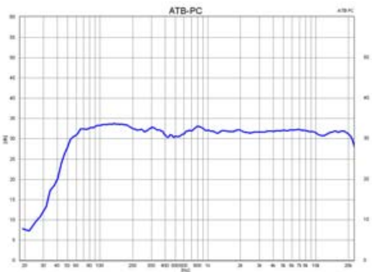
Photo Story
Speaker development, using ATB PC Pro

The as Theory marked articles give the physical basic theory to the relating themes.

1. The Analog.on Nugget

The Analog.on Nugget is a small High End speaker for Hi-Fi, Studio und Surround. The ideal connection values and the special bass tuning predestine the Nugget for use with digital amplifiers. And is therefore also well suited for plasma and LCD televisions.

Design study of the Nugget (Gold nugget)



Super linear frequency range

The extreme price worthy components

1.1 The Components

The speakers

The developed crossover

casing made of stone?

2. The Team

From the Technikerschule Braunschweig, Thomas Mohring, and Markus Sievers and Ralf Freymuth

The trainee
Jürgen Knoop

The Engineer
Leo Kirchner

Thanks very much to my colleagues and their distinctive articles.

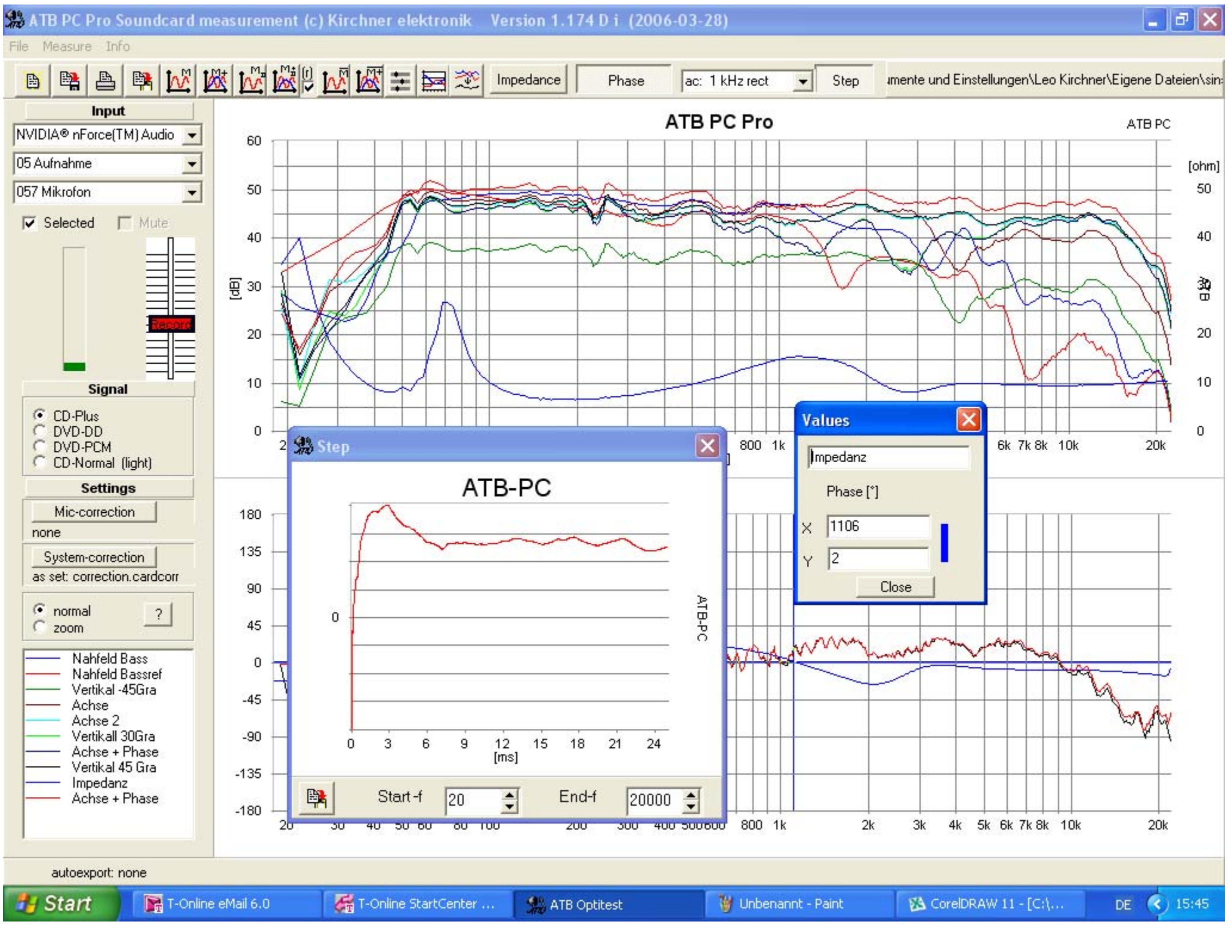
Ralf Freymuth with basics of acoustics
Thomas Mohring with *Sspeakers Choice* and *Thiele-Small*
Markus Sievers with crossovers
Jürgen Koop with the *Thiele-Small calculator program*
Head of project Dipl.-Ing. Leo Kirchner

3. The measuring system ATB PC Pro

Theory W1. Theory of acoustic measurement

For the development of the nugget the eatest of measuring equipment came to use. All measurements were carried out using the ATB PC Pro. With the ATB PC the signal for the measurements are taken from a CD or DVD player. The measurements are carried out with a PC, with use of a soundcard to record the measurement values.

The ATB PC Pro is a tool for developers. Therefore the usage must be as easy as possible so that the technician can concentrate solely on developing. This is achieved through the self explanatory windows based measurement program.



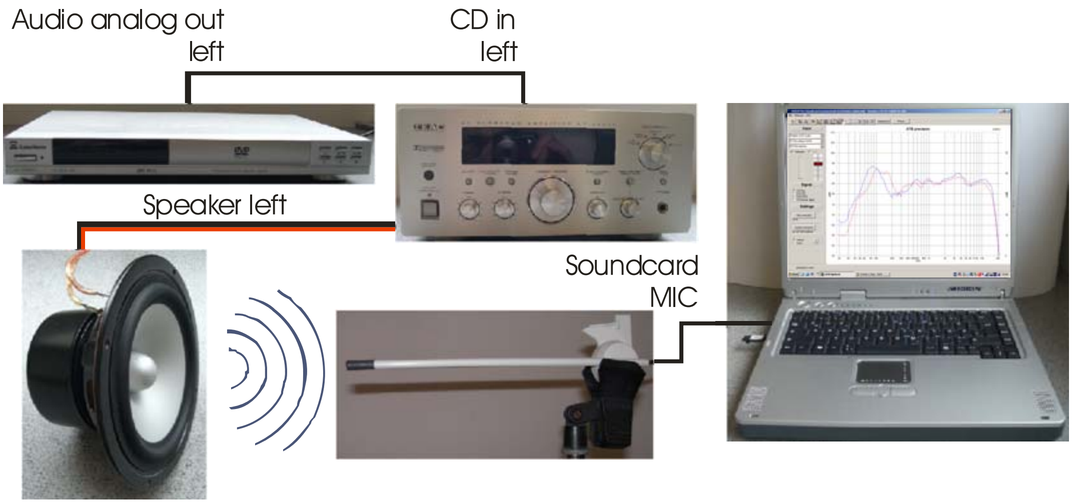
In this picture sound pressure level, impedance, phase and impulse are shown all at once. The single measurements are clearer.

System-correction

The System-Correction tests the frequency and phase range of the soundcard used. Resulting in correction files that assure high measurement accuracy.

Setup for acoustic measurements

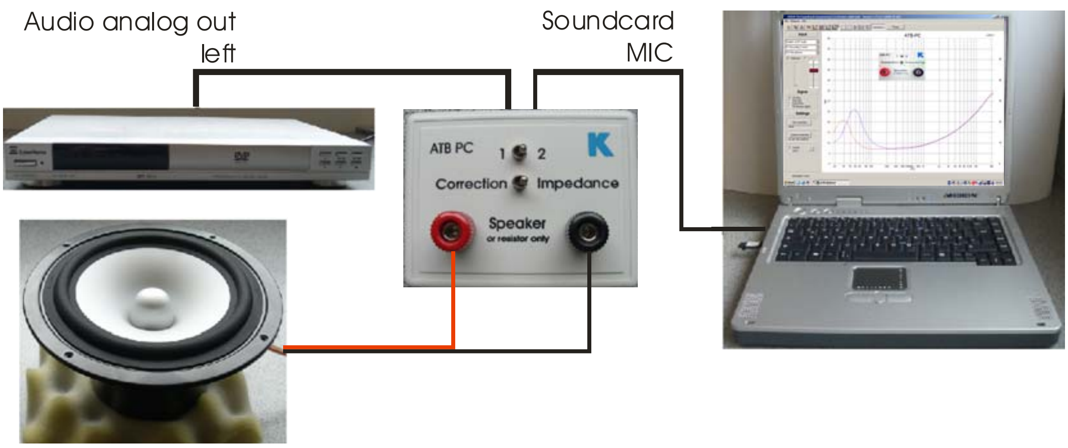
SPL



This is the setup for measuring the SPL (Sound-Pressure-Level) frequency and phase of the speakers.

Setup for electrical measurements.

Impedance



This is how the impedance is measured according to frequency and with phase and level.

4. The Speakers

Theory: W2. Speaker choice

The low-mid range woofer W 130 AL8

Speaker W 130 AL8
Aluminium membrane, low resonance Navi- form

double magnet,
magnetically screening

steel cage,
less resonance and more stability than alu-cast

The tweeter GRT 85

Air-Motion tweeter,
Invented by Oskar Heil, has proved it self since 20j in ESS speakers

folded membrane
with printed conductors makes the membrane very very light fast and clean

neodyn magnets create the strong magnetic field for high precision reproduction

5. The Enclosure

For the development of the enclosure the following points are important:
1. The size of the enclosure is determined by the woofer.
2. Speaker placement is as such to linearise the phase.
3. The inner geometry is to avoid resonance waves in the enclosure.
4. The bass reflex port should be optimize.
5. The enclosure casing should be resonance free.
6. The overall appearance should fit well to any interior.

5.1 Thiele-Small

Credit. ATB_TS_Tool.zip Thiele-Small Calculator

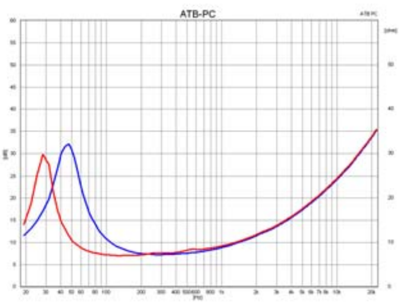
The enclosure volume is determined by the woofer.
To calculate the enclosure, the Thiele-Small parameter is needed.

To measure the impedance the speaker is laid or held on a piece of foam.

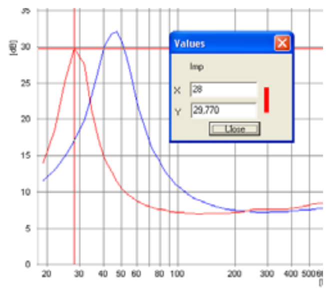
The weight is weighed to an accuracy of 2 grams

and evenly fixed to the membrane.

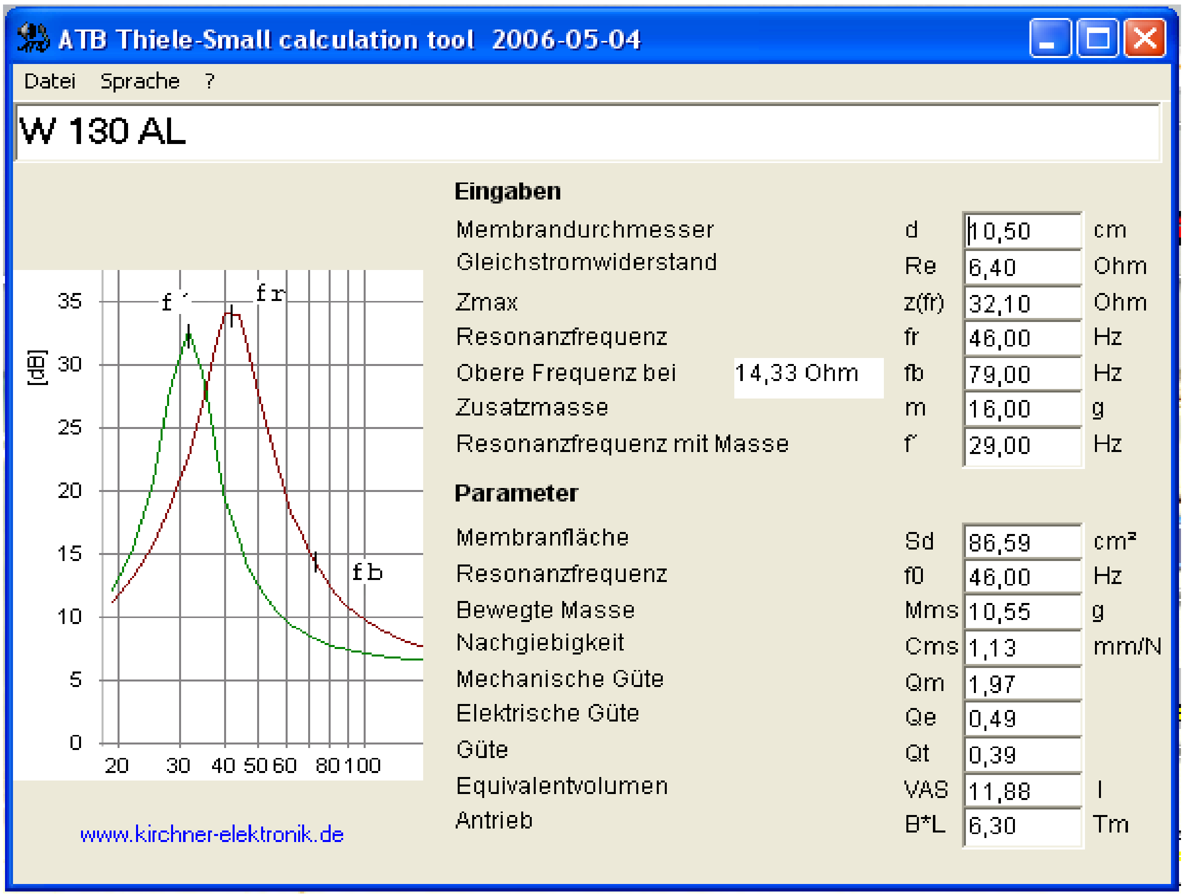
The membrane diameter is mid suspension to mid suspension.



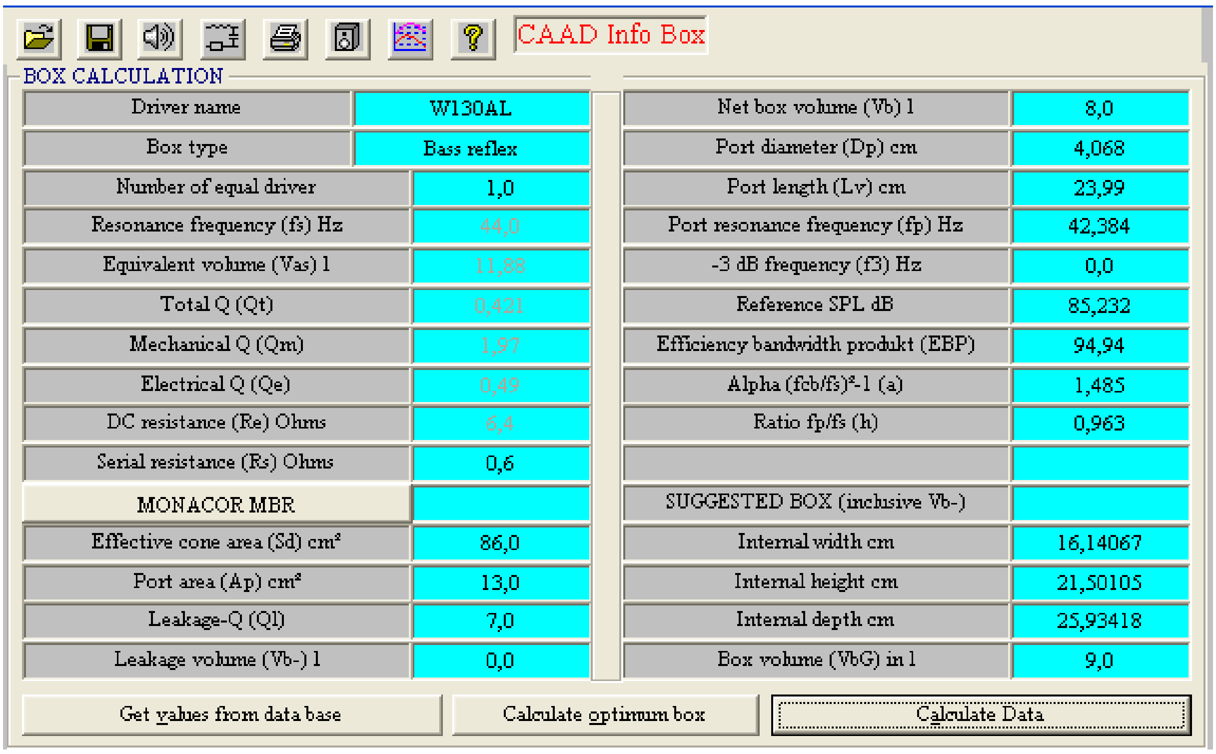
Impedance measurement
blue = with out weight
red = with weight



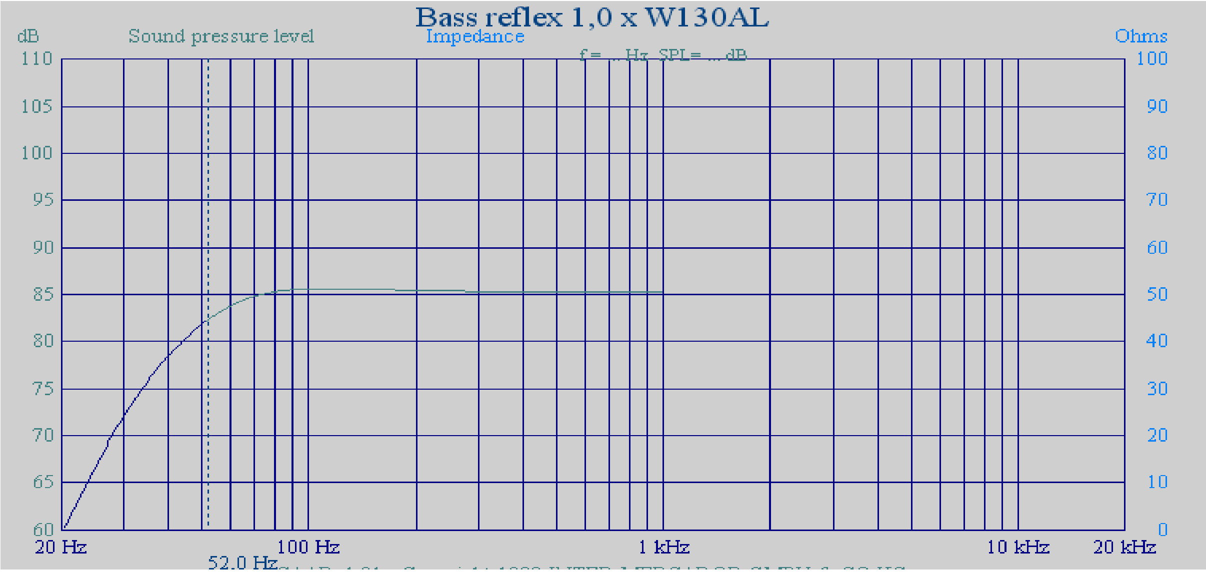
The values are taken by courser. Where by the average is used.



After the values are read into the computer, the speaker parameters are calculated automatically. The parameters are almost those given by the manufacturer.
To calculate the ideal enclosure the data is read into a CAD programme for speaker enclosures.



The volume was calculated to be 8 liters.



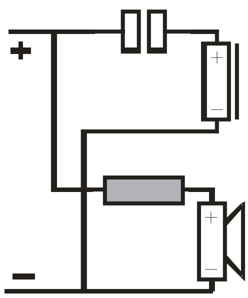
The calculated frequency response shows a light raise, of 1dB, at about 100 Hz. This is the right tuning for small speakers, as far as listening is concerned. The comparison of the measured and the calculated frequency response show a great deal of similenty. With a low cut-off frequency of 42 Hz the Nugget is a grownup speaker that is even good for orchestral works.

5.2 The Phase adjust

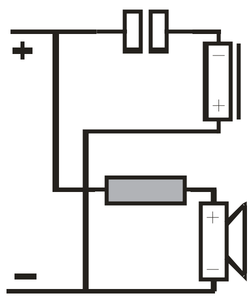
Theory: W3. The Phase

Finding the phase correction for optimal timing balance.
The correction takes account of the acoustic center of the speakers as well as the turn of phase caused by the crossover.

The speakers are mounted on test casing

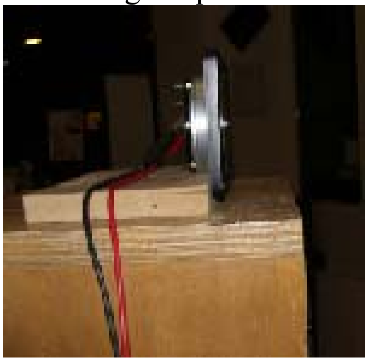


The speakers are connected to the 6 dB crossover

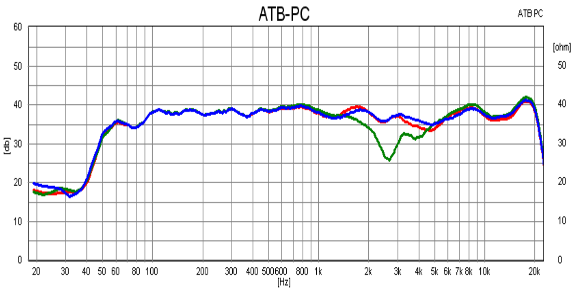


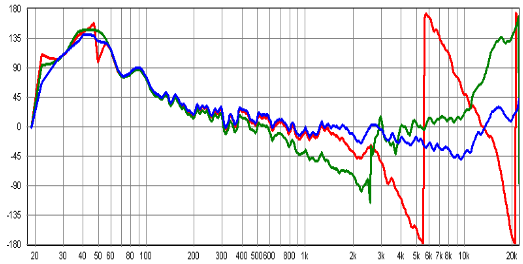
The tweeter is poled around as the phase correction would be 9 cm by otherwise

Measuring the phase.



1. Position, red 2. Position, blue 3. Position, green





Amplitude response of the Tweeter in different Positions

The Phase
1. Position red = late
2. Position blue = linear
3. Position green = early

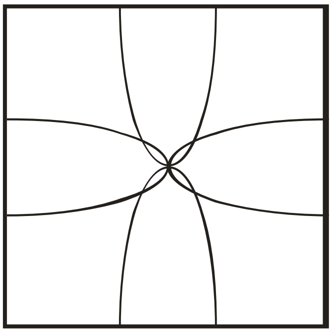
The position of the tweeter is determined with use of the pink noise signal(track 7 of the audio test CD) per hearing. The phase is right, when you hear the noise as if it is between the two speakers, in acoustical terms that is. This method is accurate to the mm. Both speakers then build one acoustical source.

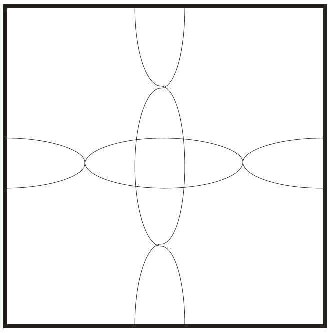
5.3. The standing waves

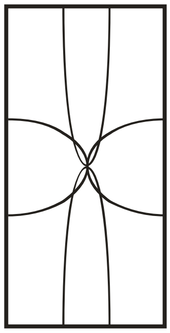
Theory: W4. Basics of the acoustic

Sound waves in a room, here an enclosure, build standing waves or modes. They are the resonance sound waves in an enclosure. With speakers that have stiff membranes the resonances have a back effect on the speaker. This causes additional time delay on the sound waves of the speaker and distorts the sound reproduction.

The modes in the enclosure with the dimensions of the finished box. The sound pressure is shown which is highest on the sides. The speed is highest at the knots.







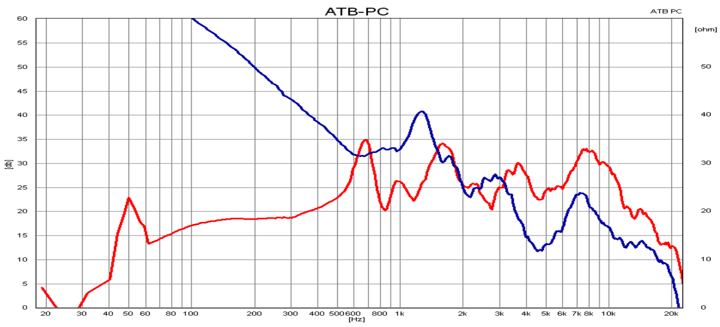
1. Mode at $\lambda/2$
Height of box 25cm
Frequency of mode
F=340 m/s x 1/0.25
= 680 hz

2. Mode, f = 1360Hz

Mode, side direction, f = 2,8kHz.
The Frequency can simply be
damped by dam material on
the side walls.

Measurement of the modes in a test casing with the dimensions of the calculated enclosure. The box was closed for measurement!

