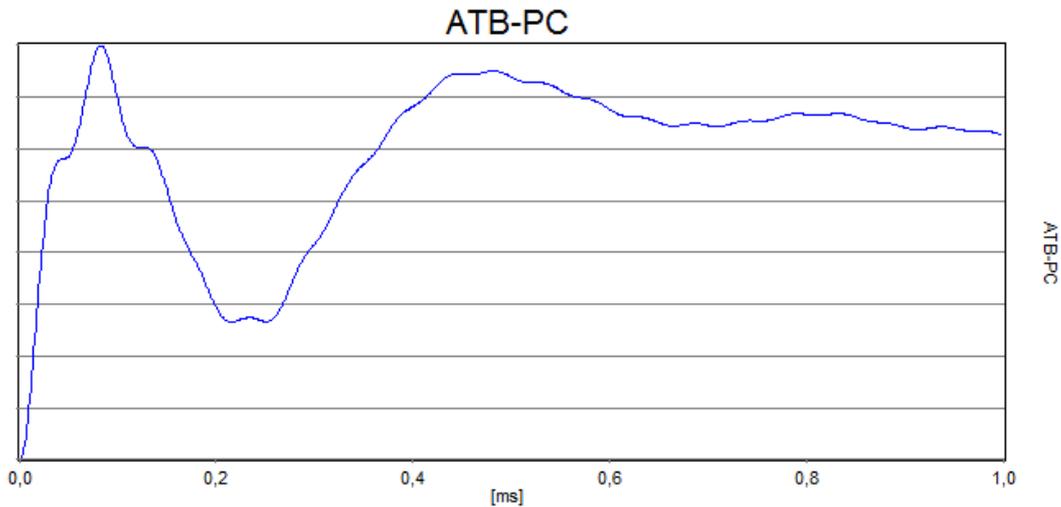
Both speakers are switched on. The microphone is held at a distance of 15 to 25 cm between the speakers. It is aligned between the middle of the two speakers and the hearing position.

In the measurement program Step and Phase are activated in addition to SPL.

As a demonstration for a to short delay of 0,1 ms

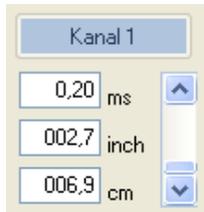


is chosen. The measurement started with M .

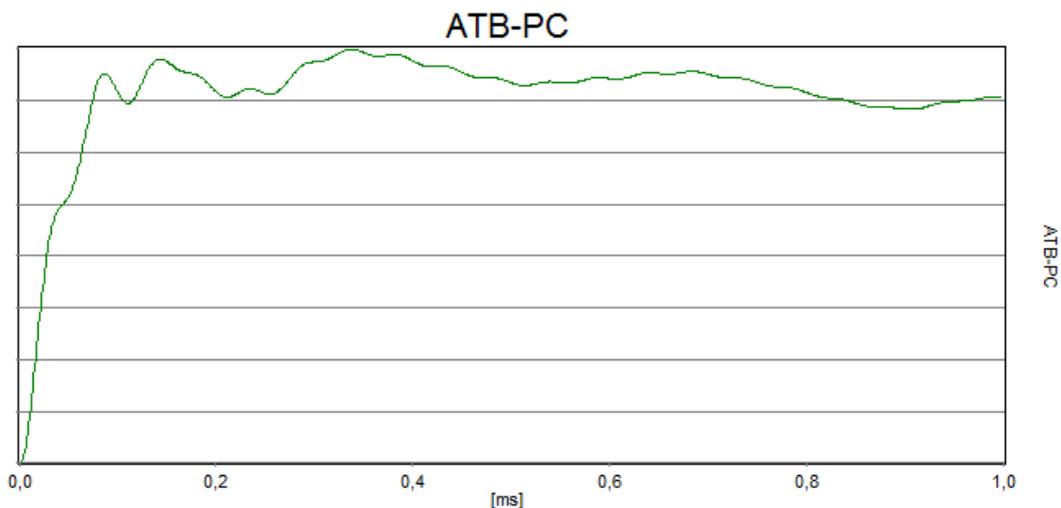


Pict. P3.1

The picture shows a curve with two peaks. The first peak is the signal of the tweeter this reaches the microphone first. After the dip a second peak which is the delayed signal of the mid-range. The lowest point of the dip between tweeter and mid-range is at 0.10ms. The 0.1ms are added to the pre-set 0.1ms and set up in the delay.