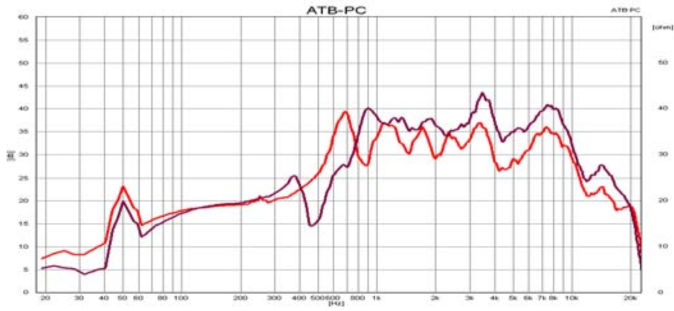
Central positioning of the
Microphone or pressure-
Level sensor
In the enclosure



Frequency analysis
blue = Measurement microphone is pressure sensor => less pressure with 1. Mode, higher pressure with 2. Mode
red = pressure level sensor => high speed at 680Hz

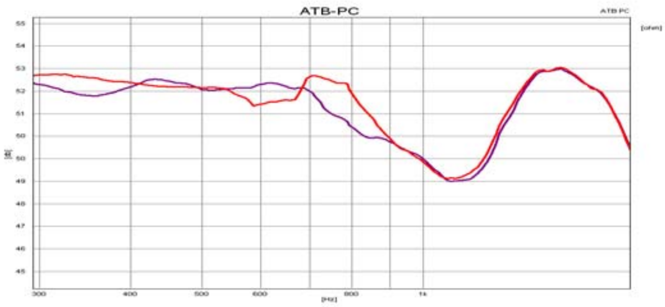
The resonances are baffled with a board diagonal to the main axis.

Position of the board for baffling resonances
1. Mode



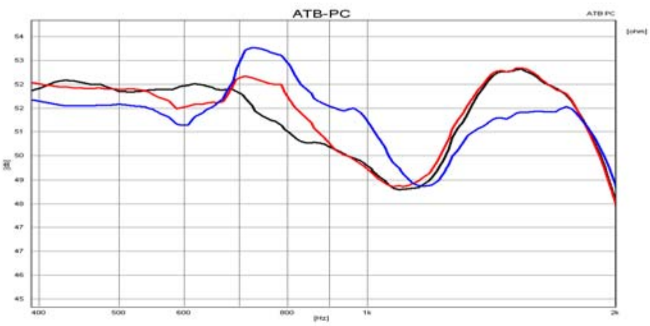
measurement in casing
red = without board
purple = with board. 1. mode has gone and the response is smoother

Near field measurement 1 cm in front of the speaker shows the back effect of the standing waves on the speaker



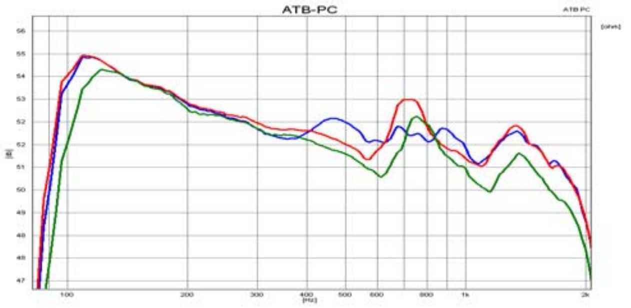
measurement 1 cm in front of the speaker
the red curve shows the effect of the resonance at 680Hz

The board was positioned at different angles. Only one angle suppresses the resonance.



measurement 1 cm in front of the speaker
blue, opening the reflex tunnel amplifies the resonance. The black curve shows the ideal angle for the board

Furthermore the influence of rounded casing sides is looked into:



This picture shows the setup for measuring the effects of rounded side walls. The casing is closed

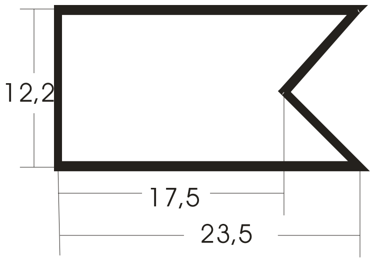
red = without board, blue = with board.
green = rounded sides have no advantage, but because of the smaller volume, there's less bass.

The sound distortion due to standing waves has less to do with unevenness of the amplitude over the frequency, rather more it's an indication of the time delayed parts of the sound reproduction.

5.4 The Bass reflex tuning

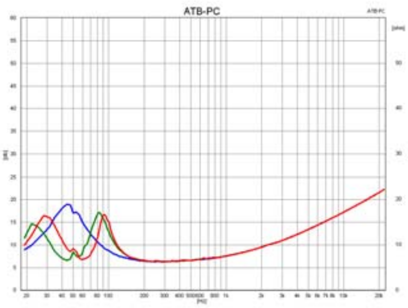
When constructing the bass reflex, the following is to be observed:
1. The opening must be on the front, otherwise putting the box on a shelf will not be possible.
2. A tube is not be used, as a long tube has resonances in the mid tone frequency range and will cause distortion.
3. An opening on the front must not let through mid range sound. That would make the sound soapy due to the time delay in comparison to the direct sound from the cone.

The construction chosen for the nugget has proved it’s self many times before.

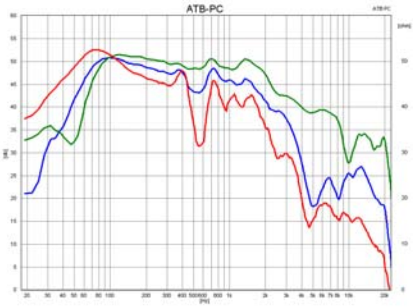


The triangular cut out on the reflex board acts as an acoustic low pass. This as such is a mid range filter and the board is also built in slanted to reduce standing waves.

Board on angle



Impedance measurement
blue = woofer without casing
green = board level, tuning to low
red = proper angle, the maxima of the impedance curve are equal



Near field measurement
blue =between woofer and opening
red = in front of opening
green = in front of woofer

The near field measurement shows the opening has it maximum sound level at the falling flank of the woofer. The blue curve is the addition of the opening and cone sound levels and is well balanced. The measurement reveals an ideal tuning.
The complicated theory of the Thiele-Small parameters is very helpful for construction as far the volume goes. But to tune the reflex opening it’s still not complicated enough.

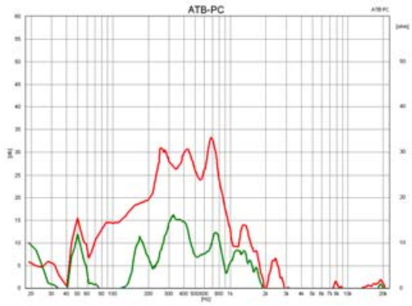
5.5 Low resonance Enclosure

Theory: W5. Acceleration sensor

The casing walls that are sent into vibration from the speaker distort the sound reproduction. That's why materials are used for building the casing that hardly get put into vibration, for instance stone. Reasonable though is the use of wooden casing with stiffening. To place the stiffening properly the casing vibrations are located by measurement. To do this you use an acceleration sensor. The acceleration sensor only measures the vibrations and not the sound level.

The acceleration sensor is connected to the mic-input of the soundcard.

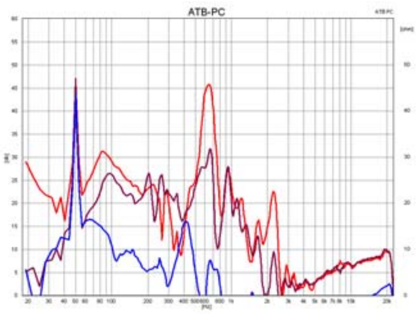
The sensor is taped on to the side walls of the casing with two sided sticky tape.



Vibration measurement of the casing side walls.
red = without stiffening
green = side stiffened by the slanted board.

Vibration measurement on the connection plate

Mounting of the vibration sensor on the connection-plate



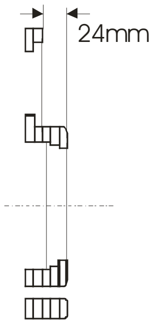
Frequency analysis: The peak at 50 Hz is a supply disturbance.
red = Connection plate without damping
violet = damped with tar
blue = Connection plate mounted with acrylic sealing

.

The measurements on the connection plate make it clear how important such measurements are! Damping with tar coated damping plates, which are often considered a high-end solution has little effect. Far more by gluing the connection plate to the casing with a 5 mm coat of acrylic sealing, the connection plate becomes acoustically dead and doesn't influence the sound any more.

5.6 The Enclosure construction

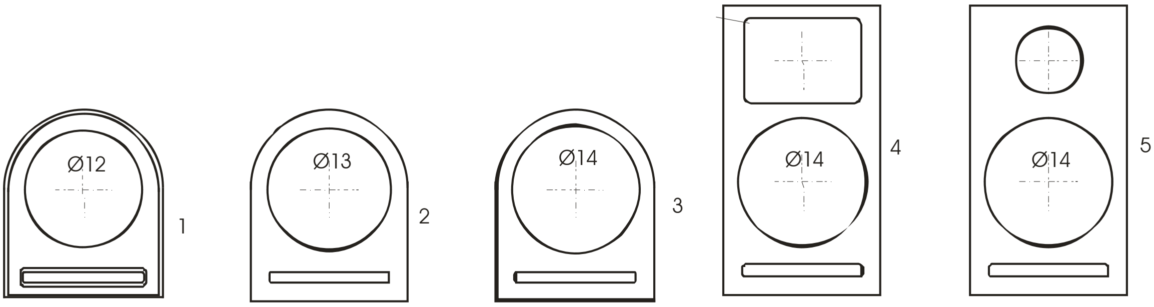
Theory: W6. Design



The tchnical drawings are developed from sketches.

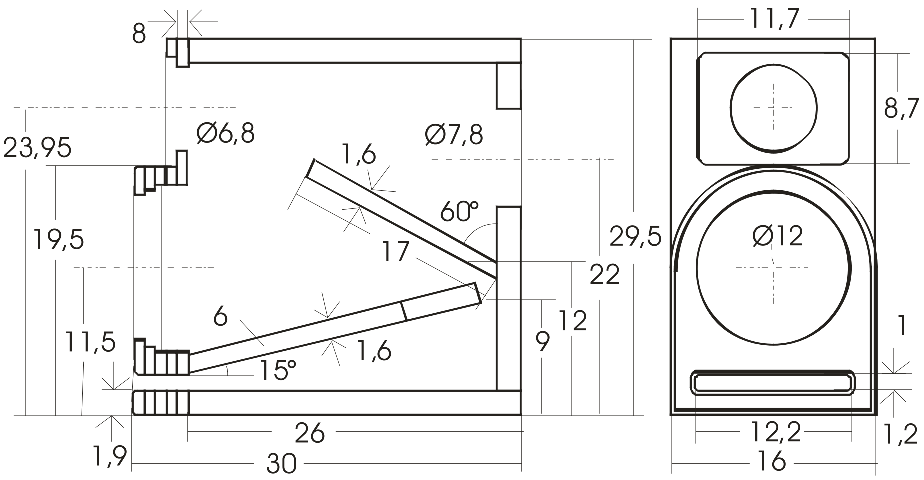
The step on the front is acoustically optimised by rounding it.

Curves on boxes have worked well on speakers in the past.



The front plate is built up from 8 mm sandwich board so it’s easy for home craftsmen and you don’t need a fraser. A carpenter would use 2 plates.

Building plan



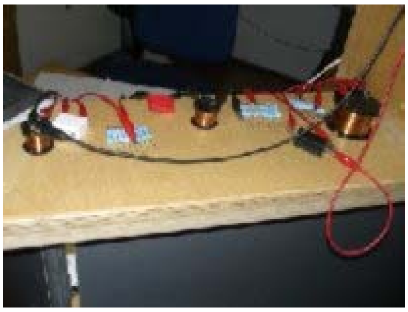
The material for the casing is 19 mm MDF, birch sandwich or beech multiplex. These materials are acoustic equivalent, when the casing is the same size.

6. The Crossover

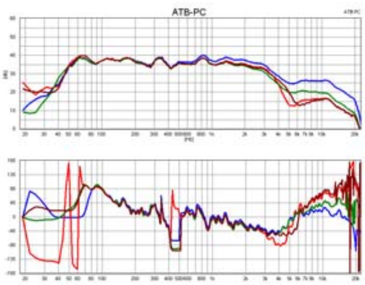
Theory: W7. The speaker crossover

6.1 Crossover development

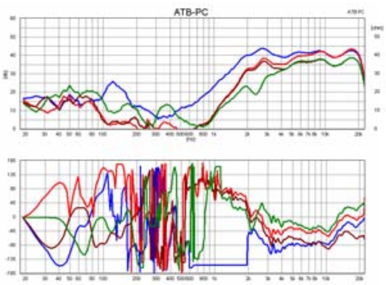
A preliminary crossover was developed for the phase correction. The crossover for the finished box has to be optimised.



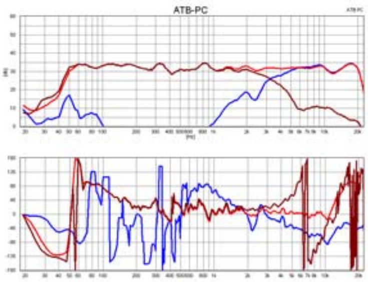
Developing the crossover
crocodile clips were used



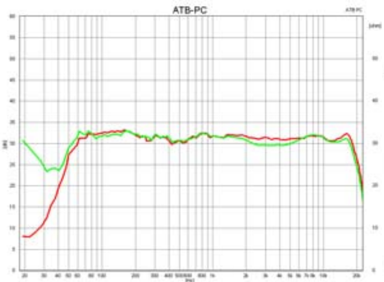
Woofer low pass
blue = without crossover
green = with inductivity
red = with drain 1.5uF
brown = with drain 1uF



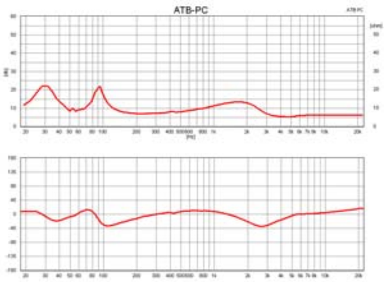
Tweeter high pass
blue = without crossover
red = with capacity
brown = with resister
green = with drain



Sound level curves
red = Addition
blue = Tweeter
braun = Woofer

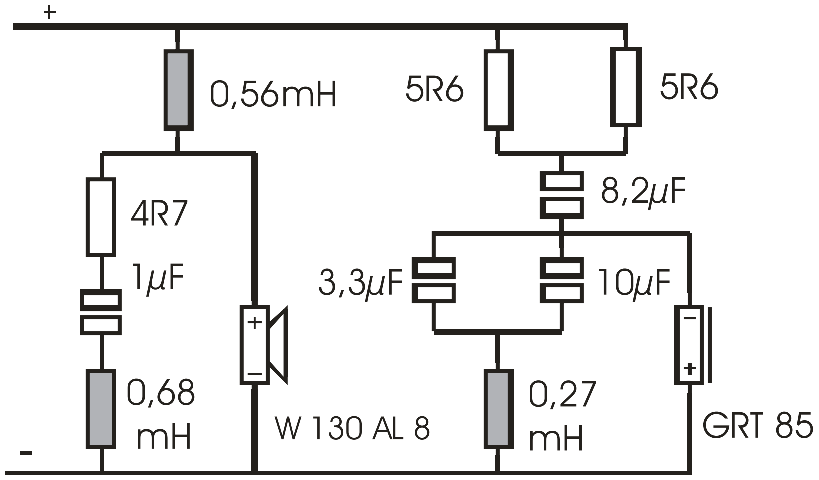


Radiation behaviour
constant measurement with average, area ± 30°
horizontal and vertical



Impedance curves
with a minimum of 5.4 ohm
and a phase angle smaller
than ± 45°, is also suited for
small amplifiers.

6.2 Crossover plan



Crossover parts:
Inductivity 0.56mH as air coil 1.4mm⌀, inductivity 0.68mH und 0.27mH as air coil 0.7mm⌀
Condenser8.2µF as MKP, Condenser1µF, 3.3µF and 10µF as MKT
Resister as 5W MOX
High End parts:
Coil 0.56mH as cfc-14Copper flat band coil 1.4mm³ Cut surface
Condenser 8.2µH as Mundorf MKP- Condenser400V

6.3 Platine layout und Mounting

Platine layout

Crossover

To be followed when developing:
The coils are to be placed as far apart as possible, the connection paths are out of well conducting surfaces, the speaker earthing are brought to one sole connection point.

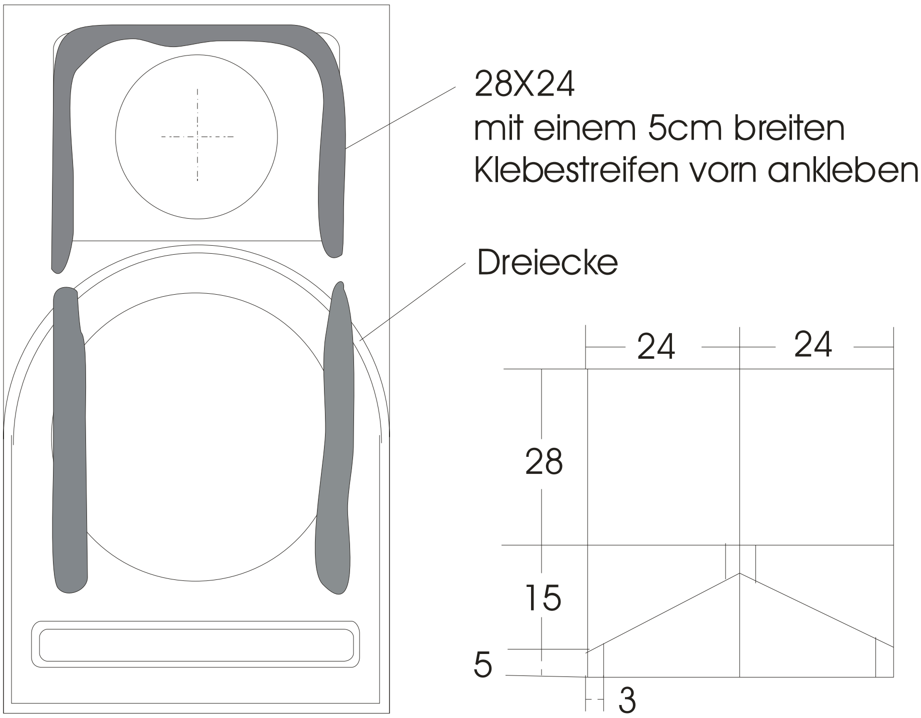
For the wiring of the speakers cables of 1,5mm² are used.
For High End connection cable Supra Kabel are used with the title Supra Classic Mini 1.6
With 0.01Ω/m the cross area is sufficient due to the short length.
As High-End connection cable we recommend Supra Rondo 4x2.5.
The four wires are crossed over to compensate the inductivity.

7. The Dampening material

The dampening material is only to absorb the mid frequencies standing waves as far as the Nugget is concerned. It is only mounted on the side walls and under the top side. The rest of the space is left free. Because of the slanted board construction this isn't needed. The advantage of dampening the walls is an Impulse true reproduction. If instead the space is filled with for instance polyester wool, the energy for the step impulse is absorbed as it swings with the sound waves, leaving no energy for the speaker.

The special dampening material is out of loose textile fazes and is self sticking.

Dampening plan



Figur 1

8. Mounting

8.1 Speaker preparation

TheWoofer is prepared with sealing tape for windows and doors 4x9mm.

The Tweeter cavities in the mounting plate can be filled with acryl sealing.This is acoustically even better than if the mounting plate were of metal. The tweeter is mounted with foam tape.

8.2 Enclosure preparation:

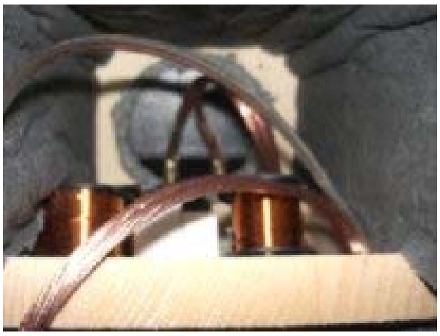
Following screws are needed:
Speakers: Pan head Z2-Bits black 4x20, 2,5mm predrill 16 pieces
Or High End: Imbus black wood screws 4x25, 3mm predrill
Connection socket: Chip wood screws sink head black 3,5x16, 2mm predrilled 4 pieces
The casing should be predrilled accordingly.

8.3 Mounting the dampening

The dampening material is cut according to the drawing in chapter 8. The triangles are stuck to the sides. The big piece is freed of a stripe of the protection foil along the long edge, 5cm wide. That piece is then glued in U-Form in the top inside, with the sticky side to the front.

8.4 Mounting the crossover

The cable on the crossover is marked as on the crossover.
The crossover is mounted on the slanted board in the middle, on top using acrylic to glue it on.



The position of the crossover in the box. The glued in crossover can be seen.

8.5 Mounting the speakers

First of all the Tweeter is soldered on. The wiring of the crossover is de-isolated and pre-soldered. The soldering points on the speaker are also to be presoldered.

> Attention:
> Watch poling! The marked wire is soldered to the red soldering point.

The presolded wire is put on the soldering point of the speaker and held in place with the soldering iron tip, thus heating up both at the same time. After the solder has melted remove the soldering iron and keep the wire held on until the solder has become solid.
After that the Tweeter is screwed on.
The Woofer is done in the same way as the Tweeter.
The marked wire also is soldered to the red soldering point.
The connection socket is soldered on observing plus and minus. Once again the marked wire is plus or red. First screw in the connection socket only, it gets glued on according to chapter *6.5 low resonance casing,* after you have tested the box.

8.6 Test measurement

The function of the box is best tested with a frequency sweep.
If the point of crossover from mid to tweeter is correct the crossover is made properly and the speakers have been connected with the proper polarity.
The most important criterion with the measurement is if the boxes have an almost identical frequency sweep curve. The frequency sweep curves will be slightly different due to the room distorting the sound when just a simple measurement is taken.

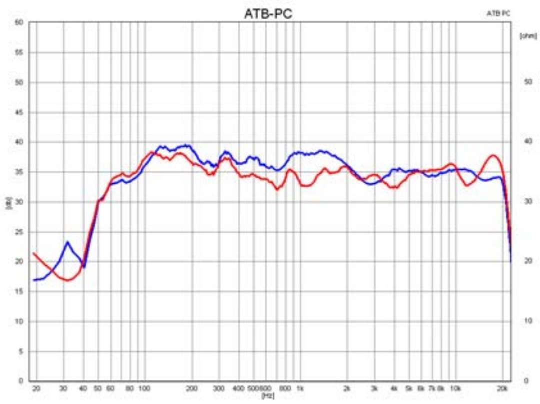
9. The sound test

A speaker can not be developed just by physical measurements.
To tune the timbre of the Nugget it had to pass through several processes. The shown crossover is the result of the sound test!
Of great help were the tests at the shop with customers from all areas of the acoustic such as simple music lovers, musicians, sound masters, High-End customers, technicians from the car HiFi area and speaker builders. For the tests over a period of three months the crossover was mounted outside the box so it could quickly be adapted.

9.1 Comparison with own developments

9.2 Comparison with reference speakers

As a reference an English High-End box was chosen.



blue = Reference box, red = Nugget

The reference speaker emphasizes the mid tones, to get more in direction of the certified tuning the mid tone of the Nugget was chosen a bit louder.

9.3 Concerning the test Amplifier

This development was completed using the amplifier from Sansui. The source was a modified TEAC DVD player.

For the further tuning a surround amplifier and TEAC CD receiver with 2x20 watts were used. The digital amplifiers showed the full potential with use of the Nugget.

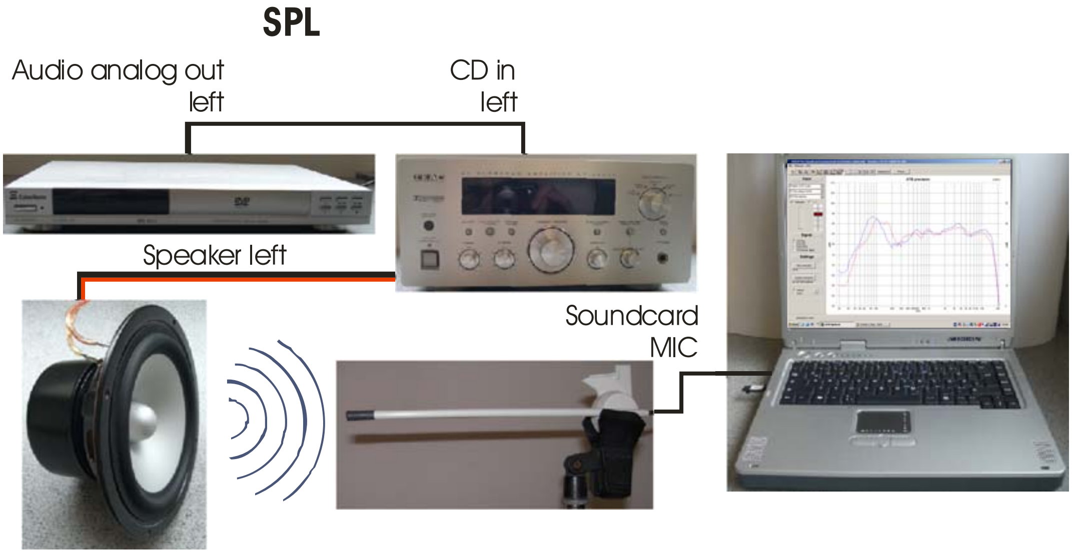
9.4. In the living room

The Nugget together with LCD TV in the living room.

Theory:

W1. Ground theory of acoustic measurement

The Sound-Pressure-Level Measurement shows the speakers frequency response.



W1.1 Measurements in rooms

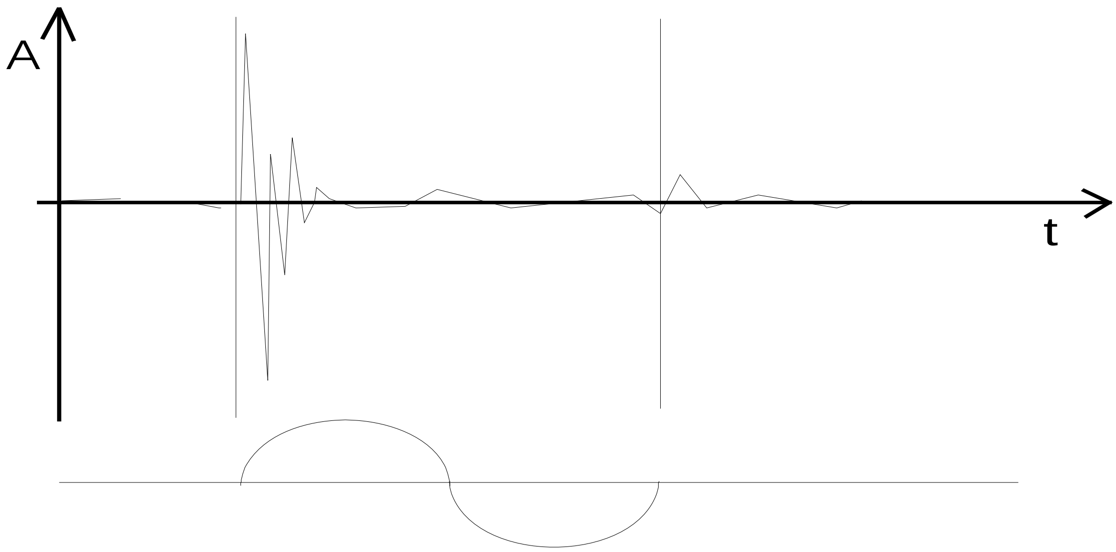
The measurement of sound level of a speaker across the frequency SPL (Sound Pressure Level), as also the acoustic phase and impulse response, are acoustic measurements. The acoustic measurements are influenced by the acoustic qualities of the environment in which these measurements are made. Not so electrical measurements. In the following we will show how such influences of the measurement environment can be reduced.
Measurements with a level scriber cannot separate such influences. A level scriber that uses a sinus sweep for the measurement needs a sound proof room for its measurements. The walls of such a room are damped in such a manner that there are hardly any reflexions. In the low frequencies' where the dimension of the room are in the same area as the wavelength of the sweep, standing waves occur that can not be suppressed. The scribing speed of the scriber can be used to smooth out a bit. But also the rest of the frequency sweep can be smoothed out by the speed; and more than few developers may not be aware of want to admit the fact.

W1.2 Time window

As not every developer has a sound proof room to reduce the effect of reflexions on his/her measurements, when measuring speakers, the FFT measurement was developed. By this kind of measurement a constant noise signal is used, in which all frequencies' have the same sound or signal level. This can be an MLS signal or the Signal from an ATB PC measurement system. When measuring, time windows are setup so that only the directly radiated sound waves sent from the speaker are taken into account, for the measurement results. The delayed sound waves of the reflections that reach the microphone are outside the time window and do not show in the measurement results.

This on first view „ideal"measurement method makes it's demands on the time window. If these demands are not reached, it will lead to false results. The demands on the time window will show the following:

To set the time window a measurement is needed. This is an impulse measurement that can be carried through with a MLS signal. The measurement result is the impulse response. By the impulse measurement the mic level is shown across the time axis.



This figure shows the Impulse Response of a speaker. The Y axis is the amplitude and the X axis is the time.

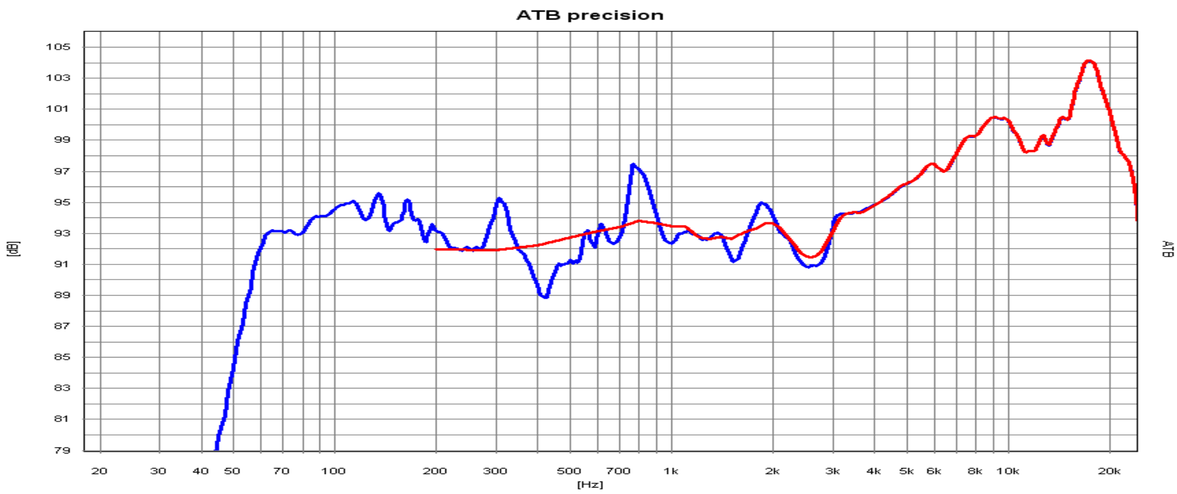
From t = 0 until the high amplitude you see the running time of the signal. The high amplitude gets less and less until the next maximum. This is the delay time of the reflection in the room. To blend out the reflexion in the following calculation of the frequency curve, the time window is set from the start of the signal to the reflection. From the measurements within the time window the frequency sweep is calculated with help of the FFT method.

Under the window a fig. of a sinus period is shown.

For the evaluation of the impulse with FFT there are following rules:

1. The wavelength of the smallest to evaluate frequency can only be as long as the time window. The shorter the time, the higher the lowest frequency. In expensive measurement programmes this frequency is shown in the frequency sweep.
2. The lowest to evaluate frequency has further consequences. By FFT it determines the spacing of the FFT measurement points. The FFT points are the frequencies that the FFT method uses to calculate the frequency sweep. The frequency spacing is constant and has the value of the lowest frequency. The magnitude values of the FFT points create the frequency response curve. This means that if the lowest frequency is 300 Hz, there will only be one measurement point every 300 Hz. That would make the response curve nice and smooth, but leads to totally incorrect measurement results.
3. In cars the time window would be so short because of the immediately occurring reflections, due to the small room, that no time window can be used.

Comparison of the ATB-PC measurement, blue, with the usual measurement with time window, red.



The chosen short time window shows a smooth curve that supposedly occurs through the blending out of the reflection that occurs after a short period. The truth is that such a nice frequency sweep is achieved by an incorrect frequency resolution. This kind of measurement inaccuracy is avoided by the ATB-PC by use of a different measurement method.

W1.3 The ATB PC measurement for mid and high frequency

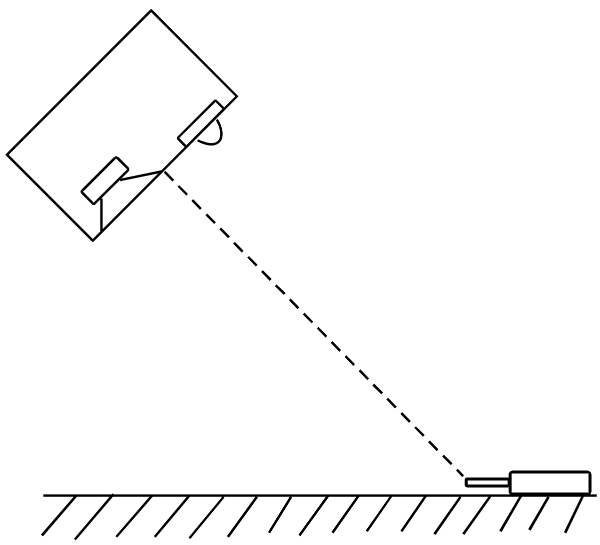
For the mid and high frequencies a constant signal and averages are used. By this measurement the microphone is slowly turned in a circle with a radius of 10 cm. Here the room modes for higher frequencies’ are measured and by averaging, the effect on the measurement is eliminated.

W1.4 The near field measurement

For low frequencies in the bass region a near field measurement is used. The foundation of the near field measurement is the physical fact that the sound pressure decreases to the power of 2 with the distance. Which means that the direct sound pressure of the speaker is a great deal louder that any reflexion occurring from the room in comparison. If the measurement microphone is placed at maybe 10 cm the direct sound pressure will be so much higher than the reflected sound, that basically only the direct sound of the speaker is measured. This measurement is carried out below 300 Hz and complemented with measurements above 300 Hz. The complimentary measurement is carried out with 1 meter distance for mid range and tweeter. The disadvantage of the near field measurement is the accumulation effect, causing an increase of sound pressure in the near field area. But this is compensated in the combine menu of the ATB-PC. If there are more than one sound source, such as speaker and reflex tube, both can be acquired with a constant measurement and then averaged. Doing this the microphone is moved slowly between the two sources during measurement.
The Deerfield measurement is also to be used in sound proof rooms. As already described even a sound proof room has lower frequency reflection that will falsify measurements in that area. Besides that the measurement in a sound proof room doesn’t relate, as when listening the speaker usually stands on the floor and the room in which you listen the woofer “sees” a different environment. The floor causes a 3dB rise for low frequencies, whereas that is also dependant on the construction of the speaker.

W1.5 The surrounding surfaces measurement

By the surrounding surfaces method the microphone is used as a surrounding surfaces microphone. The microphone lies flat on the floor and has then half ball characteristic.



This Fig. shows the position of microphone and speaker.

The sound waves of the speaker hit the floor at an angle and reflect into the room. This has a positive effect on the mid high tone range. But also in the low frequency range where the room resonances develop, the energy is only acquired to the half from the surrounding surface microphone, so that they do not effect the measurement so much. All the same the same rules apply here, the bigger the room the better.

W2. Choice of speakers

By the choice of speakers the main objective was the most up-to-date technology to moderate prices.
To be able to make first assessments you would need a frequency sweep and all necessary parameter from the manufacturer.
Good market knowledge and observance of new developments before starting can also be of advantage.
So for instance, after looking thorough specialist literature the tweeter GRT 85 caught eye, an air-motion tweeter with a two figure price. In the past this kind of tweeter could only be found in the three figure price class. As these tweeters are quite special because of their construction and historical value they will be dealt with in a separate capital (air- motion tweeters).
The choice of the woofer was predetermined to part by the size of the planned enclosure. The enclosure volume was to be less than 10 liters, so that the dimensions fit well into living room landscapes and allow use in surround setups. The aim was to create a small speaker that despite the small volume still has good low bass qualities. How that can be achieved is discussed in capital 6 The Enclosure.
The manufacturer of the tweeter also had a woofer with 13cm diameter with an enclosure suggestion of 8 liters in their program. The woofer is discussed closer in the capital 5.5.8 Woofer.
But in the end only after developing the enclosure crossover (chapter 7 The Crossover) and after the hearing test can be told whether or not the speakers match or not.
But before we describe the speakers in detail, first some theory.

W2.1 The ideal sound source

The best case would be a box with just one broadband speaker that transmits the whole frequency range of human hearing. Some manufacturers manage to build cone speakers that achieve this with relatively even amplitude across the complete listening range. Unfortunately the physics foil the calculation. The even amplitude is only there at 0° on axis. If you move to side and stand at 30° to the axis of the speaker you will notice the following:
Up until a certain frequency the amplitude stays level and then starts to sink with rising frequency. The round radiation of the speaker decreases at that frequency as the sound waves are bundled more and more with rising frequency.

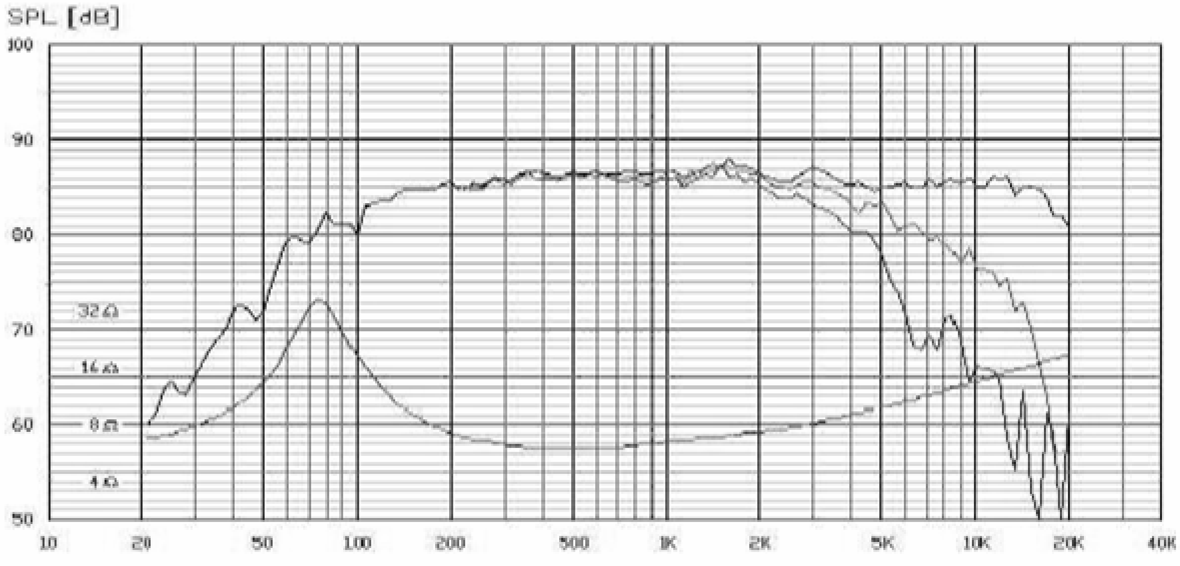


Figure 2.1

This is revealed with help of the frequency sweep in figure 2.1. The almost whole range flat curve was measured at 0°, the mid curve at 30° and the lower curve at 60°. The curve at the bottom is the impedance measurement which we will not go into at this point.

The frequency (fmax) to which the bundling is still acceptable can be calculated. The speed of sound is divided by the effective membrane diameter.

$$f_{\max} = \frac{c}{D_d}$$

If you take the manufacturer given cone area of 38 cm² and calculate the diameter for that, which is 7cm, you will come to following maximum frequency.

$$f_{\max} = \frac{34300 \frac{cm}{s}}{7cm}$$

$$\underline{f_{\max} = 4900Hz}$$

Above this frequency (Fig. 5.1) the amplitude at a 30° angle has decreased so much that at 10Khz the amplitude has gone down by half.

W2.2 The two way concept

To get a good circular radiation behaviour over the whole frequency range you have to split the frequency range to several single speakers.
With use of a crossover, consisting of high pass and low pass the splitting of the frequency range is accomplished. More on that theme can be read in capital 15 „Crossovers“.
The danger with two or more single speaker boxes is that the crossover from one speaker to the other may be heard. The speaker reproduction loses strongly on three dimensional character and detail-ability.

Why does this happen?
The human ear is well able to locate singular tones very precisely. Because of this ability humans register also the tiniest time delays. So it's possible for us over a range from 500 to 3000 Hz to hear a sound time delay from about three hundred thousands of a second.
This acoustic time delay is also called the acoustic phase. A visualised explanation to this can be found in chapter 17 "Running time, Acoustical phase". By the development of a lot of speaker boxes no attention is paid to the acoustic phase. Developing this speaker the acoustic phase was one of the main criteria, creating a box where you can't hear the single speakers and letting them sound as one.
This was achieved using practical knowledge and with use of the measurement software ATB PC Pro from the Kirchner Electronic company.

W2.3 The Air Motion Transformer

Figure 2.2

Figure 2.3

The Air Motion Transformer was invented by the German physicist Dr. Oskar Heil at the eve of the 70's. Dr. Heil (Figure 2.2) became well known by inventing the field effect transistor (FET) and registering the knowledge base for patenting in the year of 1934. Before and during the Second World War he worked on the development of microwave tubes. Later a microwave generator was named after him. After the war he immigrated to the USA. In 1973 he founded the company ESS. Here he developed the first Air Motion Transformer (Figure 2.3), which was brought on to the market in 1973 and is still produced today. Before he began developing the Air Motion Transformer, he studied the function of the human ear and origin of sound in nature.

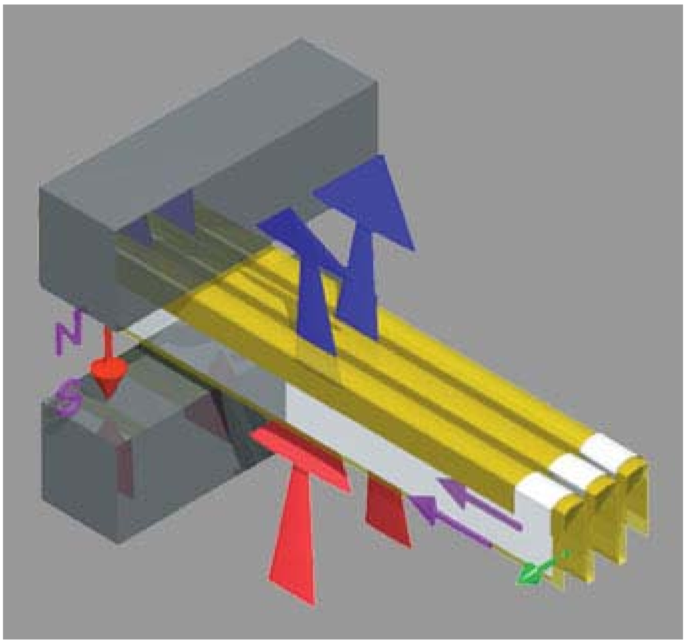


Figure 2.4

Finally he found the solution for his intentions through observing flying insects. They have light weight wings and a particular technique of opening and closing them. Using their wings to fly they intentionally or not also created sound with them. Dr Heil´s patent describes a folded foil similar to an accordion, with conduction paths on it. If the foil is put into a magnetic field and a musical signal is sent through it, the foil folds will open and close (Figure 2.4) in rhythm with the music.

Figure 2.5

The mass Mms of an Air Motion tweeter (Figure 2.5) is less than 1 gram. Due to folding these tweeters have, according to manufacturer and model, a 4 to 20 times larger membrane surface as normal tweeters, without losses in three dimensional character behaviour. Earlier Air motion tweeters were very large and heavy as strong magnetic fields and because of that large magnets were needed. Nowadays things are easier by simply using Neodymium magnets.

W2.4 The Low-Mid speaker

Figure 2.6

The biggest challenge for the Low-mid speaker (Figure 2.6) is the reproduction of very low frequencies in small enclosures and at the same time a high resolution mid range reproduction. Today this is achieved by the use of new membrane materials, the knowledge of the manufacturers and the proper application in speaker enclosures.

The Membrane material.
Coming from theoretical ideal wishes, the ideal membrane has a very low weight and is made of almost unbend able material. These demands are achieved with this speaker by using a thin aluminium metal sheet with a special geometry.

The membrane geometry.
By the membrane geometry of woofers nowadays we differentiale between the cones form and special curve forms.
Whereas the cones form is just a straight line from coil to suspension and through that tends to vibrate just like thin metal sheeting, the special curve form is a bow that changes is radius along is length. Because of this geometry the membrane reaches high stability, similar to the stability of eggs.

The double magnet screening
Because of the compensations magnet and the screening through the steel basket a position next to TVs is possible without ruining the picture.

The steel basket
The steel basket builds the link between motor, membrane and suspension of the speaker. The steel basket should be highly stable and because of that low resonant.

The woofer parameter

Mechanical parameter:
Outer diameter: 147 mm
Mounting hole: 128 mm
Mounting holes circle: 137 mm
Mounting depth: 77 mm

Thiele-Small-Parameter:
Free to air frequency: f_s = 44 Hz
Effective membrane area: S_d = 85 cm²
Dynamical masse: M_{ms} = 9.8 g
Equivalent volume: V_{as} =13.4 l
Mechanical grade: Q_{ms} = 1.75
Electrical grade: Q_{es} = 0.42
Overall grade: Q_{ts} = 0.34
Sensitivity: 88 dB (1W;1m)

Sound coil geometry
Diameter: 26mm
Height: 10mm
Linear movement length: ±2mm

Electrical Parameter
DC Resistance: R_{DC} = 6.14 Ω
Impedance: Z = 8 Ohm
Nominal power maximum: P_n = 70 W

Outer geometry
The first parameters describe the size and mounting dimensions of the speaker.

Outer diameter of speaker.
This is just the outside dimensions of the speaker basket and doesn't say anything about the actual size of the membrane.

Mounting whole diameter.
This is a measurement that has to be observed as closely as possible, as it affects the mounting surface of the mounted speaker and therefore the forces that work between speaker and enclosure.

Mounting holes circle.
By proper marking of the mounting holes you can get the speaker exactly into the wished position. It is of course important to draw the circle for the mounting holes before you cut out the mounting hole itself as you then wouldn't have a surface for the mid point of the circular. When mounting the speaker you have to be sure to tighten the screws diagonally to avoid tensions on the basket.

Mounting depth.
The mounting depth describes the smallest possible distance between mounting surface and inside back wall of the enclosure, when the speaker is build in the front.

W3. Thiele-Small-Parameter

After the mechanical parameter we now come to the most important parameters, the so called Thiele-Small parameter. First a little history. The first bass reflex loudspeaker was already patented in 1930. The patent for closed boxes, based on an air cushion, followed in 1944.
For a long time bass reflex boxes had an air of mystic about them as there was no mathematical knowledge for calculating them. To test the efficiency of the reflex opening burning candles were put in front of the opening for instance.
First through the scientific work of the Australian Neville Thiele of 1961 and the later compliments of the American Richard Small the fundamentals for calculating loudspeakers were laid. Both went through long and tedious test rows and revealed mechanical relationships.
Today the Thiele Small parameter are recognised by industry and make speakers comparable.

W3.1 The free to air frequency f_s

Fs is not only the frequency of the moved mass and parts of the suspension, but also that of parts that normally are fixed after mounting such as the basket that is sent into vibration. As the stiffness of the centering and the surrounding changes after use, the frequency should be measured after a certain "running in" of the speaker.
For this measurement the speaker is completely mechanically decoupled from its environment. The manufacturers go to great lengths at this.
This value can be determined almost exactly by laying the speaker on loudspeaker dampening watt.

W3.2 The effective membrane area S_d

To calculate this value, first the effective membrane diameter D_d in cm is determined. It consists of the membrane diameter and half of the surrounding. Or in most cases from the one highest point to the other of the surrounding. But obviously not by inverse surroundings.
The half of this value is multiplied to the power of 2 and then multiplied with π.

$$S_d = \left(\frac{D_d}{2}\right)^2 \times \pi$$

W3.3 The dynamic mass M_{ms}

The dynamic mass or moving mass is the sum of all mass that's put into movement when the loudspeaker is working. This includes the mass of the membrane, dust cap, coil and coil carrier as well as the moving parts of the centring, surrounding and air moved by all these.

W3.4 The equivalent volume V_{as}

The equivalent volume is the volume of air that when it's compressed to $1m^3$ has the same energy as that of the speaker suspension.
With this from the industry suggested value it's possible to do some calculations. For instance the resonance frequency of the mounted speaker can be calculated with the following formula: Apart from f_s and V_{as} only the volume V_b of the planned box has to be put into the formula.

$$f_c = f_s \sqrt{\frac{V_{as}}{V_b} + 1}$$

Also other important parameters such as the speakers Q_{tc} can be calculated with help of V_{as}.

W3.5 The suspension compliance C_{ms}

The mechanical compliance is a measure to how much the suspension gives way, and is measured in Newton per mm. The value of C_{ms} can be obtained by measurement.
The manufacturers do not always state is value, but it can be calculated with help of the values for the dynamic mass (M_{ms}) and the free to air resonance frequency (f_s).

$$C_{ms} = \frac{1}{(2\pi \times f_s)^2 \times M_{ms}}$$

W3.6 The mechanical grade Q_{ms}

Q_{ms} describes the grade of control over the moving parts of the speaker during resonance. The control over the moving parts is given through the centring and surround of the membrane. Q_{ms} can be calculated as follows:

$$Q_{ms} = \frac{2\pi \times f_s \times M_{ms}}{R_{ms}}$$

W3.7 The electrical grade Q_{es}

Electrically the moving parts of the speaker are controlled at resonance by the two-way influence of the electrical field in the voice coil and the magnetic field in the air gap.

$$Q_{es} = \frac{2\pi \times f_s \times M_{ms} \times R_{dc}}{BI^2}$$

By which R_{dc} is the direct current resistance of the voice coil, B the magnetic flow density and I the effective width of the magnetic field.