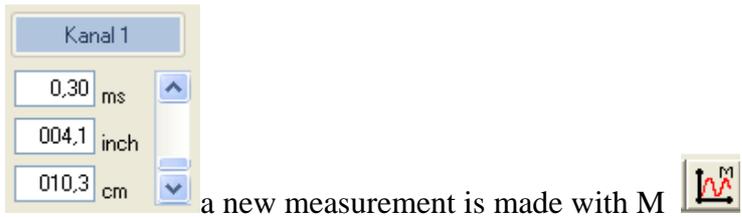
a new measurement is made with M .



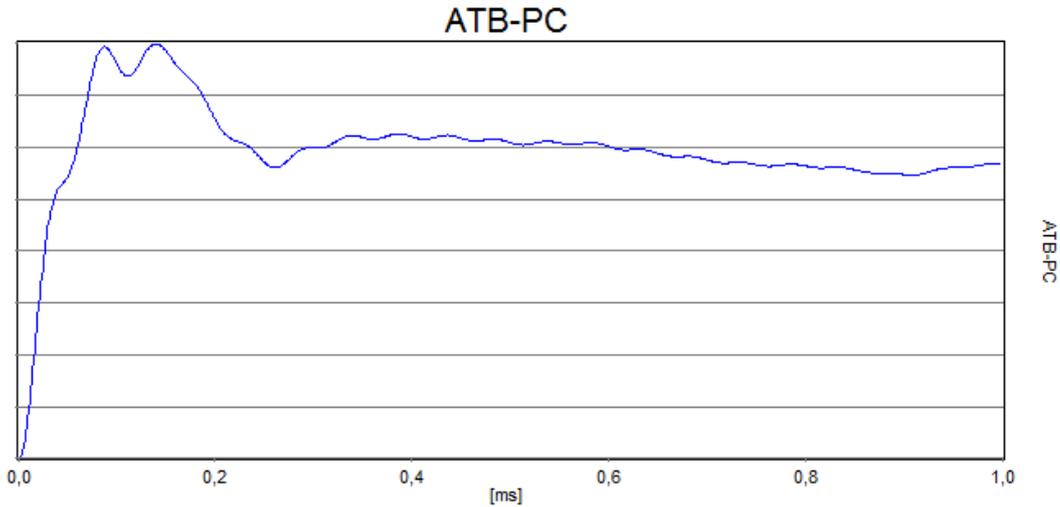
Pict. P3.2

The Step-curve shows a climate and no further peaks. Mid-range and tweeter reproduce the signal at the same time. The single signals have become one, the time relationship is correct.