W3.8 The overall grade Q_{ts}

The value of Q_{ts} is calculated from Qms and Q_{es} as follows:

$$Q_{ts} = \frac{Q_{ms} \times Q_{es}}{Q_{ms} + Q_{es}}$$

With the values of Q_{ts}, V_{as} and $V_{b\ lt}$ it is possible to calculate the overall grade of the speaker Q_{tc} in an enclosure, using the following formula.

$$Q_{tc} = Q_{ts}\sqrt{\frac{V_{as}}{V_b} + 1}$$

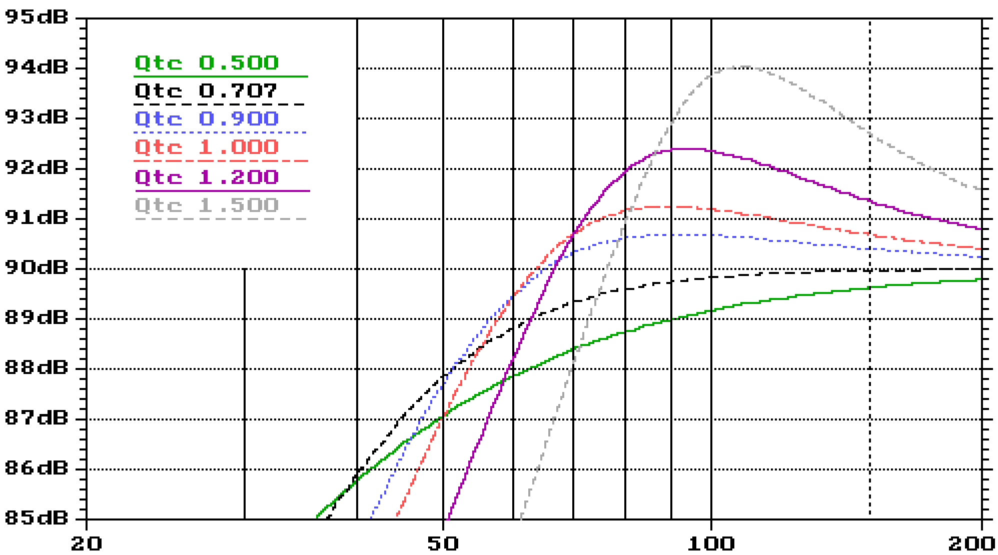


Figure3.1

The value of Q_{tc} (Figure 3.1) is a real telltale value as to the amplitude behaviour around the resonance frequency range of the speaker. It shows the behaviour of the speaker in an enclosure. Vice versa if you turn the formula around you can calculate the needed volume for a certain low bass behaviour.

$$V_b = \frac{V_{as}}{\left(\frac{Q_{tc}}{Q_{ts}}\right)^2 - 1}$$

With for instance a chosen value for Q_{tc} of 0.707 you calculate the volume V_b for a closed box as follows:

$$V_b = \frac{13{,}4l}{\left(\frac{0{,}707}{0{,}34}\right)^2 - 1}$$

$$V_b = \frac{13{,}4l}{\left(\frac{0{,}707}{0{,}34}\right)^2 - 1}$$

$$\underline{V_b = 4{,}0l}$$

With the calculated volume V_b you can then calculate the resulting closed box resonance frequency F_c of the build in speaker.

$$f_c = f_s \sqrt{\frac{V_{as}}{V_b} + 1}$$

$$f_c = 44Hz \sqrt{\frac{13{,}4l}{4{,}0l} + 1}$$

$$\underline{f_c = 92Hz}$$

We will be going into the direct current resistance R_{dc} and the inductance in chapter *W8. Crossovers*.

The nominal power maximum P_n will be disregarded in this development as it plays a subordinate role. The most speakers are nowadays have a power handling that makes it just about necessary to destroy them on purpose.

W3.9 Enclosure variations

Why an enclosure ?
To avoid that the air move through the membrane motion and the over and under pressure caused by that doesn't be compensated immediately (Figure 3.2 acoustical short circuits), the front and the back side of the membrane have to be separated by a wall. Ideally an infinite wall. As this is not practicable the wall is folded to a box which closes the back of the speaker. The air can't compensate through the closed box as the path also is infinite because of the closed box.

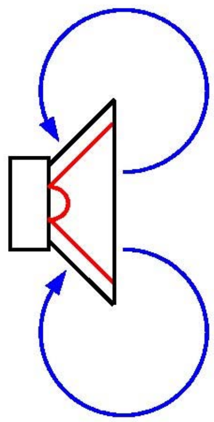


Figure 3.2

Further on the blue arrows always show the power or movement directions and the membrane is always shown in red.

Enclosure forms
On the one hand there are enclosures of different shapes, for instance round, four cornered, three cornered etc. They will be differentiated according to shape.
On the other hand there are different enclosure types such as closed enclosures, bass reflex, transmisssionline and horn.
At this point we will only go into the most usual types which are bass reflex and closed boxes.

Closed boxes

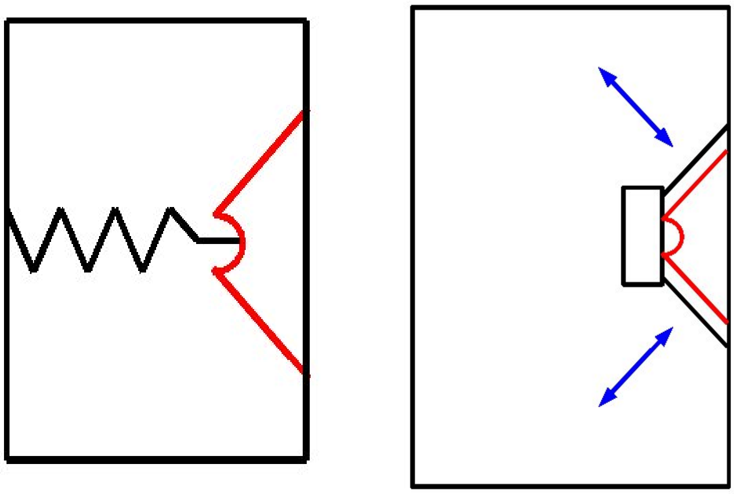


Figure 3.3

The captured air of a closed box works upon the membrane like a spring (Figure 3.3). The larger the volume of the box the softer the spring influence of the enclosure. Together with the mass of the membrane a resonance frequency is built that needs only little energy to create a sound.

Bassreflex

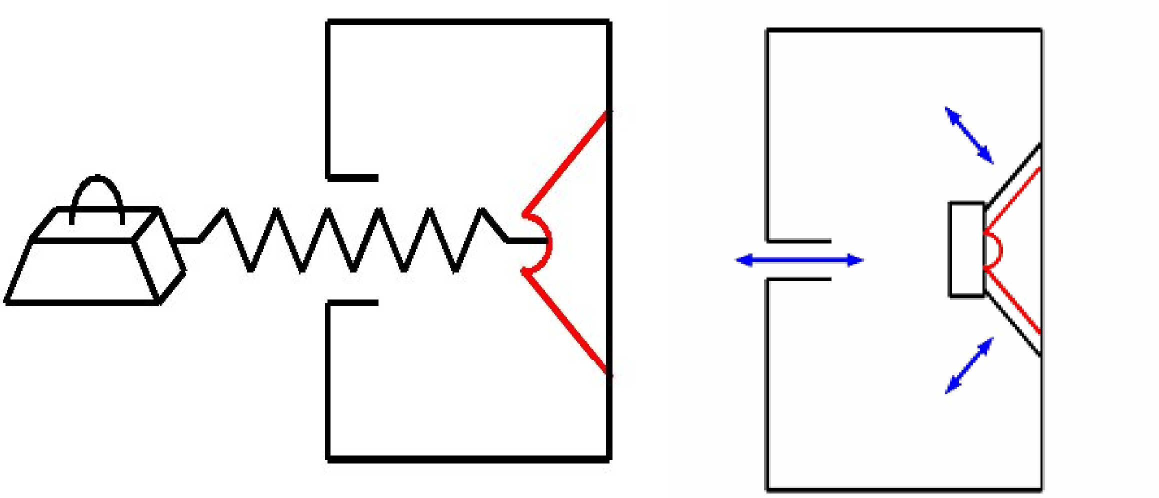


Figure 3.4

In a bassreflexbox there is in addition to the box volume that acts as a spring a second volume that acts as a spring as well and with that an extension of said spring (Figure 3.4). That volume is that of the bass reflex tube. It reacts with the air outside the box as a counterweight to the speaker membrane. This air mass-spring-membrane mass combination has two resonance frequencies', as it has two different weights an each end of the spring.

W3.10 Calculating Enclosure size

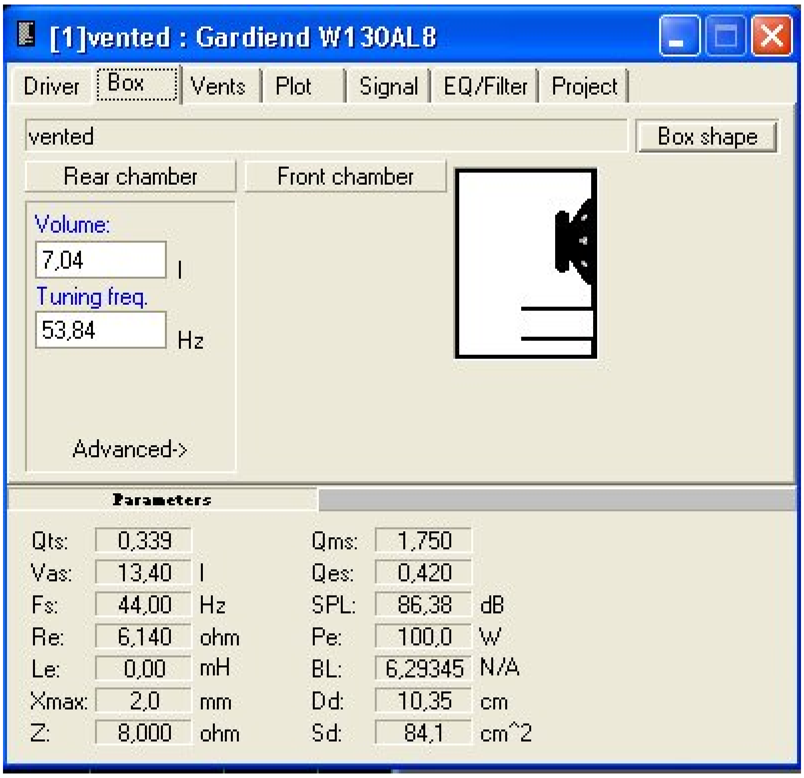


Figure 3.5

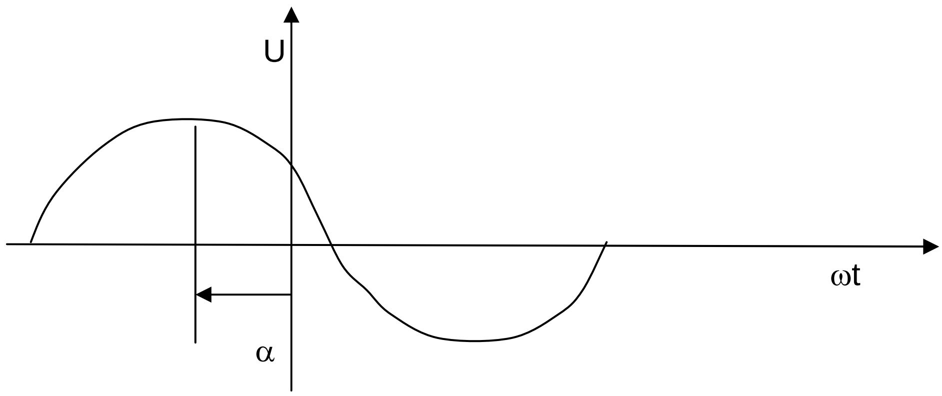
To calculate the enclosure size we used a simulation program, in this case „Win ISD pro alpha" (Figure 3.5). At this point we would like to note that there is a great variety of speaker simulation programmes.

Enclosure size
This lead to an enclosure size of 7.04 Liters with a reflex tuning of 53.8 Hz. The width of the box was given by the diameter of the woofer; the height should leave enough room for the reflex opening and the tweeter, so that the depth resulted in the needed volume. On the basis of that a first testing enclosure was set to: height 30cm, depth 28cm, width 16cm.

W4. The Phase

W4.1 Basics

The phase is used when describing sinus shaped processes such as mechanical oscillation, alternate currents, radio waves and sound waves.



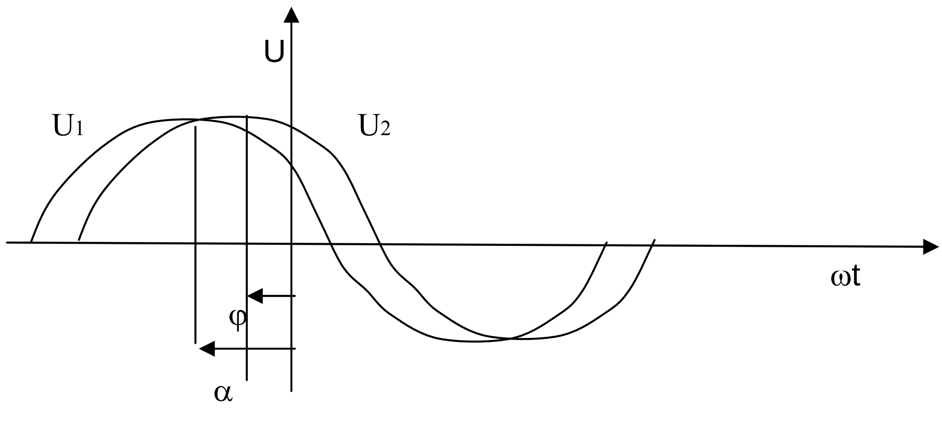
$U(t) = Us \cos(\omega t+\alpha)$

t Time, U(t) Instantaneous value of voltage, Us peak value (max. Amplitude),

f Frequency, T = 1/f Period duration, $\omega = 2\pi f = 2\pi/T$ Angular frequency,

α Null phase angle

The phase angle α already appears in the in the basic equation. It is purposefully positioned with the point of reference TIME 0. With two or more oscillations (waves), as with e.g. acoustic signals, the phase angle is of absolute importance for measurement description.

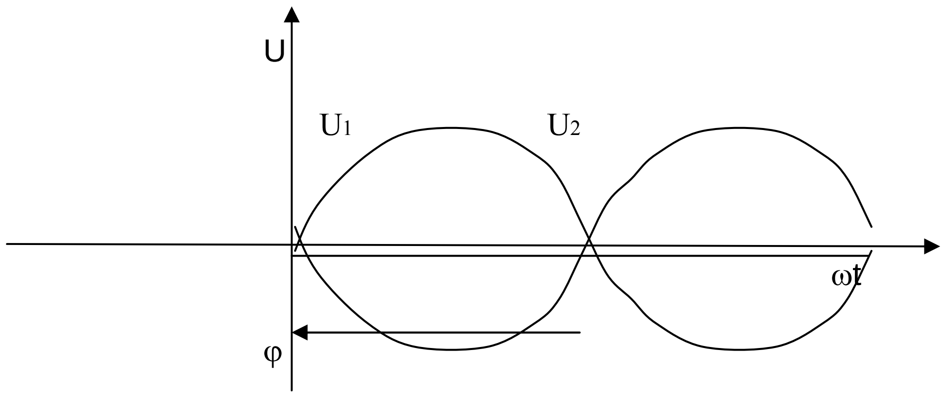


$U(t) = Us_1 \cos(\omega t+\alpha) + Us_2 \cos(\omega t+\alpha+\varphi)$

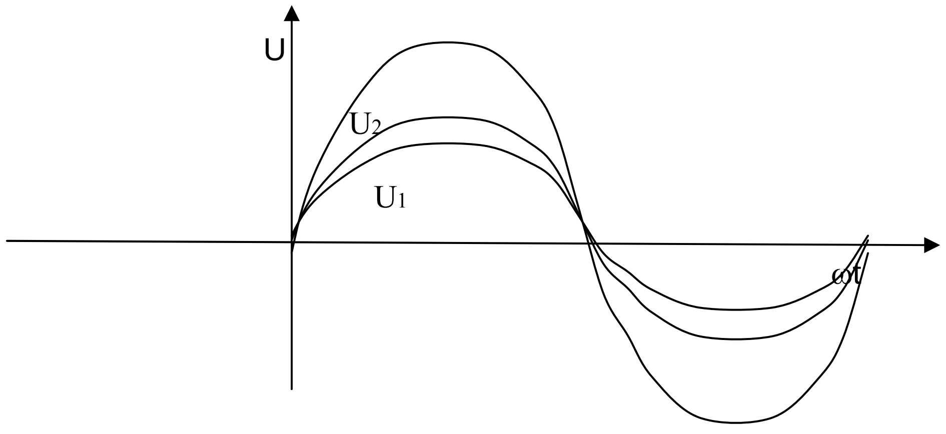
The angle φ defines how the oscillations (waves) overlap.

Examples of how the angle φ affects the waves :

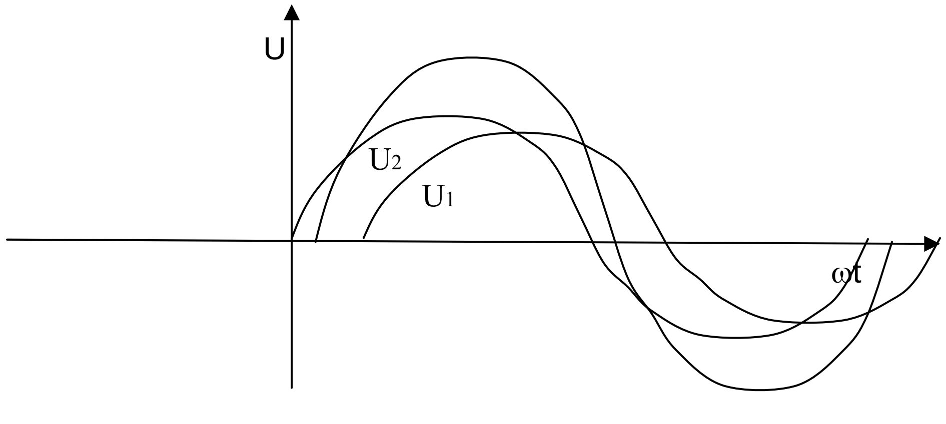
φ = 180° => blanked out



φ = 0°, 360° => maximum amplitude



φ = 90°, 270° => partly amplified, or also partly blanked out



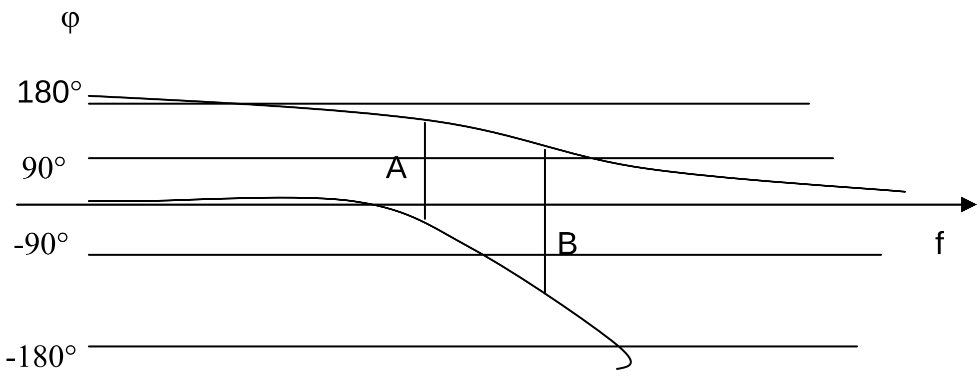
W4.2 The Phase and Loudspeakers

Loudspeakers also have to comply with the basics of physics. Amplitude and phase are essential for a correct description of the sound pressure emitted. The common theory that the phase can be determined in the frequency response is <u>not</u> true. This only complies with simple electronic circuits. The complicated transfer function, including duration, inhibits phase recognition in the amplitude frequency response. A box with an absolutely smooth frequency response can produce extreme and therefore audible phase jumps.

A common method for detecting the phase relation in loudspeakers is as follows :

To determine the phase relation from the frequency response in the transition between e.g. the high and middle tones of two single loudspeakers, the crossover network is constructed to produce the maximum blanking out (extinction). When, after switching the poles of one of the loudspeakers, the frequency response is well balanced then the phase position is also correct. The phase values are usually about 180°, 360° or 540° apart and the transient behaviour is bad.

This method is not successful beyond the break frequency.



The figure shows the phase relation of the low and high pass

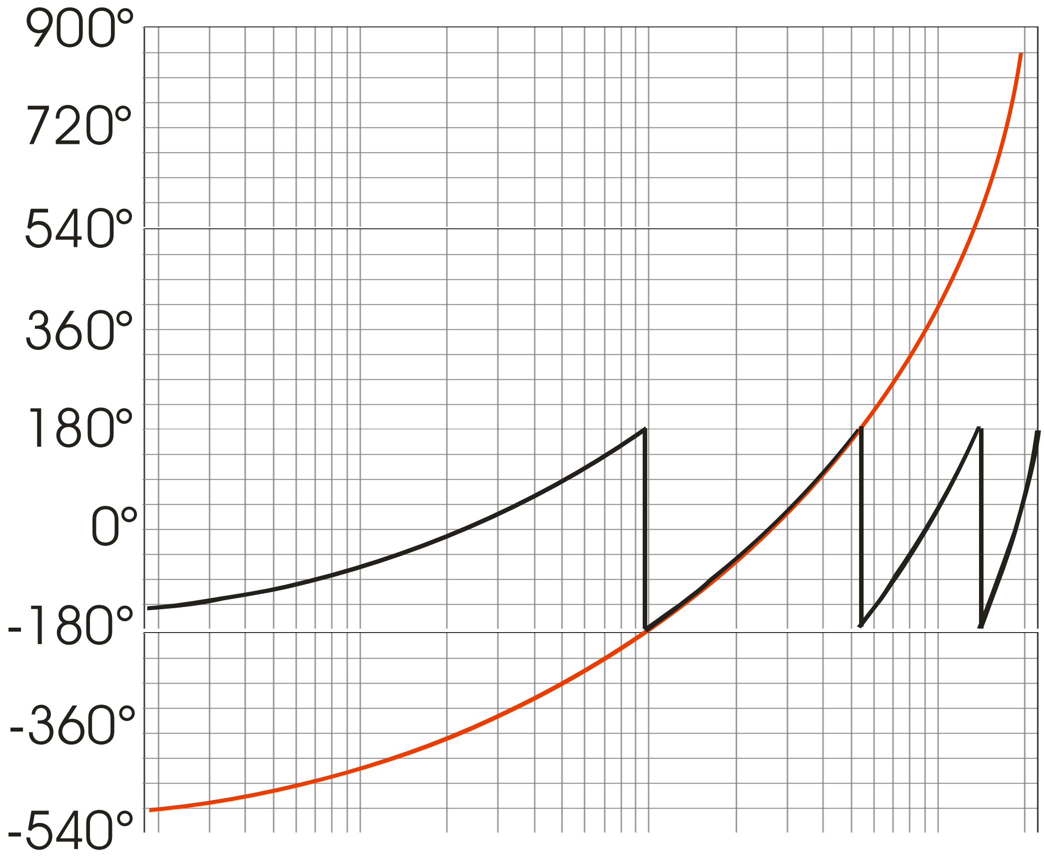
A = the determined phase of 180° using the method described above

B = phase angle of 270° . This angle partially erases the sound elements of the middle and high tones. A switching of poles on the loudspeaker shows the same frequency response, thus the fault cannot be detected with the usual method. With a phase angle of 90°, 270° the loudspeakers are too loud and produce an annoying sound quality. In this case the usual frequency response measurement has no meaning.

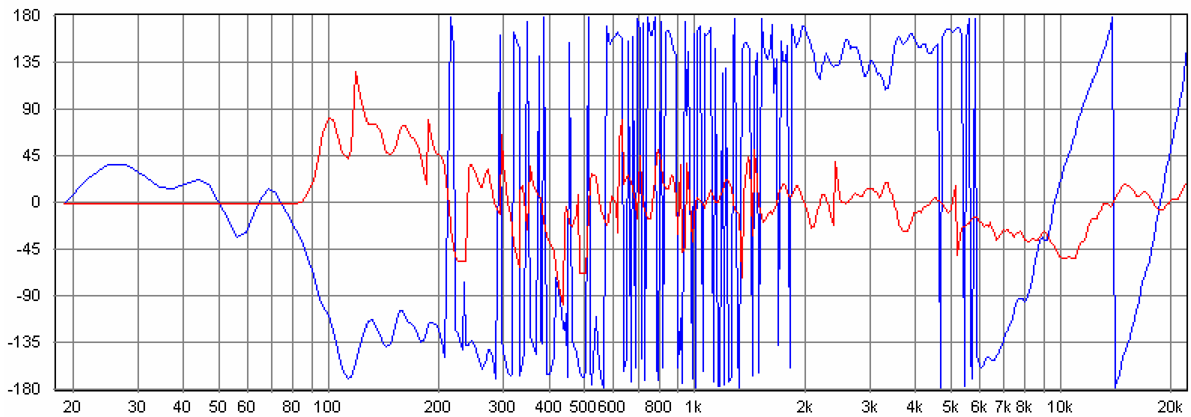
Many tests show that a balanced phase frequency response is absolutely necessary for a natural sound reproduction.

W4.3 Phase illustration

The graphic illustration of the phase using a plot is shown in an angle range of -180º to 180º. This range is insufficient for the turn of phase. A larger angle range worsens the illustration because of the absence of resolution of the y-axis.



The picture shows the real phase (red curve) and illustration (black curve).
The steps in the illustrated curve are not phase steps but are due to illustration methods. This leads especially with acoustical measurements to flash interpretations. When the phase angle is in the range of180º or -180º , the curve then springs between the two values. The blue curve shows a mis poled speaker.



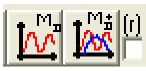
W4.4 Electrical Phase Measurement

With the electrical phase measurement we can measure for example loudspeakers impedance. This measurement is essential for a fault free operation of loudspeakers. Large phase angles show a capacitive and inductive behaviour of the loudspeaker impedance. The result is an overloading of the amplifier. The amplifier can also become instable and cause oscillation. After a system correction has been made available the loudspeaker can be connected to the clamps of the Test box. Using the button IMPEDANCE we can switch from frequency response measuring function to the impedance measuring function and activate phase measurement function. In phase the option ELECTRIC is selected.

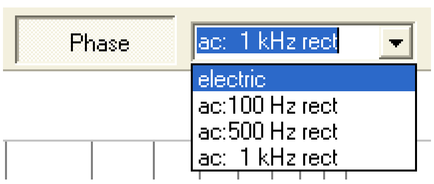
With the phase measurement the quality of soundcard and CD/DVD player are shown. For frequencies > 5kHz the measurements can be faulty. This is shown by the toward negative angle descending curvature.Whilst measuring it is important that there are no other noise sources in the vicinity. Noises can cause the measurement to be incorrect because the loudspeaker acts like a microphone during the impedance measurement.

W4.5 Acoustic Phase Measurement

The acoustic phase measurement of a loudspeaker can only be carried out in quiet rooms. This test is more sensitive to disturbances than the frequency response measurement. Strong reflection in the room in which the test is being carried out can also influence results. In small rooms the distance should not be less than 0.5m.The measurement should be carried out as a continuous measurement :



The automatically phase calculation is highly complex and faults can occur. This can be easily recognized as the curve will appear to be broken up. Next to the phase button there is a drop down list with a series of adjustment possibilities to support the measurement.Whilst the continuous measurement is running one of the three ranges are selected.



The correct range has been found when the phase curve remains the same. When the curve is intact in the high amplitude range then a correct result has been attained. When measuring loudspeakers separately it is only possible to compare the phase behaviour. Phase differences of 180° cannot always be correctly determined. After the simultaneous measurement of loudspeakers the results are once again correct. Because distance measurement is not necessary with ATB PC, every measurement technician gets the same results.

W5. Basics of the acoustics

W5.1 Basic terms

Hearing
Oscillations that we perceive with the human ear are called sound and this perception is called hearing.

Sound
In physical therms sound can be seen as a wave. To put it more precisely sound is oscillations of molecules in a medium that spreads in wave form. These sound waves can move in air, water or even solid elements.

Sound transmission
Sound transmission is a change of air pressure. This causes the single molecules to collide into one another and spread. Sound is constant change of pressure level. The transmission speed in air is approximately 343 m/s (Meter per second) at room temperature. Sound waves that hit flat and even surfaces are reflected to an extent.

Sound pressure level
Human perceive sound pressure in terms of sound pressure level. The higher the pressure, the louder the tone. The distance between ear and sound source changes the perceived loudness. The further away a sound source is, the lower is the perceived loudness.
The loudness of a tone is measured in sound pressure (or sound pressure level). In physics the values are in decibel (db) and in conjunction with the human ear as dB(A).The pain threshold for the human ear is about 120 to 130 decibel. In comparison: the flattering of leaves is measured at 10 decibel, a normal conversation at 50 decibel and a starting jet plane at 125 decibel.
An increase of 10 dB(A) means that the ear perceive a doubling of the sound pressure level.

What is frequency ?
The change of pressure (in air or another medium) per second is called frequency. The unit for this is determined as hertz (Hz).
In a medium, as for instance air, the molecules themselves do not move with the speed of sound, but only its oscillation (frequency), similar to a “wave” at a football match. A human with healthy ears can hear in a range from 20 Hz to 20000Hz. As we get old this range decreases. Sound waves with a frequency lower than 16 Hz is called infrasound and higher than 20000 is called ultrasound.

The ear
The ear has three parts the outer, mid and inner ear. The sound waves go through the hearing channel to the eardrum and put it into oscillation. In the mid ear the oscillations are amplified through the hammer, anvil and stirrup and transferred to the inner ear. In the inner ear the different tones stimulate different areas of the hair cells in the ear snail and transmit them to the brain over the hearing nerves. Overloading destroys the hair cells. The ear as to say the ear snail is made to amplify tones of 3000Hz at most.

Sound source
A sound source is anything that will make air oscillate, for instance musical instruments, loudspeaker and voices.

Tone
Hearable constant oscillations.

Timbre
The combinated sound of more than one tone. With the E string of a contrabass that would be the basic tone of 44Herz and its overtones (frequency multiplications) at 88Hz, 132Hz and so on. The pressure level of the single overtones depends on the resonance body of the instrument and gives the instrument its timbre.

W5.2 Standing waves

It is generally known that in an enclosure where sound waves are transmitted through air movement, that at a certain frequency standing waves, also known as modes, build up. There are two ways to illustrate them.

The speed maximum

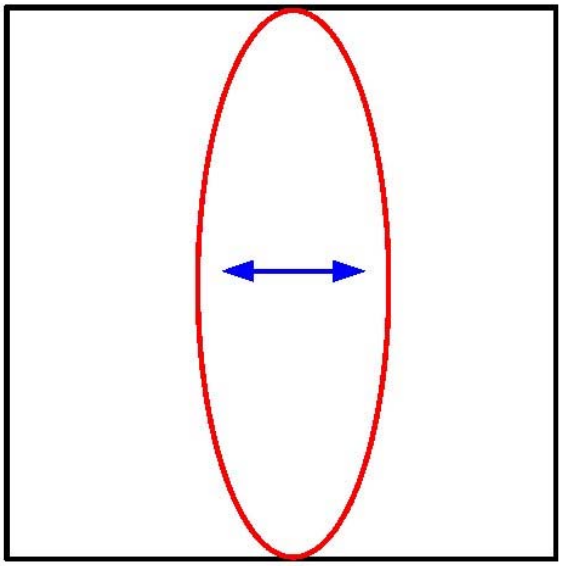


Figure 5.1

The standing wave as speed maximum (Figure 5.1) means that highest speed is given as wave maximum.

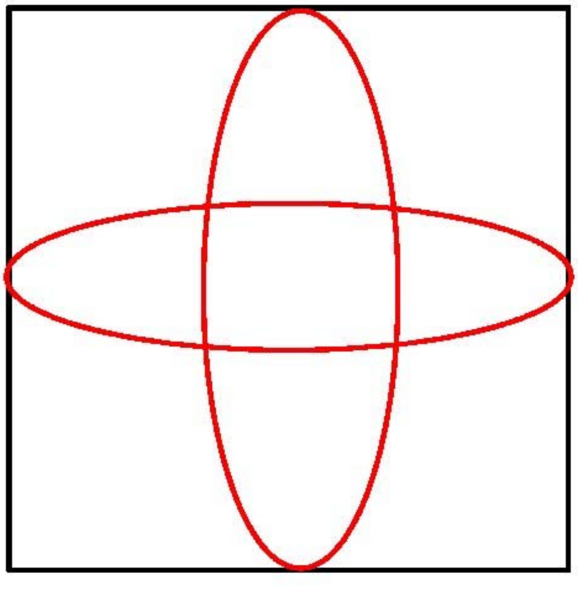


Figure 5.2

In Figure 5.2 the vertical and horizontal waves are shown.

The pressure Maximum

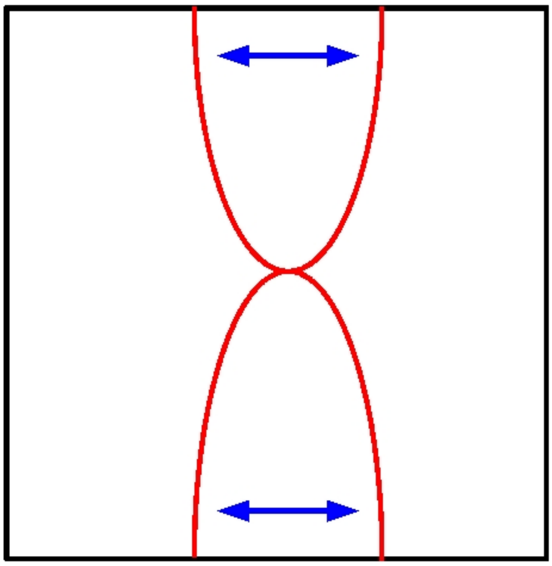


Figure 5.3

The standing wave shown as pressure maximum (Figure 5.3) means that the pressure is given as maximum of the wave.

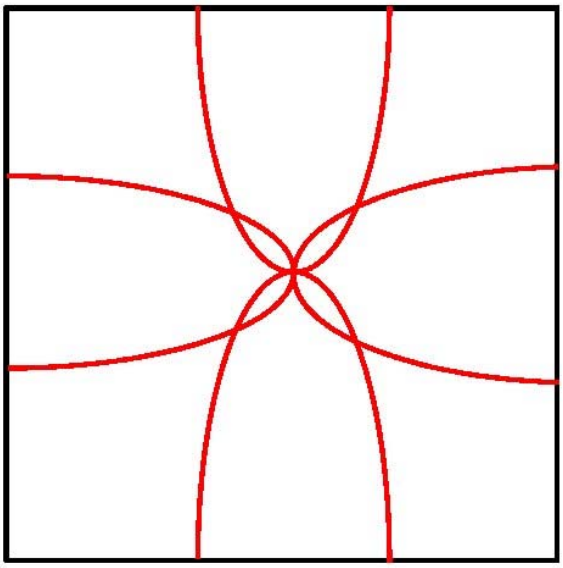


Figure 5.4

In figure 5.4 the horizontal and vertical waves are shown.

The Frequency

The frequency of these standing waves can be calculated using following formula:

f = 340 $^{m}/_{s}$ • ((1 ÷ length in Meter) • ½)

For an enclosure with a wall distance of 25 cm that means:

f = 340 $^{m}/_{s}$ • ((1 ÷ 0,25) • ½)

That follows to:

f = 680 Hz

For the speaker that means that the air builds up a resonance at 680Hz. This resonance has back effects on the membrane causing an additional time delayed sound transition. This transmission should be suppressed as it creates tones that don't belong to the sound reproduction as such.

W5.3 Sound waves in enclosures

In the following we will try with the help of simplified models to show how the conditions in an enclosure are and how the air molecules move there in.

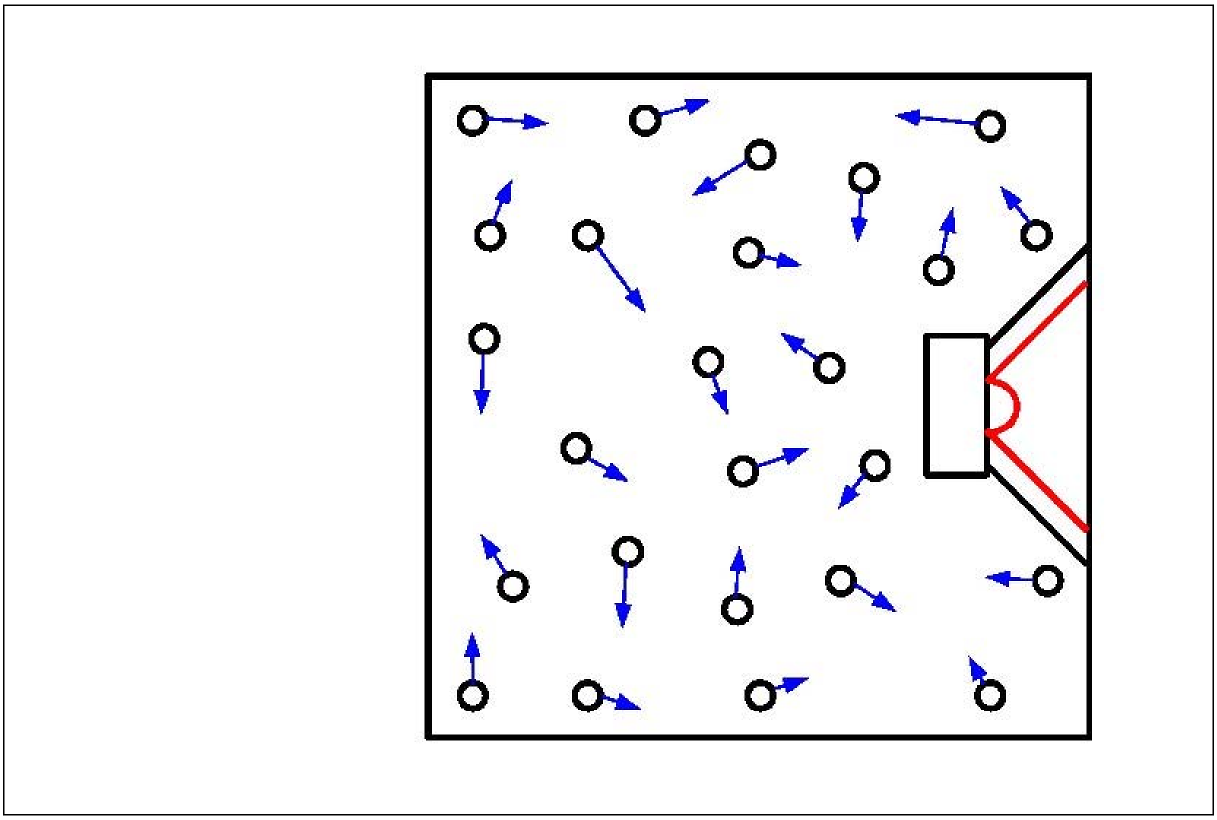


Figure 5.5

In the first model (figure 5.5) the molecules are shown as little circles. The molecules are spaced chaotically as they would be in nature. The blue arrows show in length the speed and in position the direction of the molecule movement. In the addition all powers add and subtract to zero: Thermal Balance.

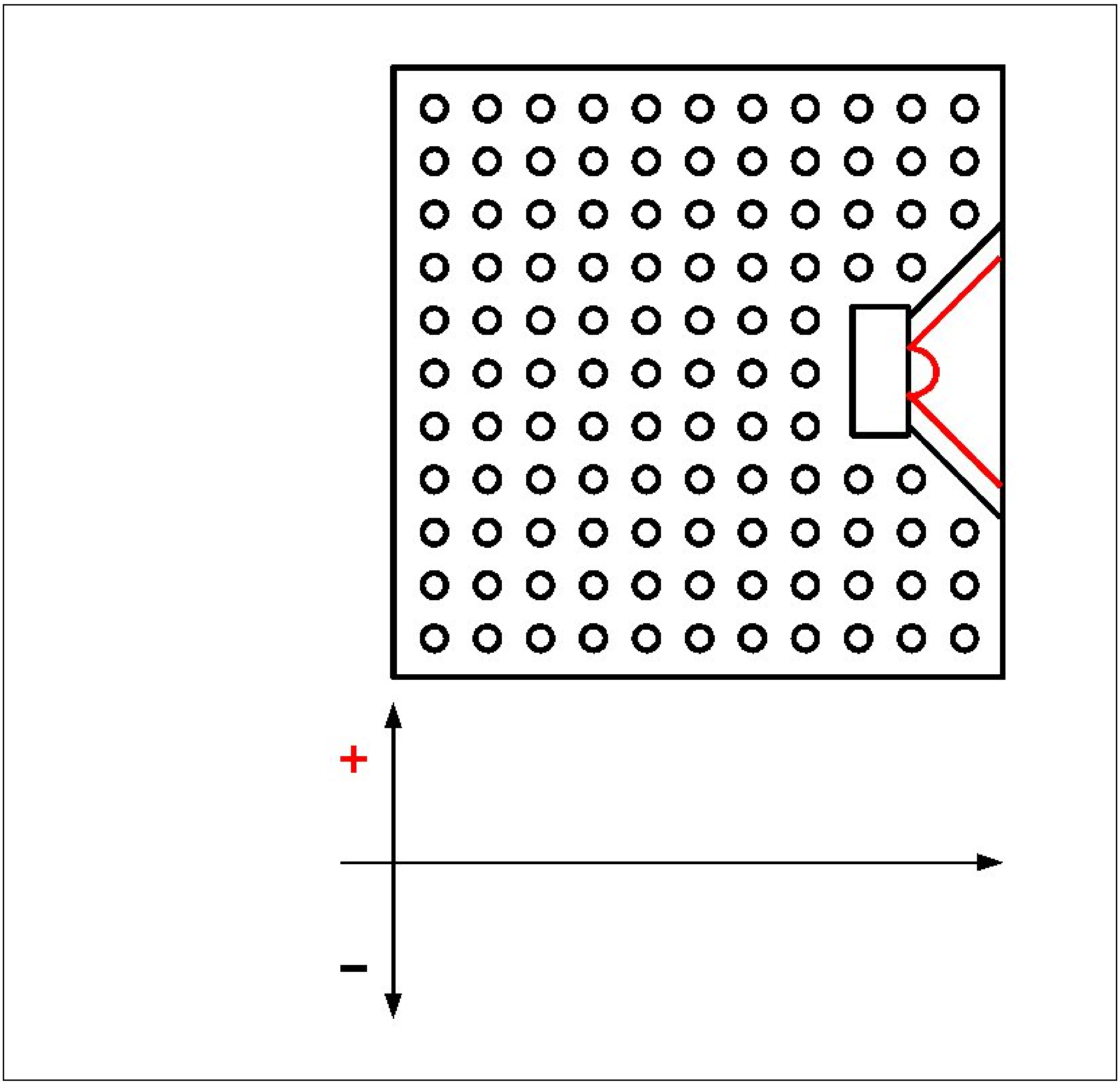


Figure 5.6

To illustrate we chose a simplified model (Figure 5.6). Underneath the speaker there is a coordinate cross illustrated. The X axis is the pressure through the speaker and the Y axis the pressure on the speaker.

In Figure 5.6 the distance between the molecules is even overall. The distance between the molecules represents the pressure on that position.

Impulse in enclosure

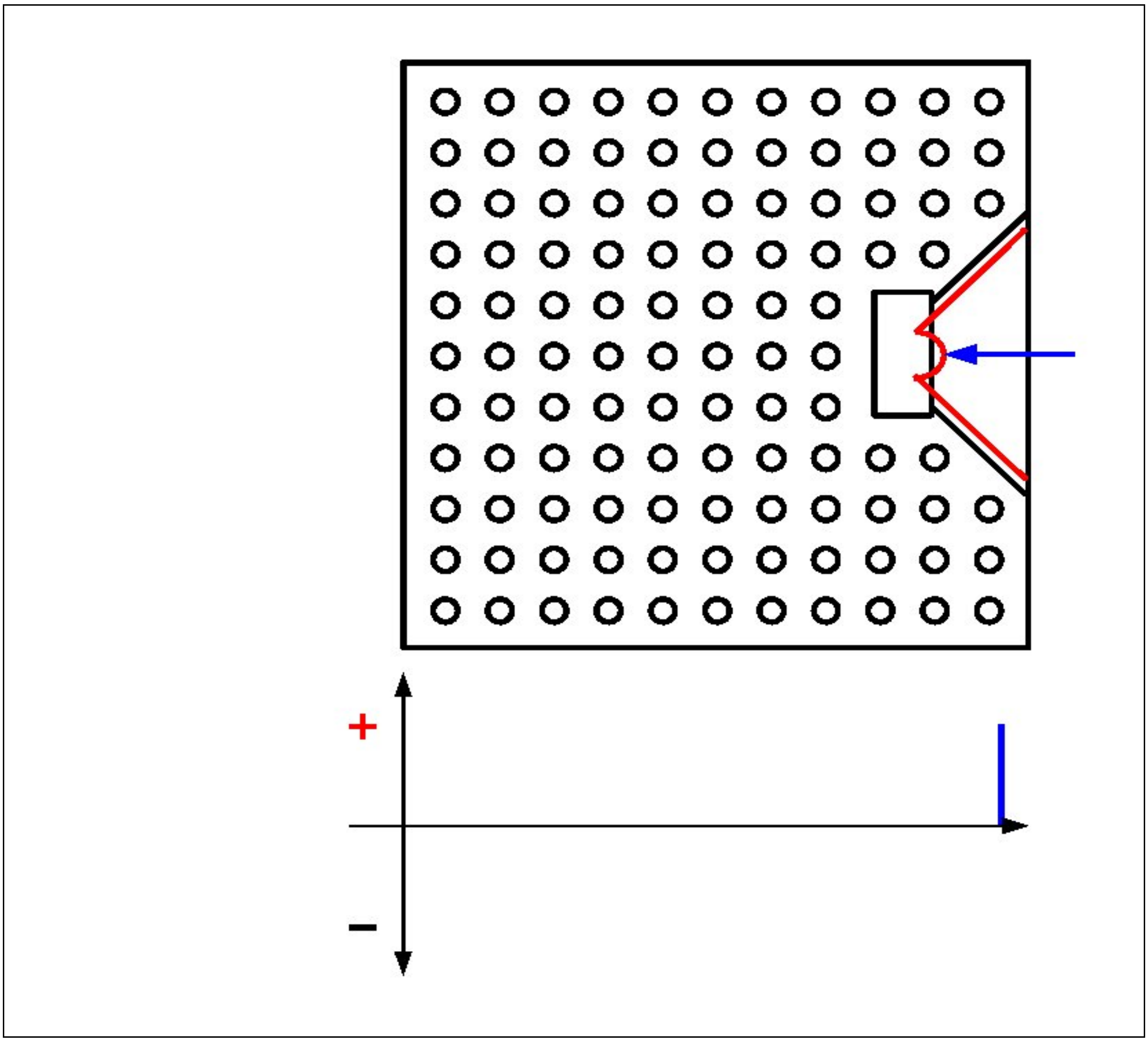


Figure 5.7

As to be seen in figure 5.7 the model series starts with a impulse. Here the speaker membrane is pressed into the enclosure. At first this results only in an pressure gain, shown as a blue highpoint in the diagram on the X axis.

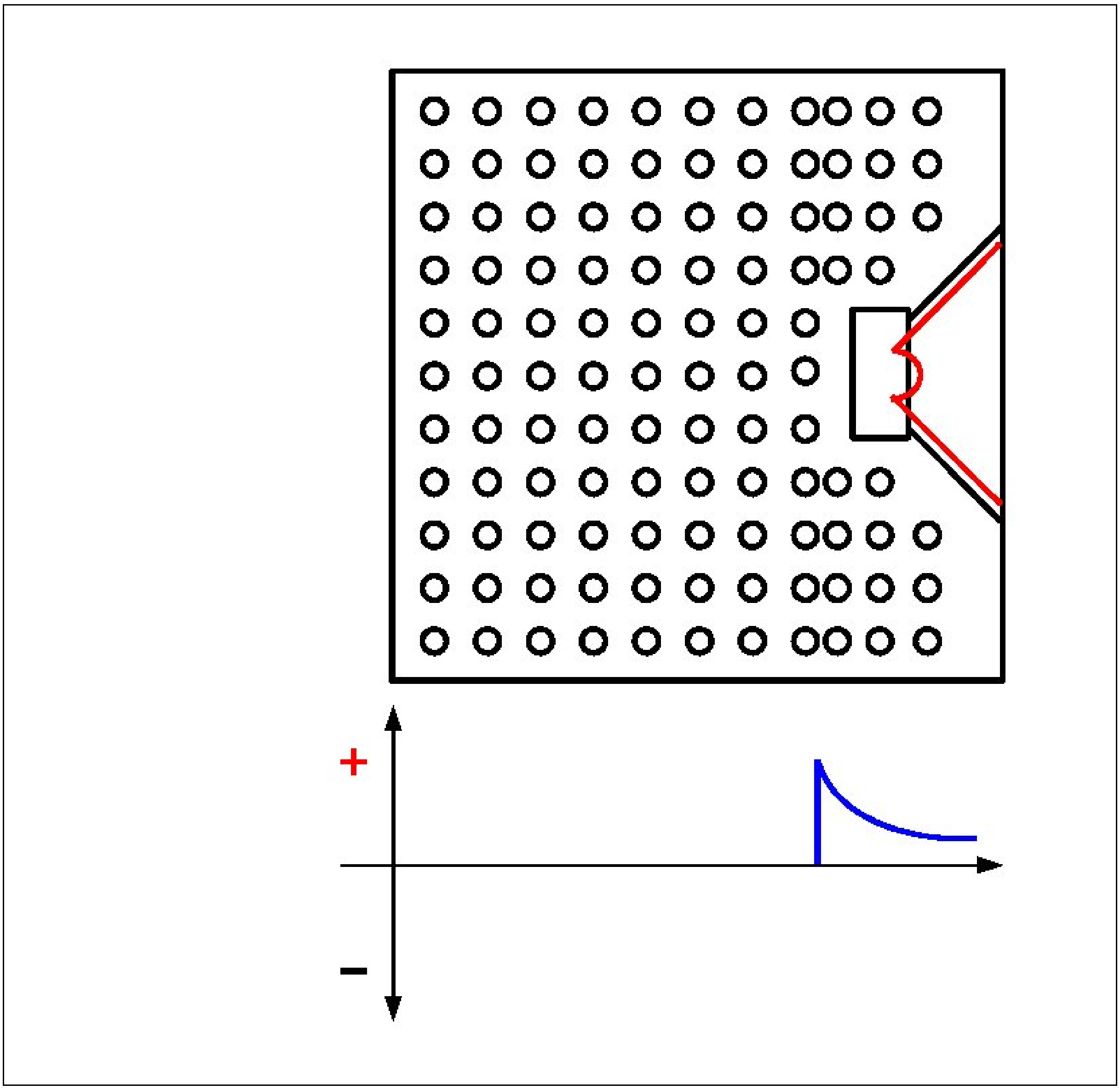


Figure 5.8

Following that the highpoint travels through the enclosure. Here then accelerated molecules bump into each other in the enclosure thus transmitting their energy. This causes a compression at that point. After the impulse the higher compression falls away as the path goes on.

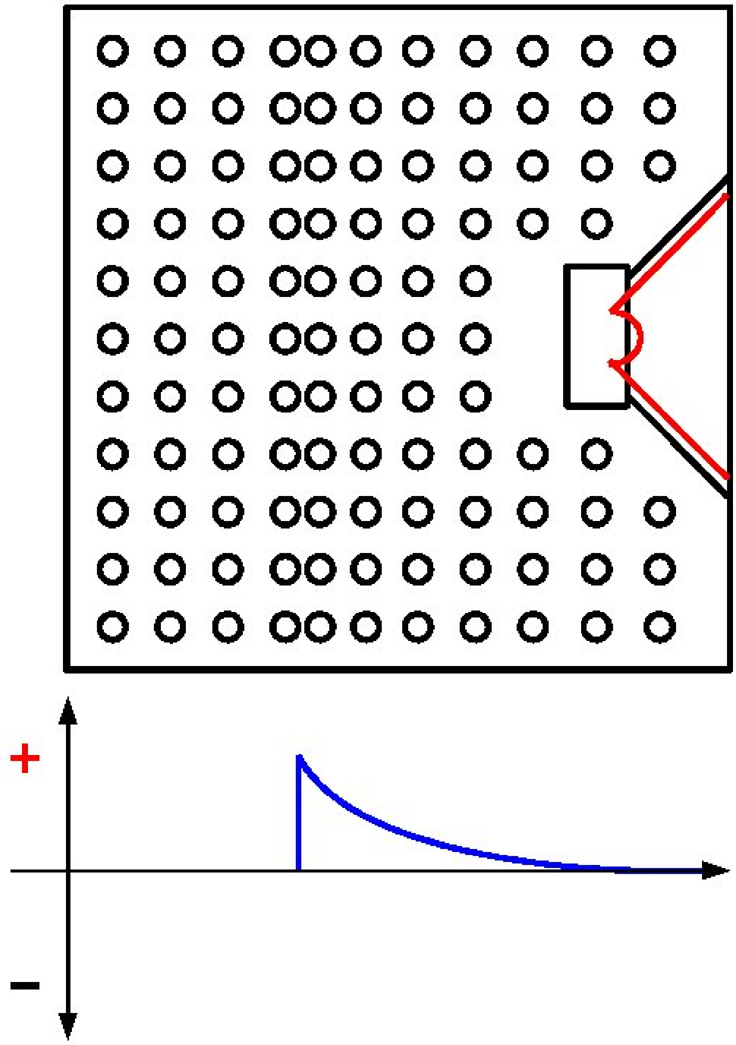


Figure 5.9

Figure 5.9 illustrates that the pressure falls back to zero at the start of the enclosure as the impulse travels through the enclosure. The impulse travels on.