After that the delay is prolonged and set to 0.3ms



a new measurement is made with M



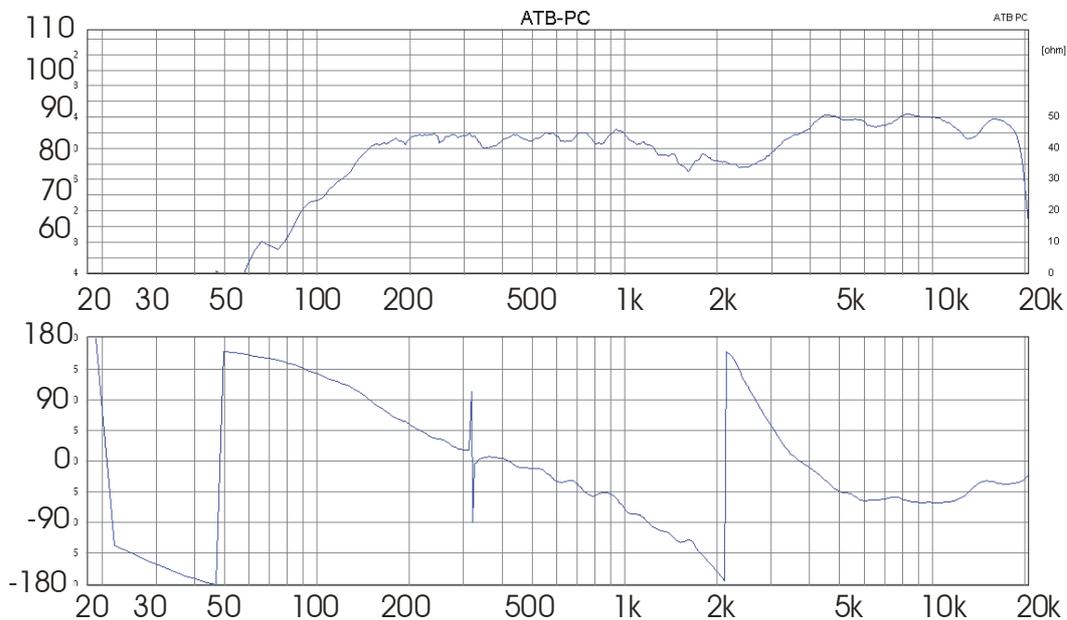
Pict. P3.3

The curve shows two peaks, a too long delay. The first broad peak is that of the midrange and the following peak the tweeter, which is less dominant.

Frequency response and phase of the delay set-ups

The correct set up for the delay is also shown in the frequency response and acoustic phase.

To display Step is switched off.



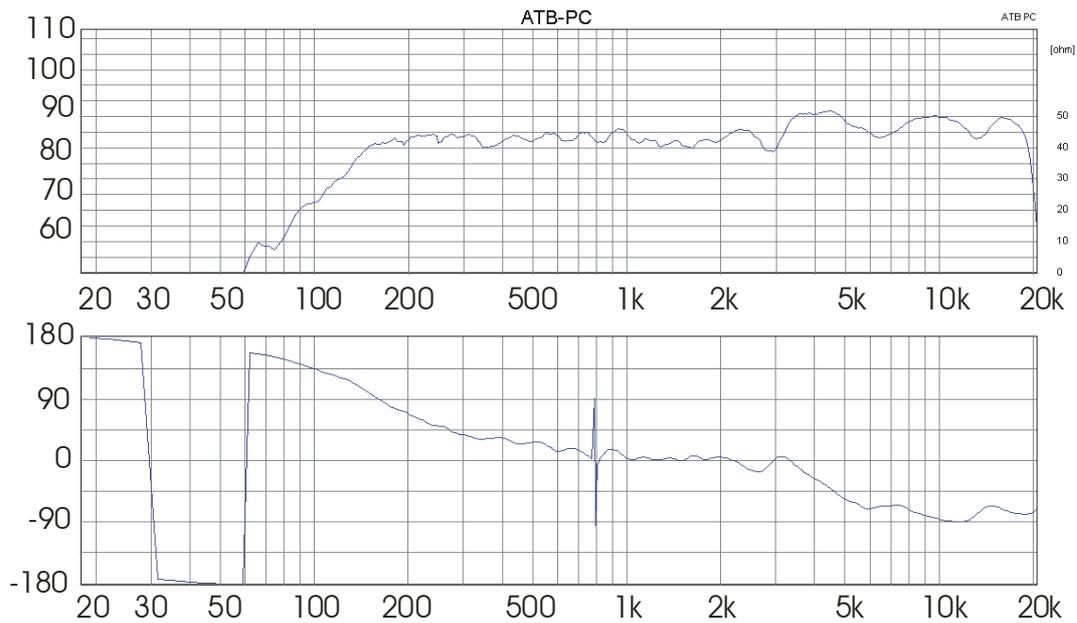
Pict.P3.4

Without delay the frequency response curve shows a break-down at transit-frequency.

The phase springs at transit-frequency.