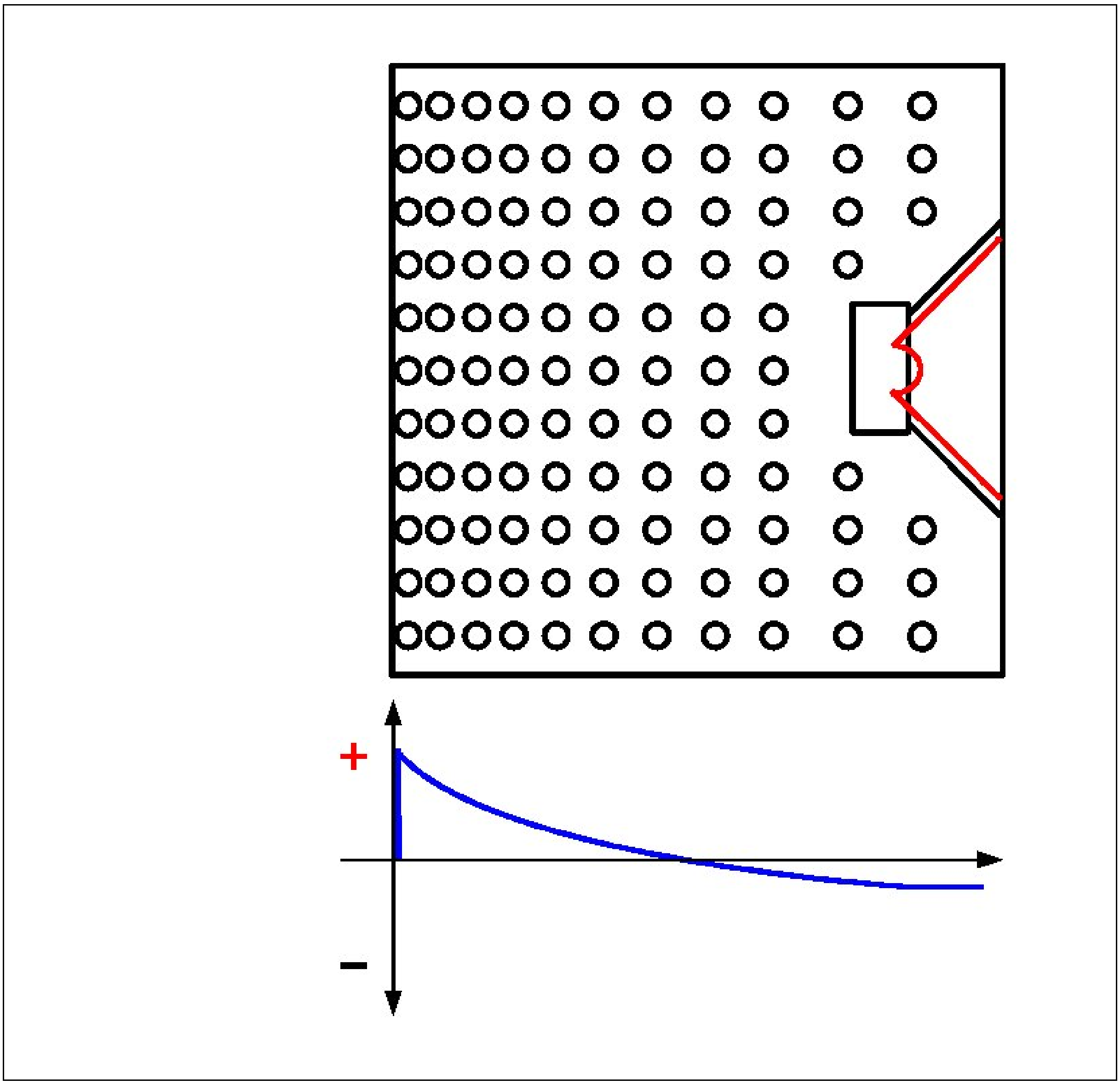


Figure 5.10

This illustration shows what is decisive for a standing wave (Figure 5.10). The mass of the accelerated molecules have not yet come to rest and cause under pressure at the start of the enclosure. This procedure is called over swinging. With a standing wave this over swing is maximal.

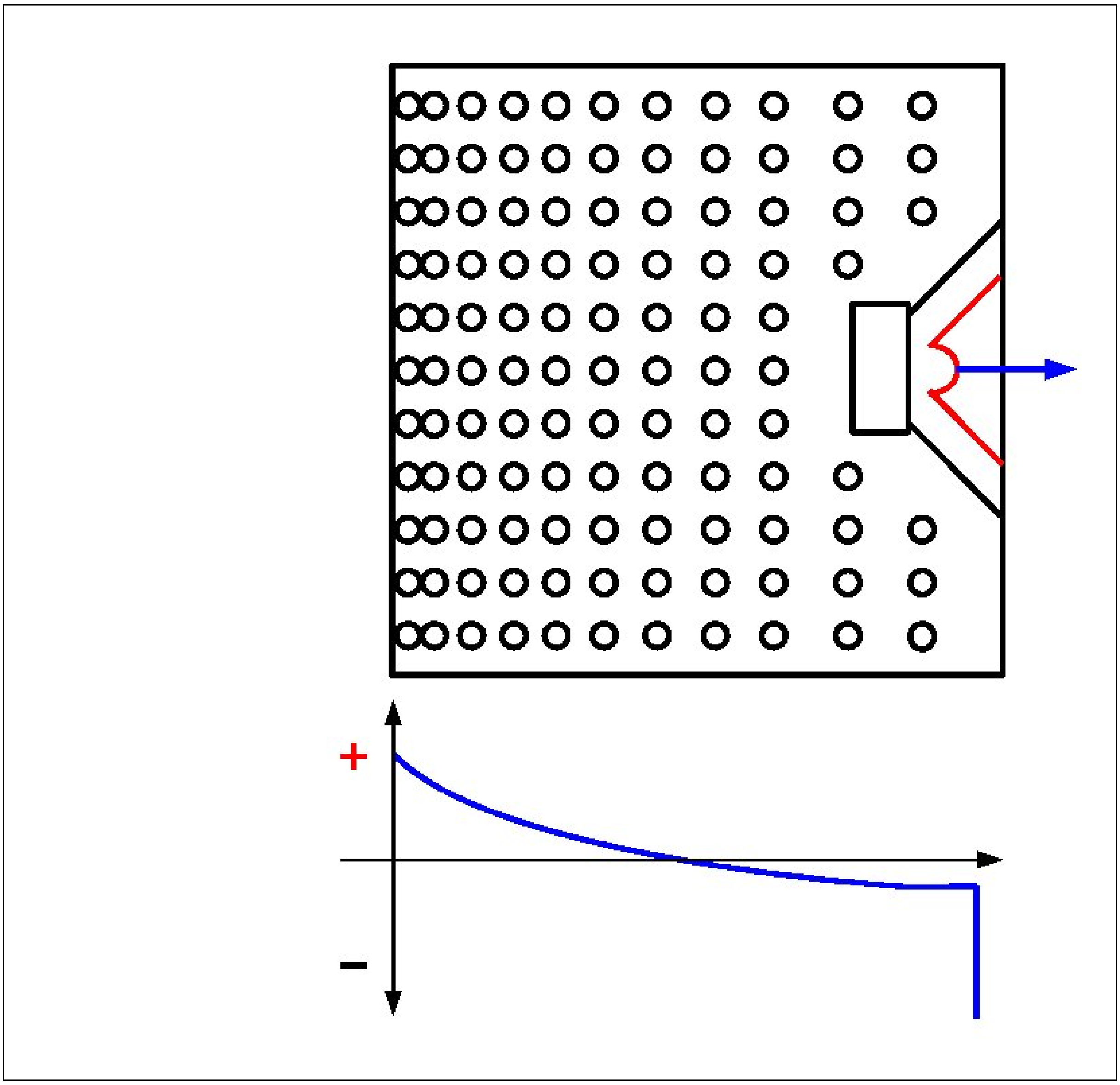


Figure5.11

The frequency determines whether or not the membrane of the speaker is swinging out at this point (Figure 5.11). For a standing wave this is necessary, then this swinging out of the membrane causes additional under pressure to that caused by the impulse.

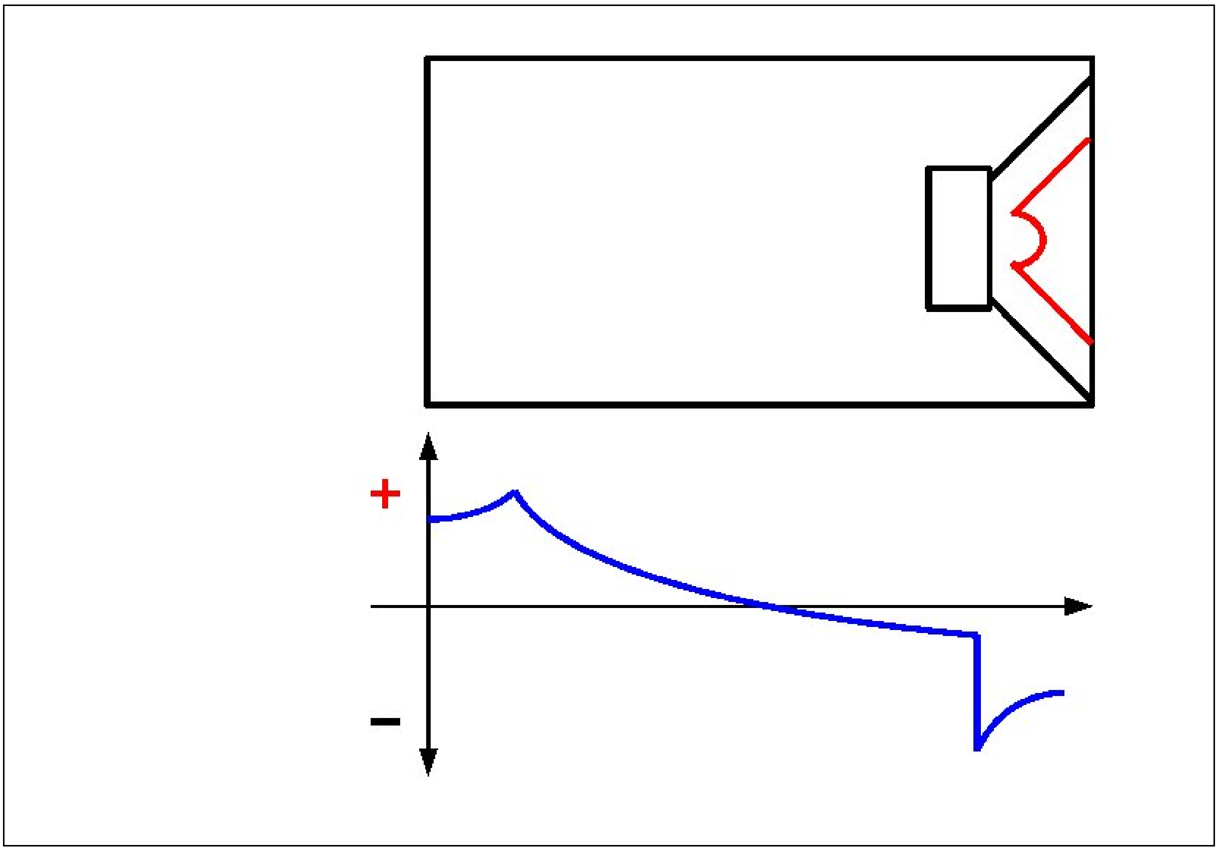


Figure 5.12

In Figure 5.12 is to be seen that now this new impulse travels through the enclosure. Additional the original impulse is also reflected back through the enclosure. The graphic has been simplified here as the motion of the air molecules were illustrated in figures 5.7 to 5.11.

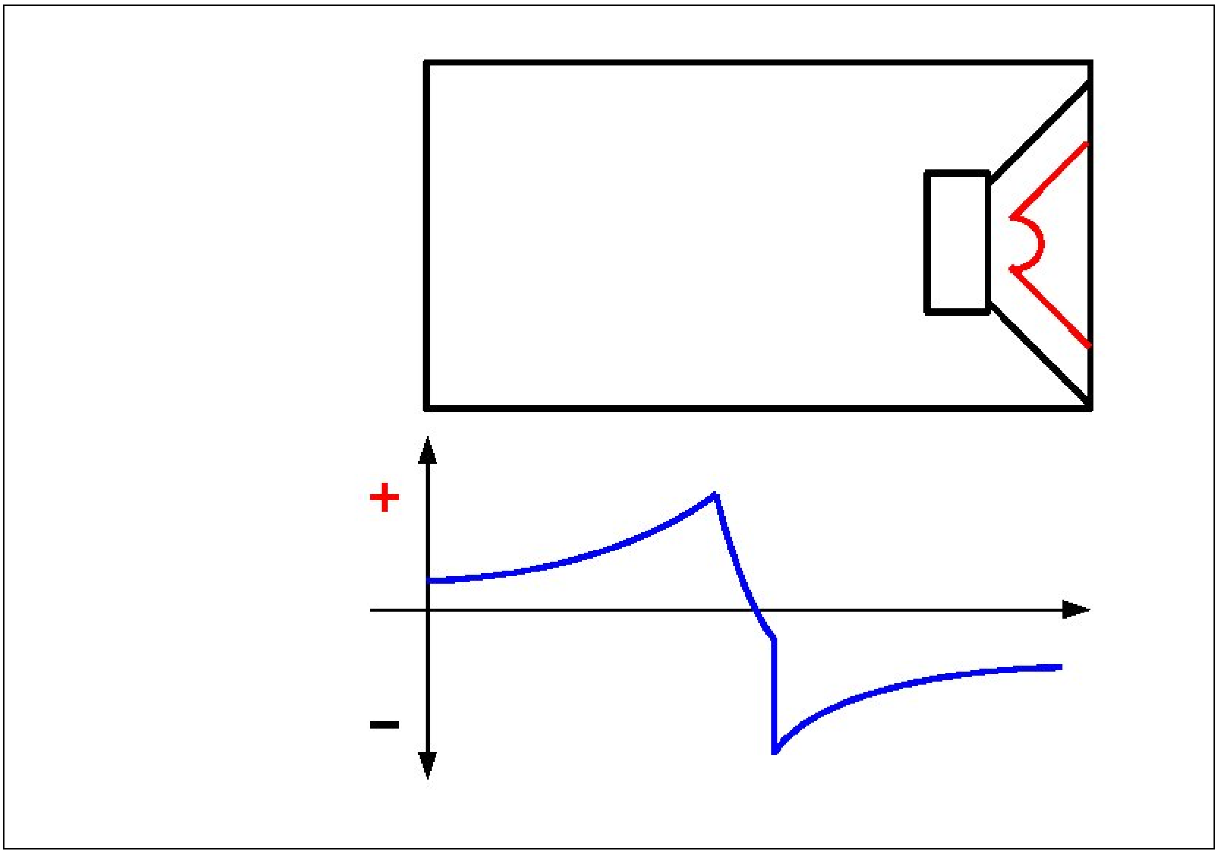


Figure 5.13

The opposite impulses approach each other and get closer, where as the pressure falls at both sides of the enclosure (Figure 5.13).

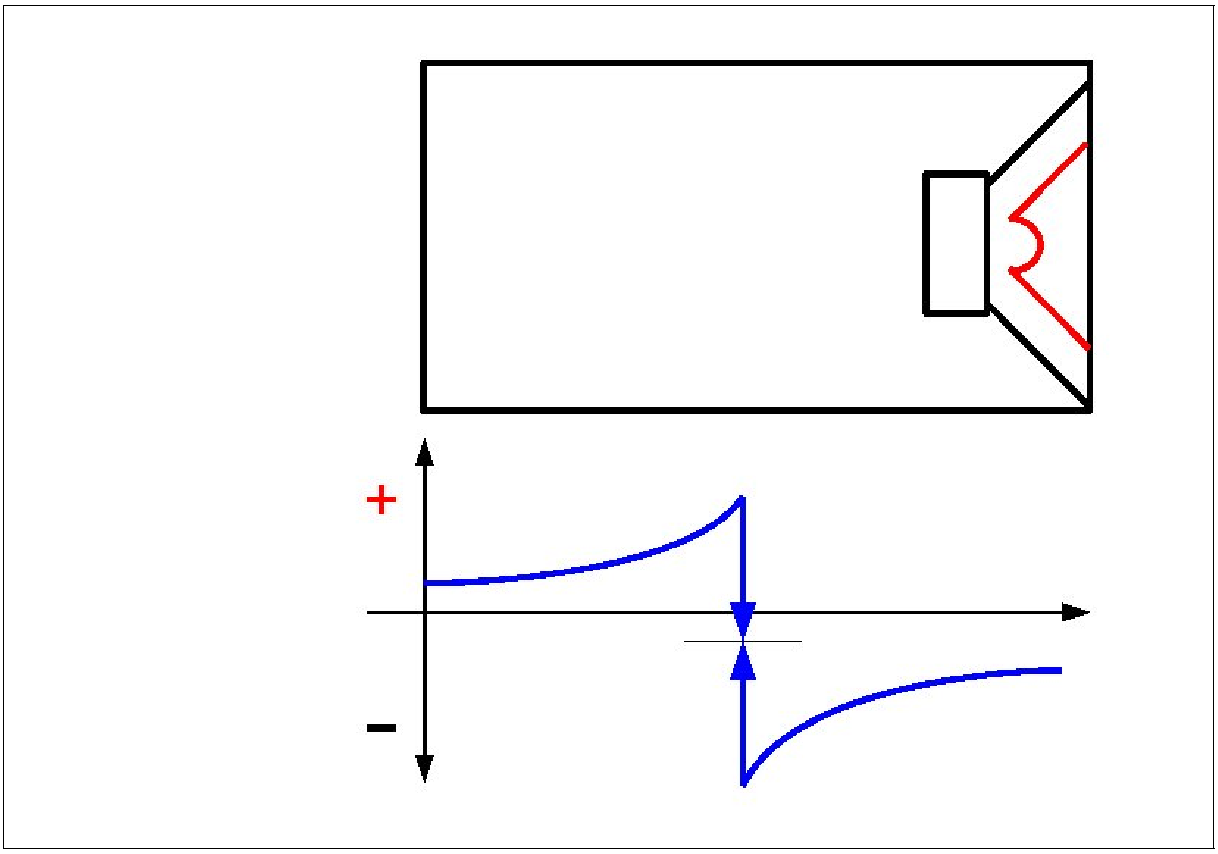


Figure 5.14

At this point both impulses reach each other (Figure 5.14). The under pressure caused by the initial impulse is shown here as value. Both pressure waves are off center to the X axis to the amount of that value. The compensation causes the high speed of the air molecules.

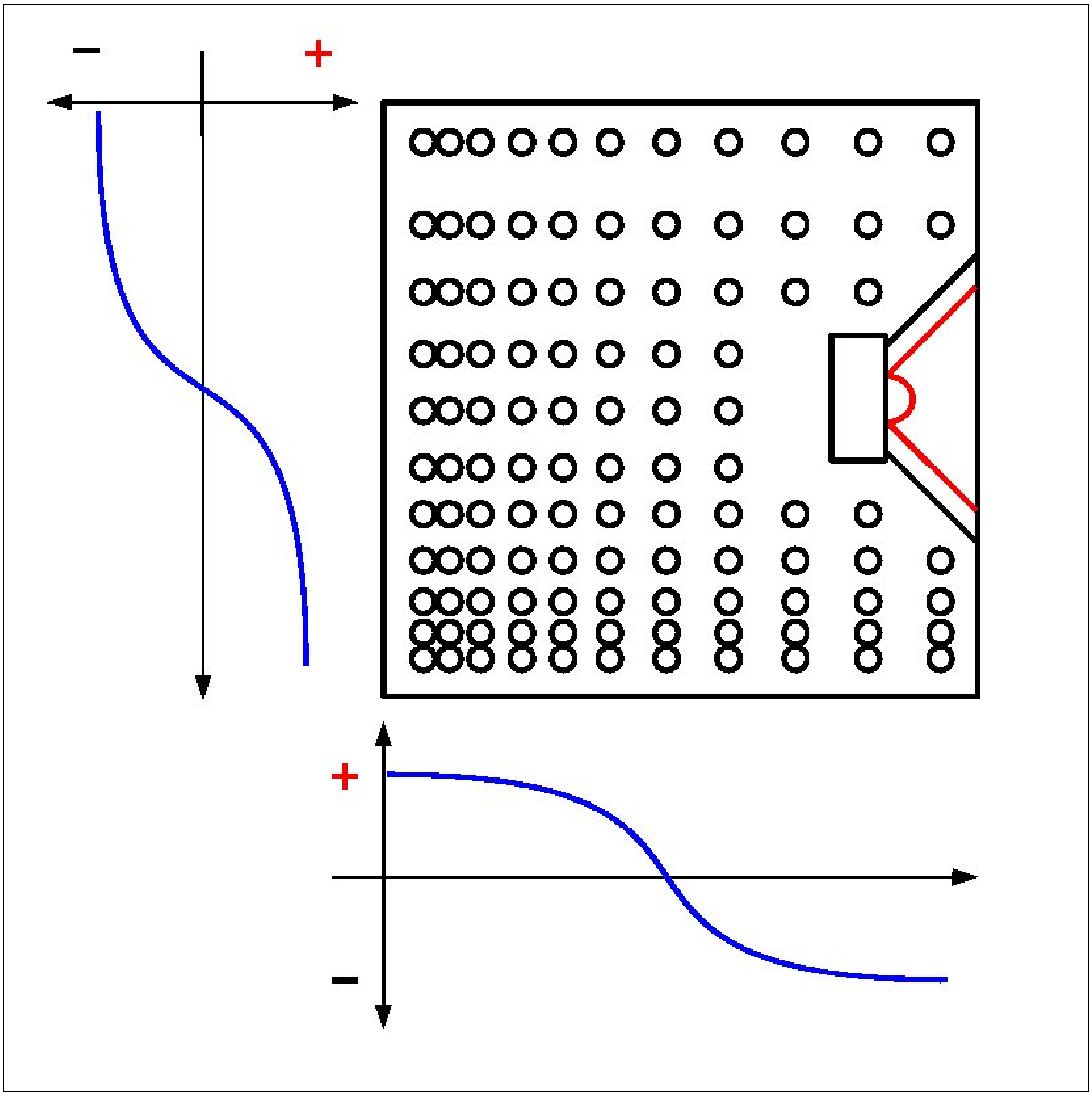


Figure 5.15

Figure 5.15 illustrates the resonant condition of the sound wave. The wave can be observed in the horizontal and vertical. It changes its pressure and under pressure maximum with the frequency. In the corners an over pressure maximum changes to an under pressure maximum. The change happens over the middle and causes a speed maximum at that position.
This means that you need an acceleration and a pressure sensor to measure such waves.

W5.4 Microphones

At this point pressure and speed measurement microphones will be introduced.

Generally speaking there are two main groups of microphones. There are microphones with kidney characteristic and microphones with circle characteristic.

Circle characteristic

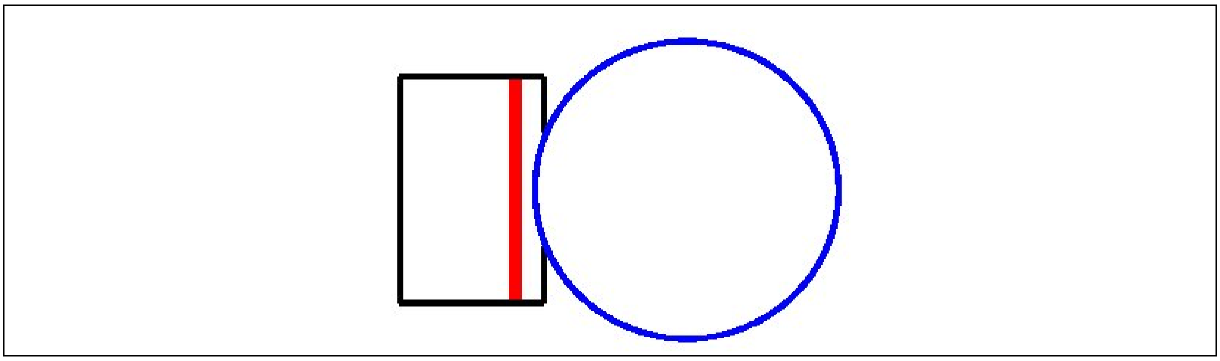


Figure 5.16

With the circle characteristic the back of the membrane (red) housing is closed.(Figure 5.16). Because of this the microphone reacts strongly to pressure change. The membrane cannot follow fast motion because of the cushion effect of the closed housing. Microphones with circle characteristic do not have directional sensitivity, that’s why their recording area (blue) is circular.

Kidney characteristic

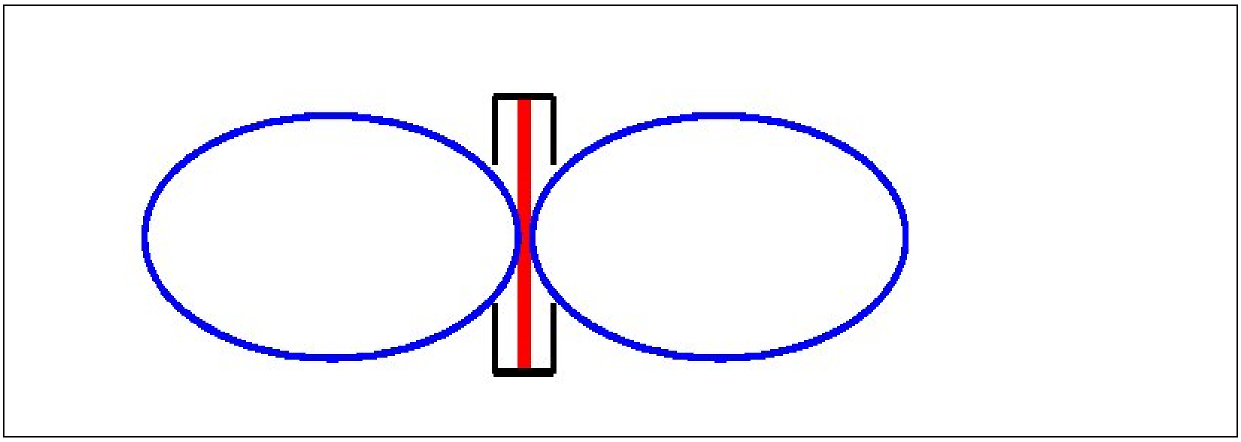


Figure5.17

Microphones with kidney characteristic have a font and back open mounted membrane (red) (Figure 5.17). That puts them in the position to record air motion caused by speed changes. There are direction sensitive, meaning they register sound sources on axis stronger that those off axis to the membrane. That’s why the recording area (blue) is kidney form.

W5.5 Sound transmission

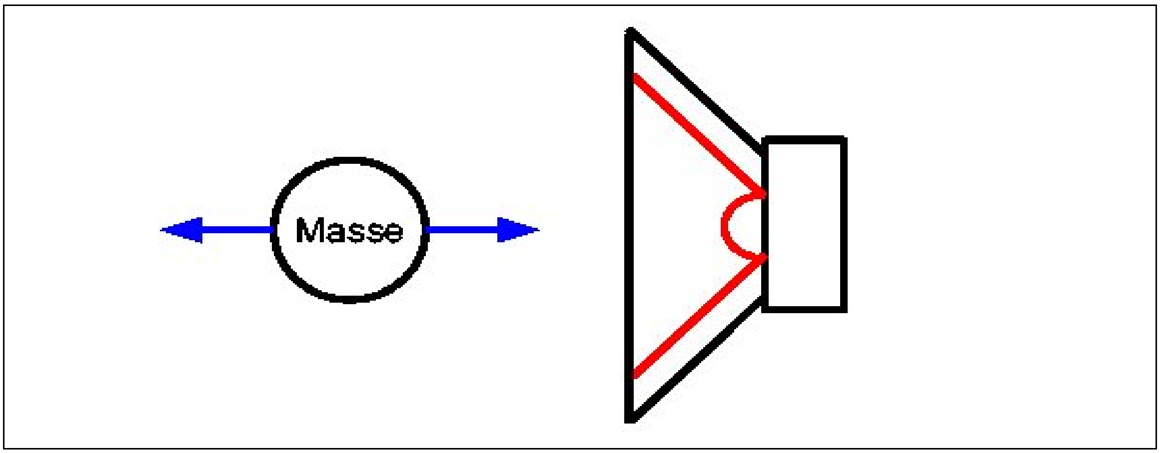
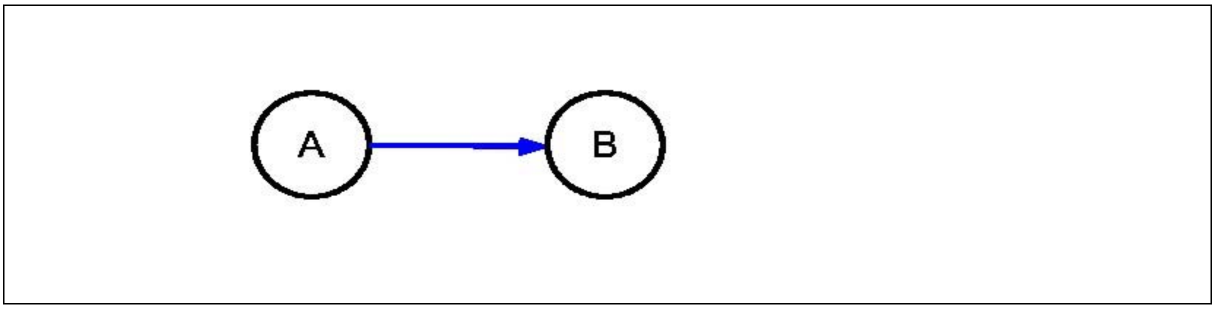


Figure 5.18

Every air molecule has a mass (Figure 5.18). If sound is to be transmitted through movement of these molecules, their mass has to be accelerated. To accelerate this mass you need energy. For a fast acceleration of this mass, well high frequencies, you need a lot of energy. If the mass is to be accelerated slowly, low frequencies, you don't need as much energy. But to turn over the same energy you have to move far more molecules for low frequencies'.



Figur 5.19

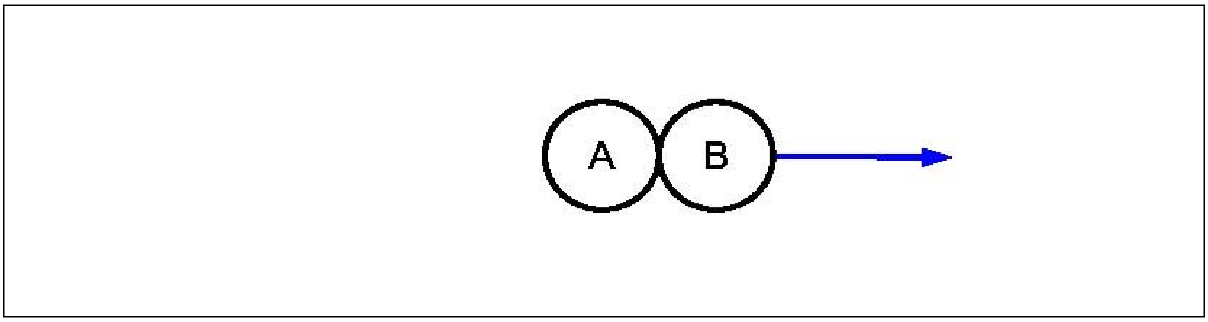
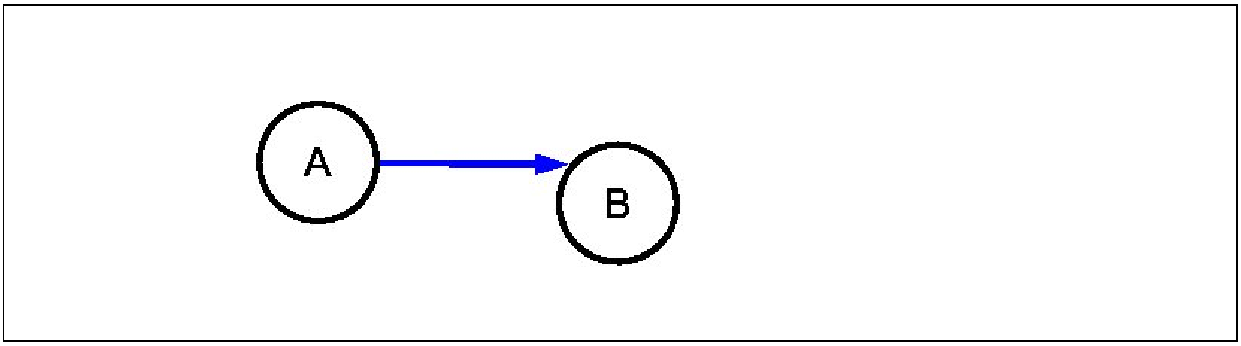


Figure 5.20

The high energy needed to transmit high frequencies' is only passed on optimally with a direct hit (Figure 5.19 and 5.20). This is the reason why high frequencies' transmit directionally. Then only the direct path has the efficiency for transmission.



Figur 5.20

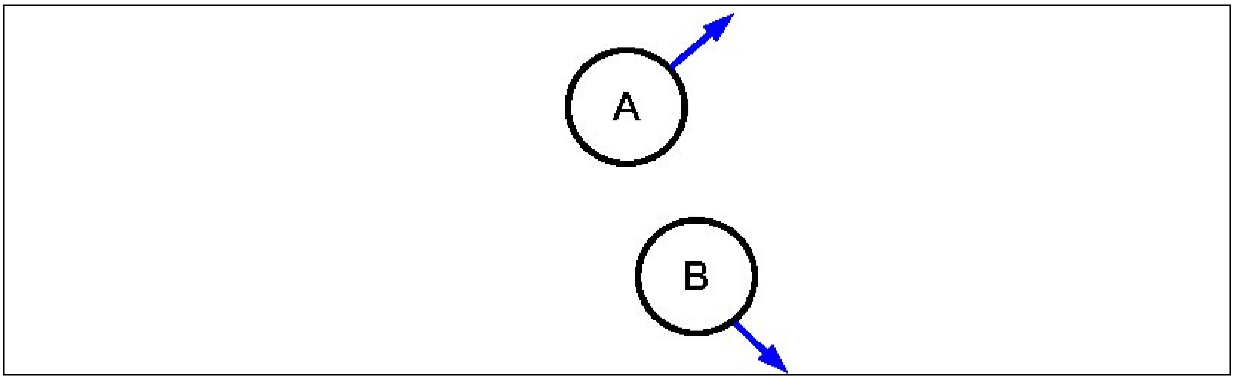


Figure 5.21

If a fast accelerated molecule hits another molecule that is not in the direct flight path (Figure 5.20), it will not be able to transmit the whole energy to the next molecule and flies off in a different direction with part of the energy (Figure 5.21). If it then hits another molecule it repeatedly can't optimally transmit its energy.

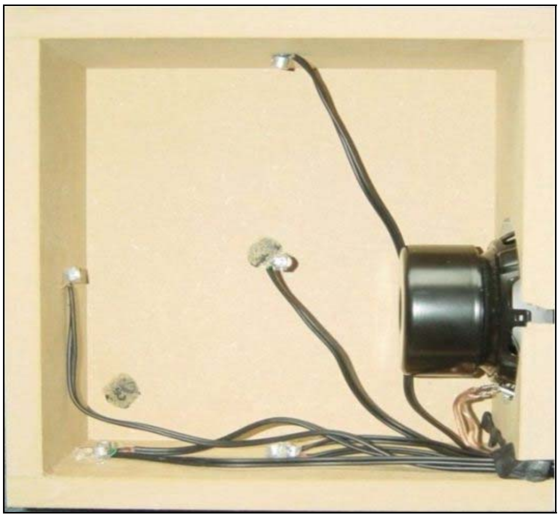
This explains why high frequency bundle, then only the direct path has the efficiency needed for transmission.

In the low frequency range the molecules are only stimulated to a low energies level through the slow oscillation. This energy level is not much higher than that of their natural condition. That means that the energy can be passed on with indirect hits and change of direction.

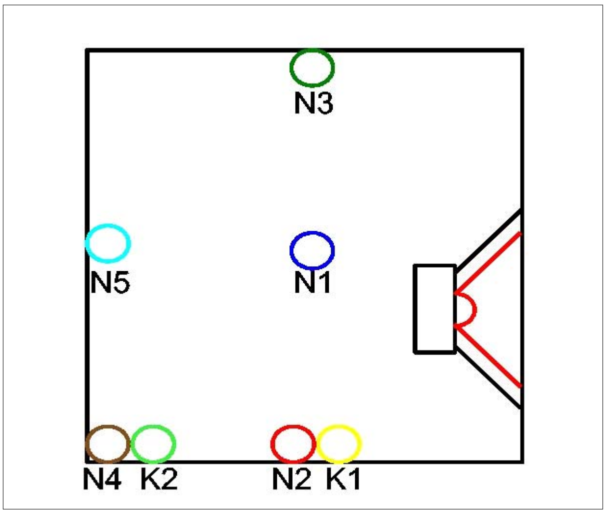
Transmission can also be described through the radiation resistance. At low frequencies it has a low value. The sound pressure is built up by lots of molecules with low speed. For that you need big membranes. At high frequencies the same pressure level is built up with fewer molecules with high speed. Because of the high radiation resistance at high frequencies, the surface area of tweeters is small. A tweeter needs only a small membrane, but has to transmit the high speed to the molecules. This is only possible with a small dynamic mass, using a light membrane and drive coil. Because of the light voice coil the power ability is set to bounds. High power is achieved by using a horn as these have a higher radiation resistance.

W5.6 Measuring in empty casing

Measurement 1:



Figur 5.22



Figur 5.23

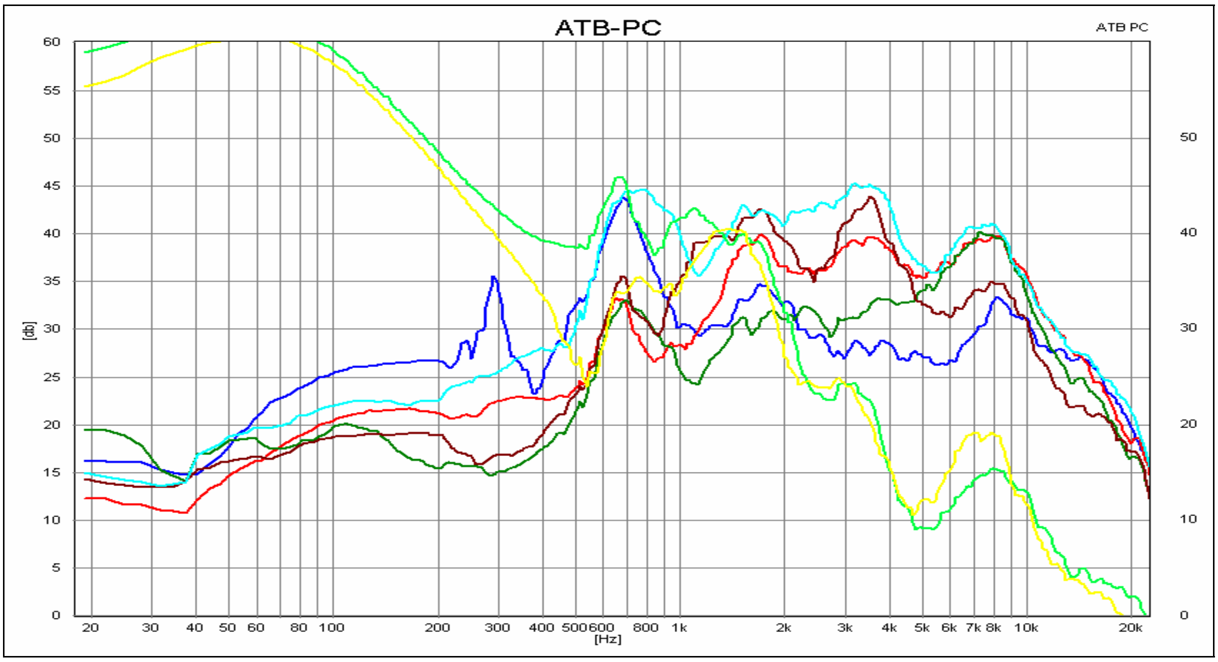
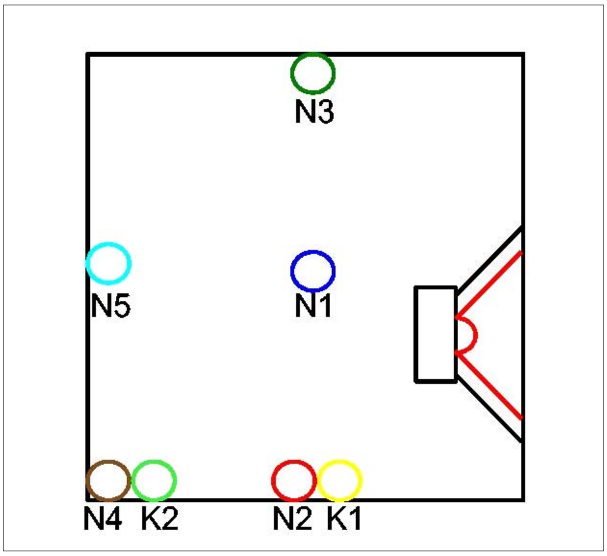


Figure 5.24

As shown in Figure 5.22 the first measurements were made in a closed casing. As calculated, all microphones show an strong increase, due to standing waves at 680Hz..The acceleration sensor N5 lies unfortunately directly opposite the sound source and gets because of its directive qualities a large part of the energy. N1 shows the maximum speed in the middle. N2, N3 and N4 show the speed due to the constant pressure change, which is near to identical in all corners. With the sum K2 you can also recognise the horizontal and vertical pressure maximums in the corners.

Mesurement 2:



Figur 5.25

Figur 5.26

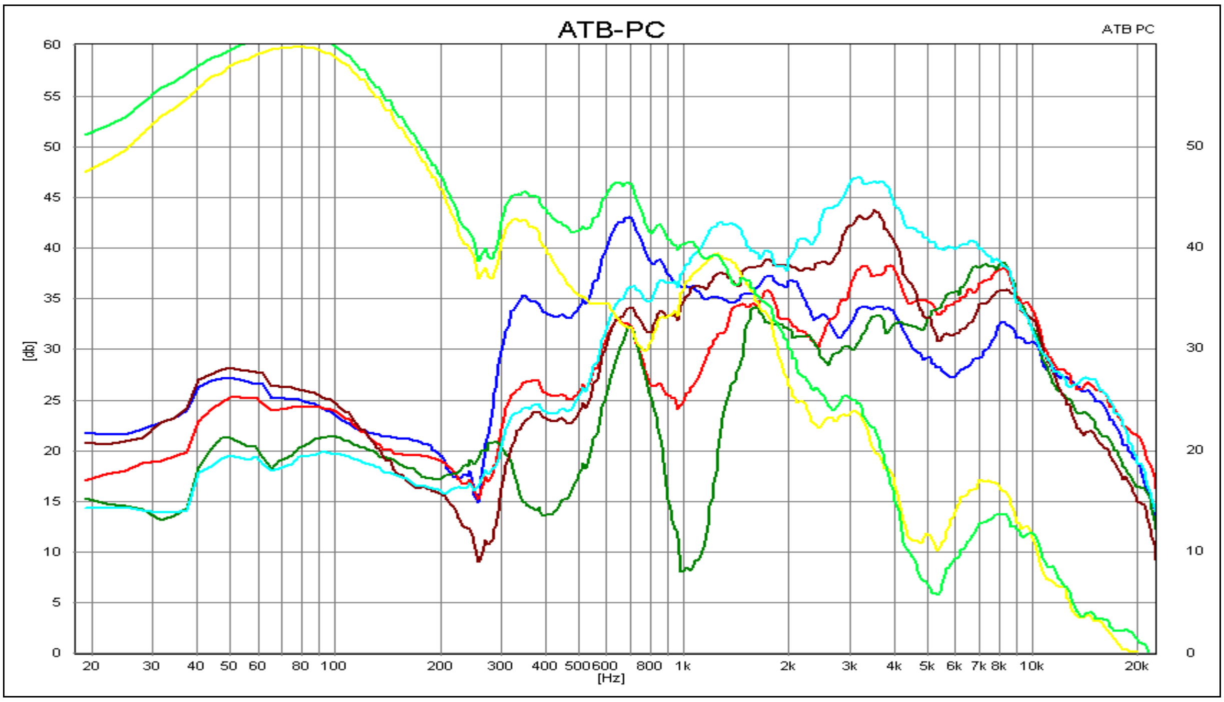


Figure 5.27

In the 2nd measurement the influence of a right-angled board is looked into. As shown in Figure 5.27 the standing wave at 680Hz is still there. In addition to that there is a new one at 340Hz! This is not prominent as the first one but still well noticeable. N1 shows the to be expected speed maximum in the center.

Measurement 3:

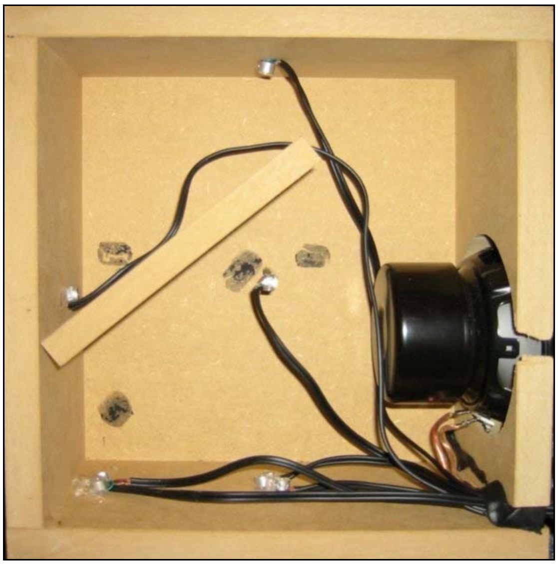
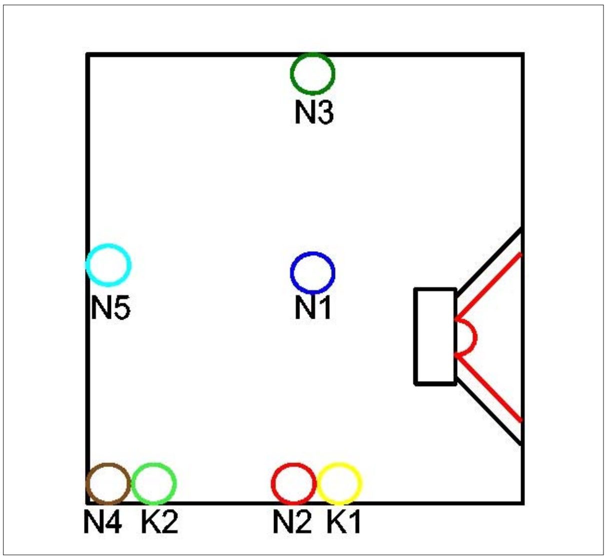


Figure 5.28



Figur 5.29

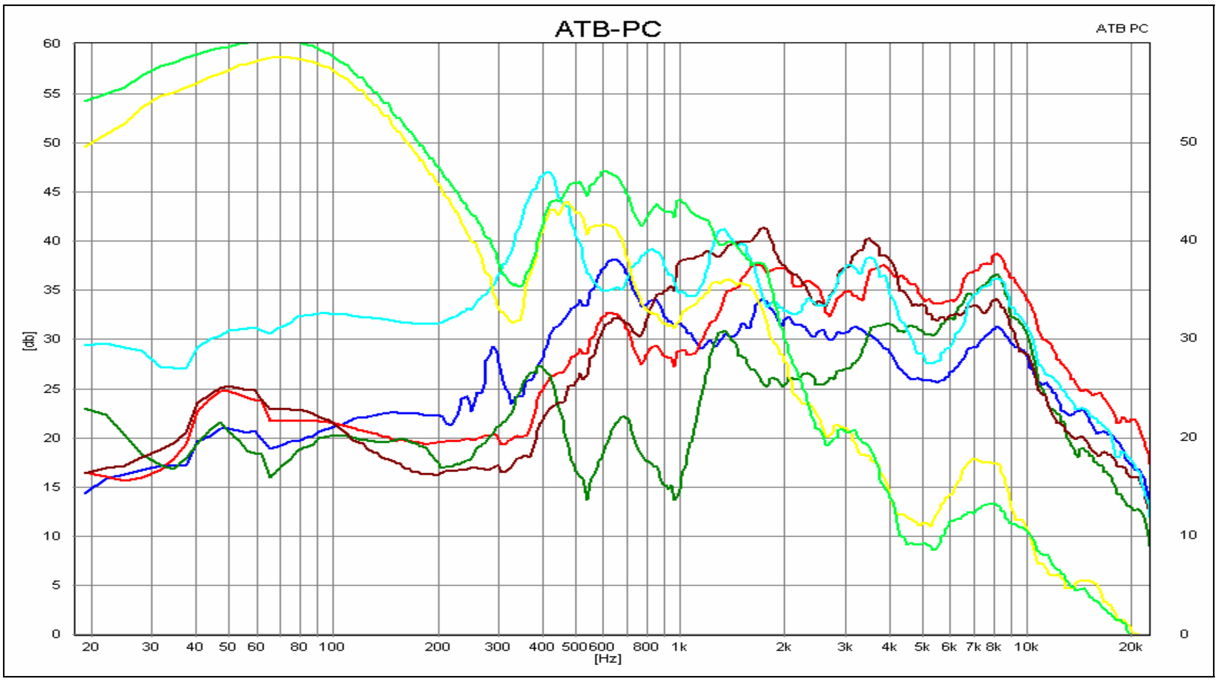


Figure 5.30

The last measurement shows the influence of a slanted board. The board splits the inner space nonsymmetrically. With this two different large air volume spaces are created above and below. Two different masses are hard to be put into resonance. In Figure 5.30 only a small rest of the resonance is to be seen that is wide spread.

W6. Acceleration sensor

The acceleration sensor is used to measure vibrations on speaker casings, car body's or machine parts. A acceleration measures the actual substance vibrations only and not the air sound waves. During the measurement the vibrations are transmitted to the sensor casing. In the casing there is an elektret foil, whose mass works against that of the casing. Thus creating a voltage that relates to the casing vibration.

W6.1 Measuring

The sensors are not gauged, the shown amplitudes are not defined values. The measurable frequency range is 30Hz – 3KHz. The sensor shows the frequency of the vibration and by comparing you can suppress the vibrations by choice.

W6.2 Mounting

The sensor is fixed to the measurement object with two sided sticky tape.

W6.3 Connection

The sensor is connected to the microphone input of the soundcard. Here the sensitivity has to be increased by turning on the Mic- boost or microphone amplifier in the system control.