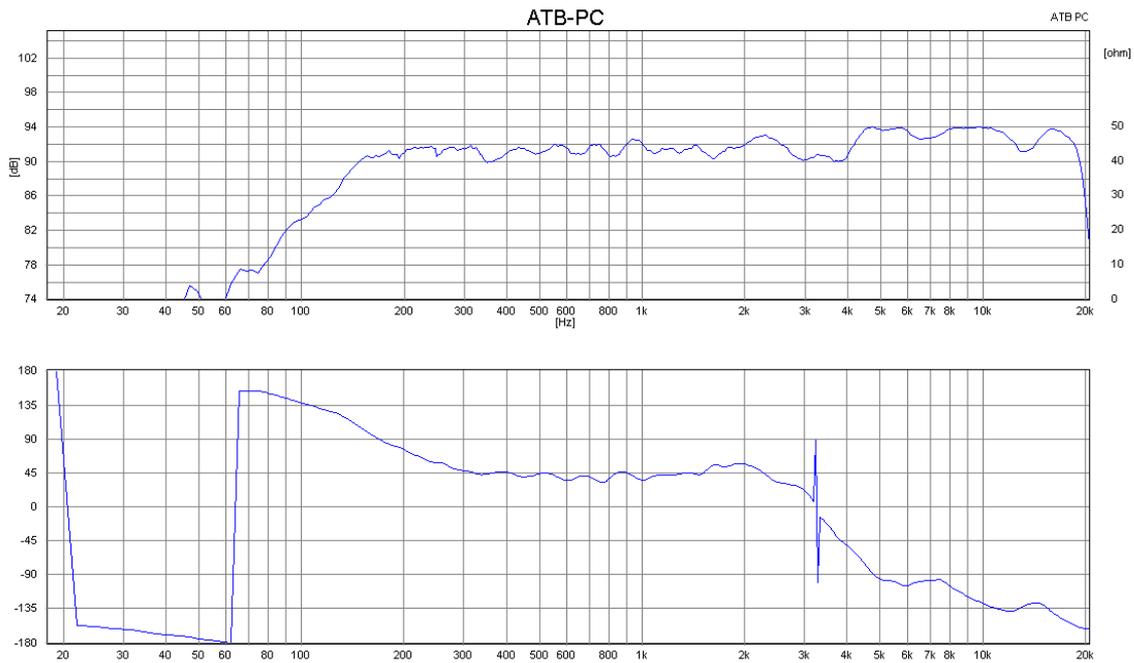
Attention: The jumps in the phase curve from 1kHz onwards are due to the display range of 180° to -180°. When the phase is for instance -185° it cannot be displayed continuously because the graph has a maximum of 180°. The -185° causes a jump of 360°. The -185° are displayed as 360° - 185° = 175°.

The large jumps at 300Hz are due to calculation and have no effect on the sound reproduction.



Pict P3.5

With the correct value of 0.2ms for the delay the frequency response becomes even. Also the acoustic phase describes a straight line at transit frequency.

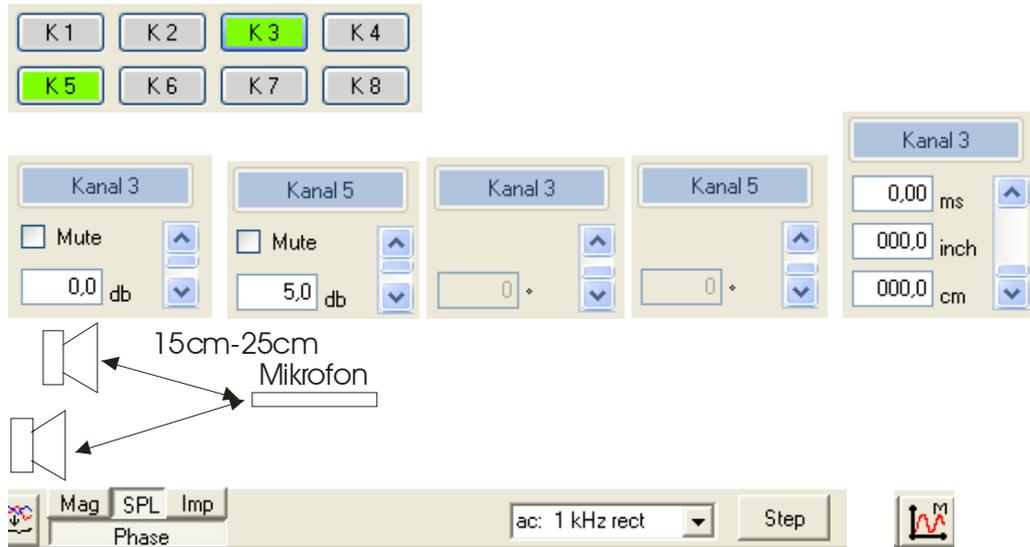


Pict. P3.6

The too long delay of 0.3ms causes a break-down at the transit-frequency. The phase of the tweeter drops because its signal comes in too late.

P4 Setting up the delay for the mid-range with ATB PC Pro

For the crossover between woofer and mid-range the delay is very important for the impulse reproduction. With the delay the time-behaviour of the speakers are adjusted to each other as such that the impulses add up. This way the sound becomes more natural. The pressure of the bass can be felt more strongly. The usual strong boost of the bass in cars can be reduced.

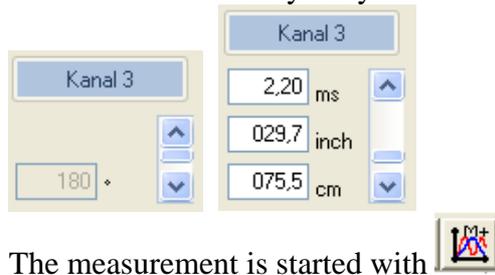


For the measurement both speakers are switched on, phase-shift and delay set to zero. The measurement microphone is held 15 to 25cm between the two speakers in a line with the hearing position.

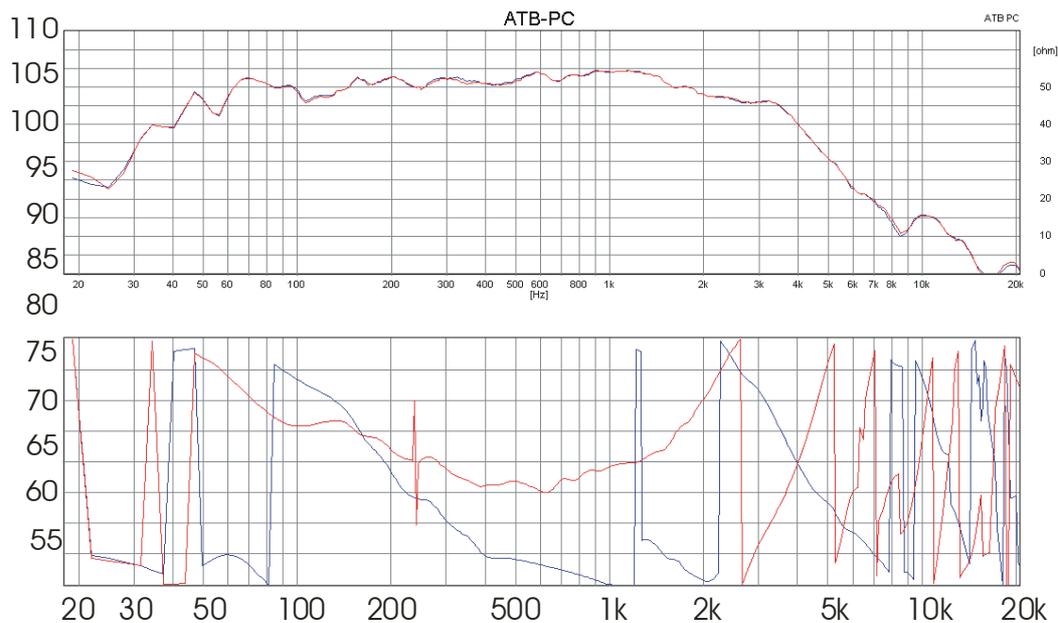
In the measurement program phase is additionally activated to SPL measurement.

For the following two measurements the delay is set to the values from chapter 16.

During the measurement the speakers are cross-poled in accordance with the polling scheme and time delayed by 2.20ms.



The measurement is started with

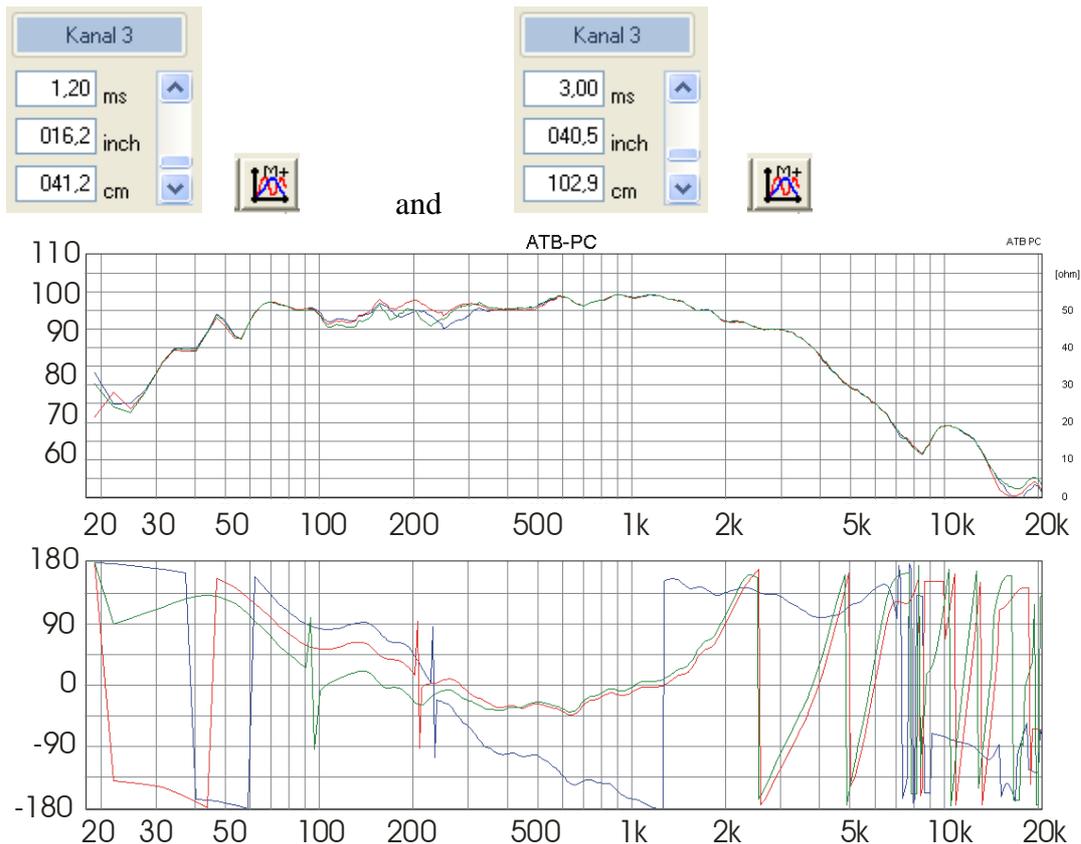


Pict. P4.1

The measurement shows the frequency response and phase in near-field between midrange and woofer. The blue curve shows the frequency response of the speakers without phase-shift and without lag through the delay. The frequency response is even, but the phase shows a big jump at transit-frequency. The red curve is with a phase-shift of 180° for the mid-range and a delay of 2.2ms. The frequency response is also even, but the phase-jump at transit-frequency is very small. As the phase shows the time alignment of the sound-waves, the curves of the woofer and mid-range (Red curve) now fit together much better. Because of this the bass gains more pressure and impulse. The bass is simply experienced.

To check the result measurements are made with a bigger and smaller delay.

The measurements are the same as in the red curve of picture P4.1 and can be applied. The further comparing measurements are done with following values for the delay:



Pict.P4.2

The red curve with a delay of 2.2ms shows the most even frequency response with no break-down between mid-range and woofer and no phase-jumps.

The blue curve shows with its long running after phase the too long delay of 3ms. The green curve shows the on running phase of the too short delay of 1.2ms.

In the phase measurement the +/- peaks at 200Hz are not caused by the crossover. Attention: The jumps in the phase curve from 1kHz onwards are due to the display range of 180° to -180°. When the phase is for instance -185° it cannot be displayed continuously because the graph has a maximum of 180°. The -185° causes a jump of 360°. The -185° are displayed as 360° - 185° = 175°. The phase-jumps from 6kHz onwards are due to there being no signal to calculate from there. The frequency response of the mid-range has dropped too far. The jumps at 100Hz and 200Hz are due to calculation and have no relevance.