W7. Design

Designing Small speakers
From dipl. des. Waldemar Namyslo

The most noticeable property of this design study is the material separation and because of this the focus on the X –Jet Tweeter and the Low/Mid-speaker. The step due to the specific the construction of the speakers is to be masked. Through the fact that both chassis are embedded in the same material the relationship between them is enhanced. The front blenders in which the speakers are embedded are interchangeable. And in this manner each box can be given an individual note.



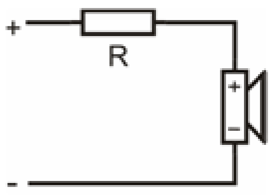

W8. Crossovers

W8.1 Crossover function (Passive)

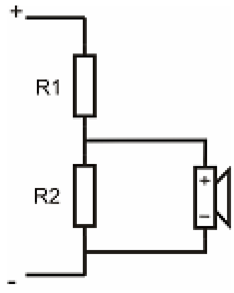
Crossovers are needed for loudspeaker combinations with more than one speaker. It has the job of splitting the frequency range into different ranges. The aim is to give each speaker the range of frequencies in which it works best or for which it was constructed.
A tweeter can not process low frequencies that is to say it can destroy it and a woofer can’t follow high frequencies because it’s too heavy. These split ranges are called paths. If then a tweeter and a woofer are used a two way crossover is needed. The first path channels the high frequencies’ to the tweeter and blocks the low frequencies. That is called a high pass. The second path channels the low frequencies to the woofer and blocks the high frequencies. That is called a high pass. If in addition a mid range speaker is used then a 3 way crossover is needed. This third path channels only the mid range frequencies to the mid speaker. This is also known as band pass crossover as it only lets a certain band with of the signal pass.
There are also constructions that use two woofers. As long as both woofers use the same frequency band. These are also 2 way speakers. Which in effect means that not the number of speakers is the criterion, but the amount of signal paths when we refer to one, two, three or what ever speaker.
The crossover is connected between amplifier and speakers and is made of resisters, coils and condensers.
Ideally the crossover just splits the frequency into different ranges without further distortion. Generally speaking it can be said use as many parts as needed, but as few as possible. The crossover should the due time spent on it as it carries the sound of the box in a most particular way. All the same a crossover can only “show the way”. Even best of parts can’t tickle out a frequency sweep from a speaker that is not able to reproduce. That´s why the Thiele-Small-Parameter and the frequency sweep of the speaker have to be found out or known before the development of the crossover.

W8.2 Parts for a crossover

Resistors
Resisters function linear, which means the frequency has no remarkable influence on the resistance value. A resister, as the name says, resists the current flow in a conductor path. The resistance value determines to what measure the current flow is weakened. A resister cannot use the current for other users, but only turn it into warmth.
When using resisters it is therefore imperative to find out how much energy is used up as to avoid over heating. The unit value of resistance is measured in Ohm [**Ω**], the formal sign is R. They are used to influence the speaker an for fitting the speaker to the right level. This can be done by putting them in row to the speaker (Figure 8.1) or by making a resistance splitter (Figure 8.2). A resister should only be used in row up to app. 5dB otherwise the overall resistance will get too large.
From 5dB onwards a resistance splitter is the best bet, as then the overall resistance stays virtually constant. In the resources you will find a table for resisters for values from 1dB to 20dB.
These circuits are only used for mid and high tone speakers as the power loss would be too high for woofers. That only in the most seldom of cases and when then the use of another speaker should be thought about.



Figur 8.1



Figur 8.2

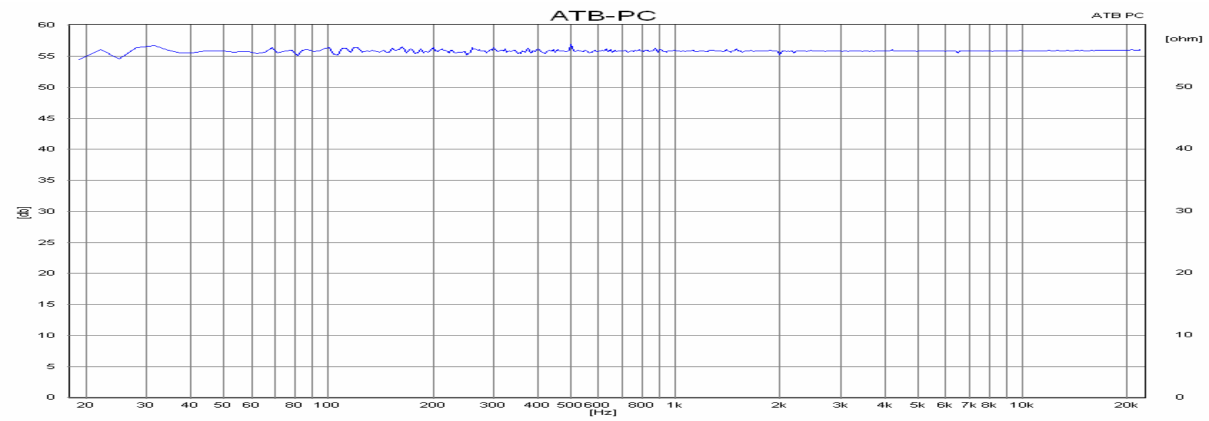


Figure 8.3

The blue sweep (Figure 8.3) shows the resistance of a 56Ω resister over the frequency 20Hz until 20kHz.

Figure 8.4

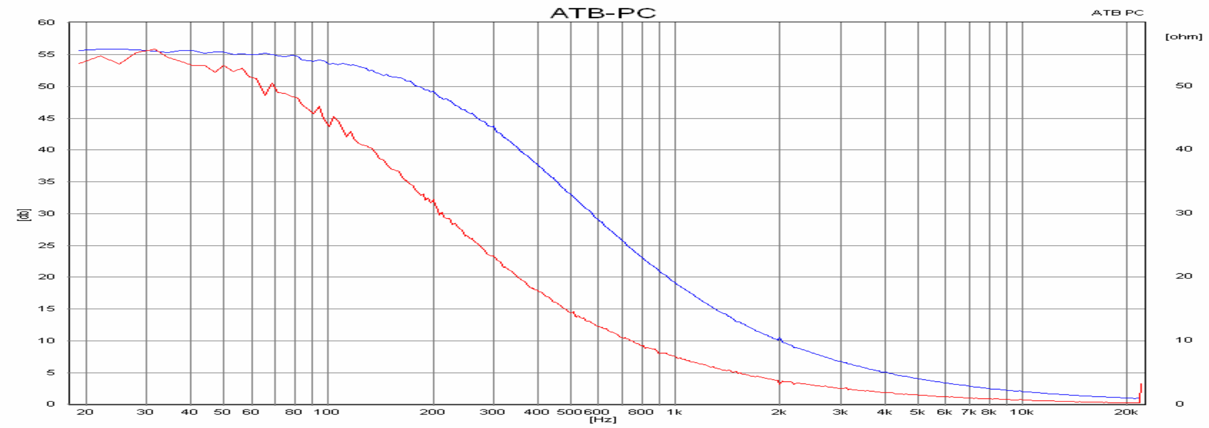
Figure 8.4 shows the measurement setup (switch position impedance).

Capacitor

Capacitor function is none linear, meaning the frequencies influence its resistance value. The function of a condenser is not as simple as that of a resistor. A condenser is made of three plates, two outer plates are conductive and have connecters. The middle plate is an isolation foil, dielectric, and non conductive. This foil can be made of plastics, paper and other isolating materials. The aim is to get potential carriers to the opposite side without a current flow. If a direct voltage is connected to the condenser, electrons are pushed to one side of the plate and drawn from the other side until the potential has built up and the plate is fully loaded (potential build-up). This process takes a certain time and depends on how big the condenser is and how high the charging current is. The condenser only acts conductive during this charging time, after that it acts as a break.
A music signal is an alternating current, which means because of the constant change of polarity, charging and discharging also consists. If the condenser manages to charge up quick enough, it becomes a high resistance or break. As with rising frequency the condenser doesn't have time to fully charge it lets the signal more and more through with out resistance. The resistance is high at low frequencies and low at high frequencies'. That is why it is possible to separate low and high frequencies with a capacitor.

There are two things to be aware of concerning capacitor:
The voltage resistance must be high enough. With rising voltage resistance the packet size rises because the dielectric needs to be thicker.

Because of the quite high currents you have to be aware of the loss factor. The loss factor expresses the non ideal isolation of the dielectric layer, resulting in electrical resistance that causes heat.
The value unit of condensers is Farad[F] and formal is C. Condensers for speakers lie in the range of µF [$1\mu F = 1x10^{-6}F$].
The higher the value the sooner the resistance sinks.
To illustrate this there are two sweeps in Figure 8.5 on the next side.
The red sweep shows the resistance of a 22 µF condenser, blue that of a 8.2 µF condenser over a frequency range of 20Hz until 20kHz.
The condensers were connected with a 56Ω resistor parallel
The resister is necessary to avoid over powering the input, as the resistance rises over 50Ω at low frequencies.



red = capacitor 22µF
blue = capacitor 8.2 µF

Figure 8.5

The sweep shows that the resistance sinks as the frequency rises. On the vertical axis the resistance can be read and on the horizontal axis the frequency.

Figure 8.6

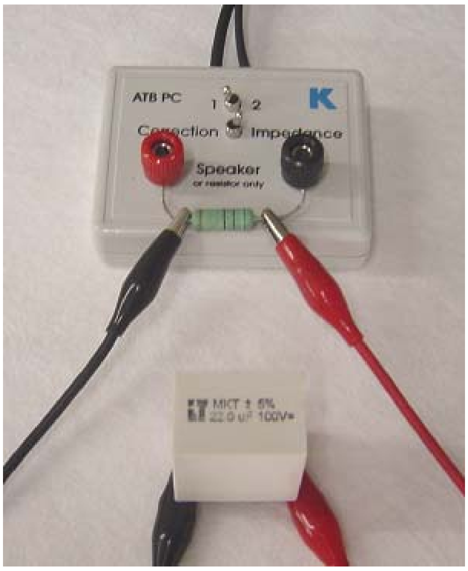
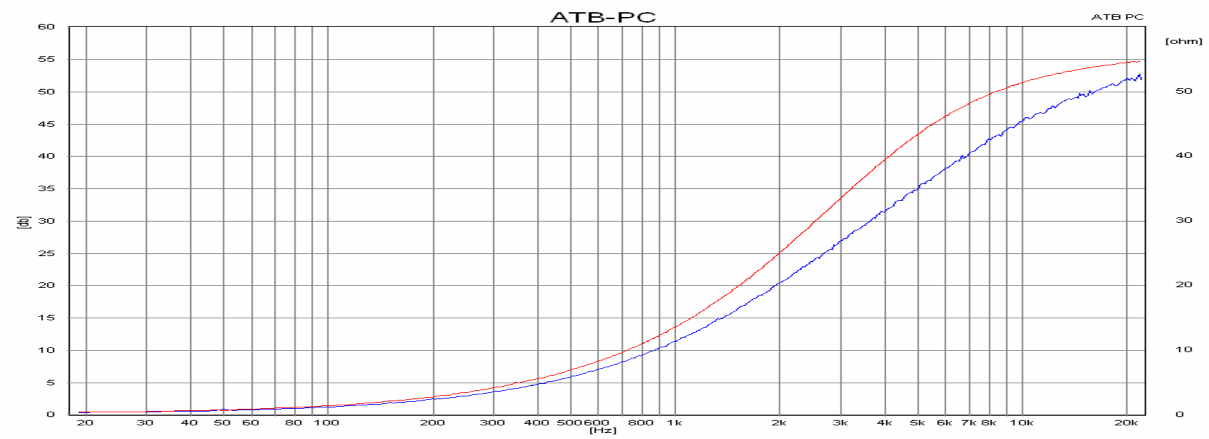


Figure 8.7

Figure 8.6 shows the measurement setup (Switched to impedance) with 8.2µF, the Figure 8.7 the measurement setup (Switched to impedance) with 22µF.

Coils

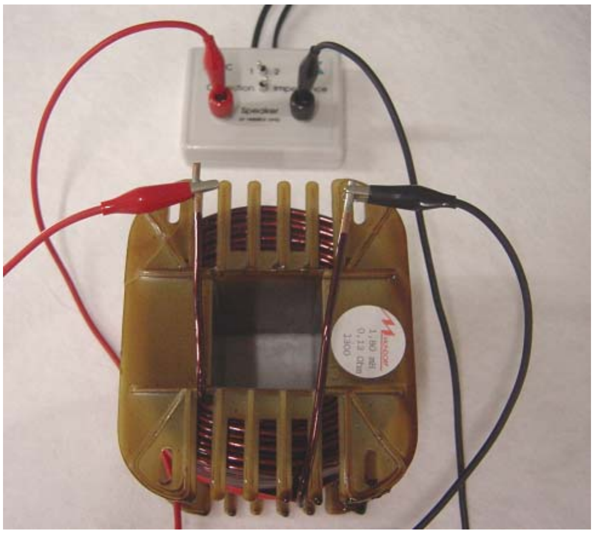
Coils behave exactly opposite to capacitors, even when the function of a coil is some what more intricate. A coil is made of a long isolated wound piece of wire. If a direct current is laid on it, it builds up a magnetic field just the same as an electromagnet.
If the voltage alternates or a music signal is connected the polarity of the magnetic field changes constantly. This change causes a voltage opposite that of the current voltage and is in effect a resistance to the signal. This effect grows with the frequency, so that the resistance of a coil is low at low frequencies and high at high frequencies. Because of this coils also can be used to separate high and low frequencies. The measurement unit is Henry [H] .
Coils for speakers are in the range mH [$1mH = 1x10^{-3}H$].
The bigger the value of the coil the sooner the resistance grows.
To illustrate this there are two sweeps in Figure 8.8. The red sweep shows the resistance of a coil with 2,2mH, the blue a coli with 1,8mH over a frequency range of 20Hz to 20kHz.
The coils are also connected with a 56Ω parallel.



red = coil 2,2 mH
blue = coil 1,8 mH

Figure 8.8

The frequency sweep shows that the resistance value of coils rise with the frequency. The vertical scale shows the resistance value and the horizontal scale shoes the frequency.



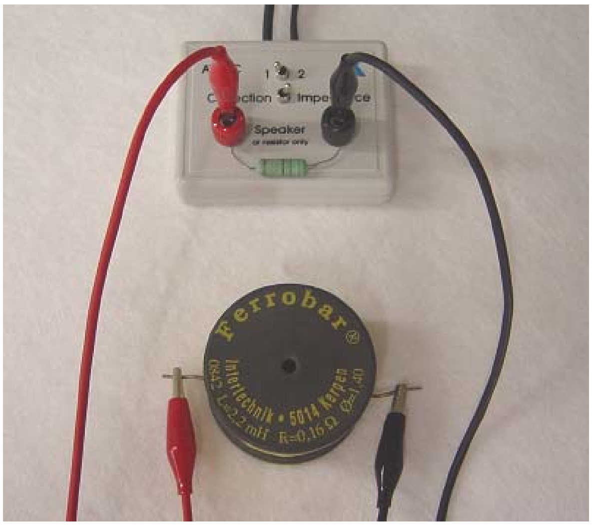


Figure 8.9 Figure 8.10

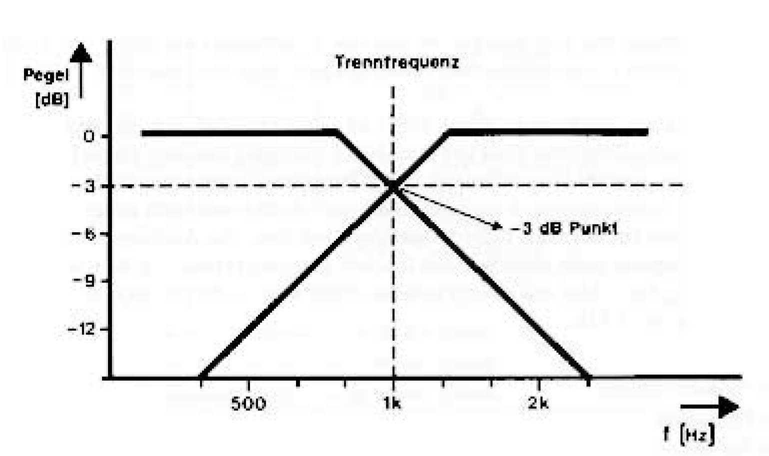
Figure 8.9 shows the measurement setup (Switch in impedance position) with 1.8mH
Figure 8.10 shows the measurement setup (Switch in impedance position) with 2.2mH

W8.2 Crossovers 1. and 2. degree

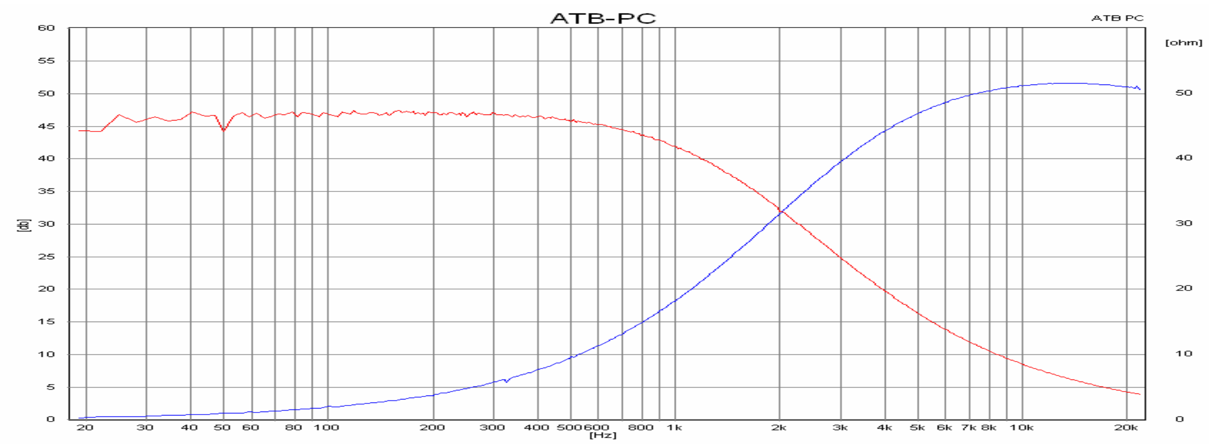
To keep things simple, in the following chapter we will only look into crossovers first (6dB) and second (12dB) degree according to Butterworth.

Crossovers 1. degree

A first degree crossover manages with only one element in each path.
It has the best phase response. Figure 8.13 on the following page shows the elements and calculation of the elements. The slope rising angle is 6dB pro Octave, meaning that the voltage on the speaker changes 6dB every double up of the frequency. The splitting or boundary frequency is there where the voltage has dropped by 3dB, that is to say 70.7% of the input signal. (figure 8.11). Figure 8.12 shows the crossover response of a 6dB high-low pass.



Figur 8.11



red = 6dB low pass
blue = 6dB high pass

Figure 8.12

The only part of the voltage that’s let through to the speaker, is that under the curve.

The area beneath the red curve (low pass) gets smaller as the frequency rises. This means the woofer receives less voltage with rising frequency there for getting quieter at higher frequencies.

The area beneath the blue curve (high pass) gets smaller as the frequency declines. This means that the tweeter receives less voltage as the frequency goes down, there for getting quieter at lower frequencies.

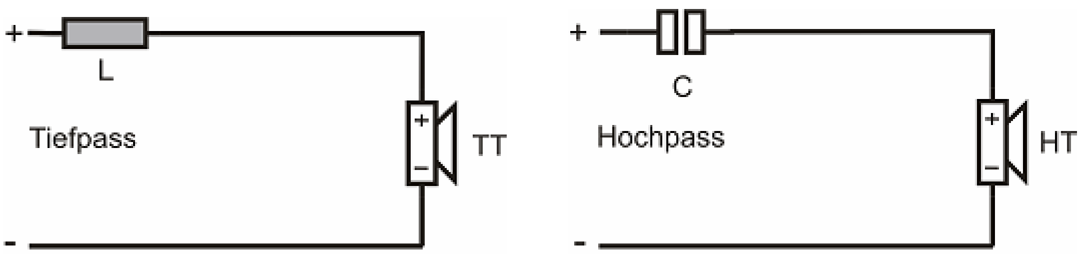


Figure 8.13

Calculation:

Needed values:

L = Inductivity (coil)
C = Capacity (condensor)
Z = Virtual resistance
π = Pi (3.14)
F_B = Boundary frequency

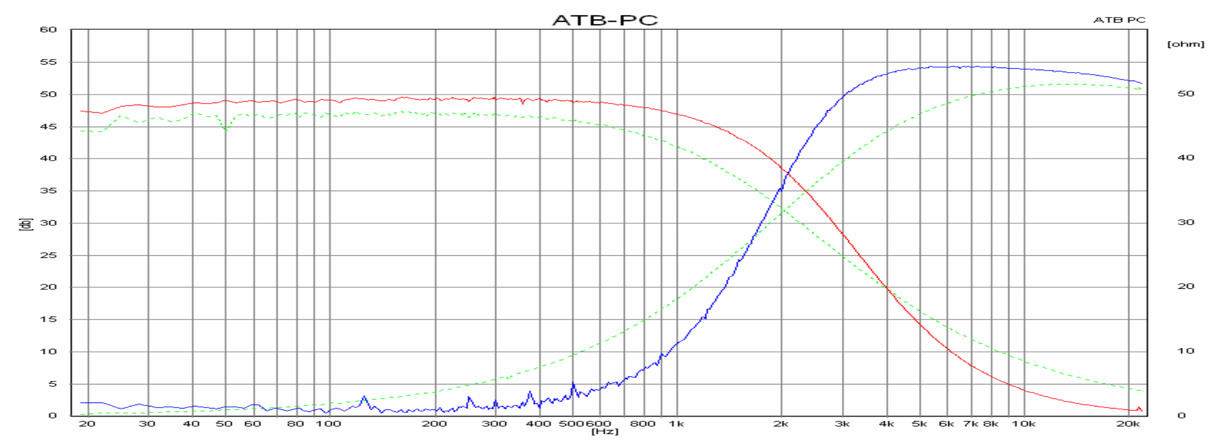
Formal: $$L = \frac{Z}{2 \cdot \pi \cdot f_G} \qquad C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_G}$$

In the appendix you will find a table with the element values for crossover frequencies' from 50Hz to 20kHz.

An example of calculation can be found in the chapter "The crossover of the nugget, stations of development".

Crossovers 2. degree

A crossover of the second degree needs 2 elements in each branch. The circuit and calculation are shown on the next page.
The sloping rate is 12dB per octave, meaning that a double up of frequency changes the voltage on the speaker by 12dB. The boundary frequency is that where the amplitude has dropped by 3dB. In other words 70.7% of the input signal.



red = 12dB low pass
blue = 12dB high pass
green = 6dB high und low pass

Figure 8.14

Figure 8.14 shows the crossover response of a 12dB high and low pass. Again the only part of the voltage that reaches the speaker is that under the curve similar to the function of a 6dB crossover, here in green for comparison.
As the curves show the decrease of voltage is with a 12dB crossover far steeper.

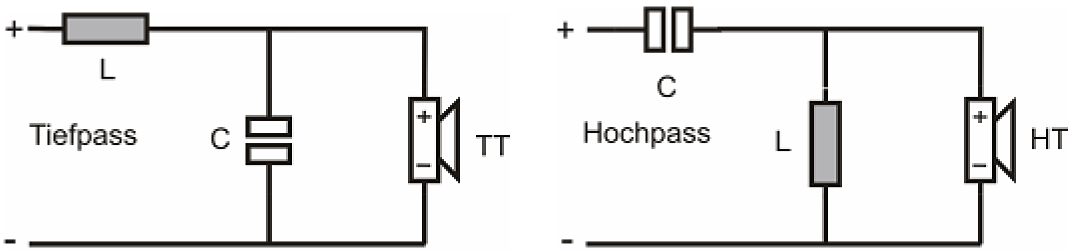


Figure 8.15

Calculation:
Needed values:

L = Inductivity (coil)
C = Capacity (condensor)
Z = Virtual resistance
π = Pi (3.14)
f_B = Boundary frequency

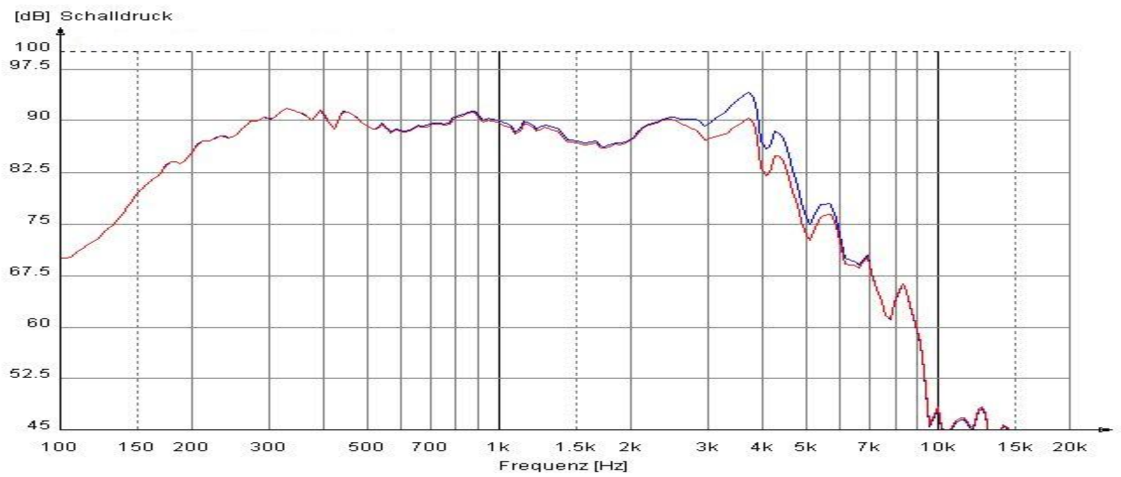
Formulas: $L = \frac{\sqrt{2} \cdot Z}{2 \cdot \pi \cdot f_G}$ $C = \frac{\sqrt{2}}{4 \cdot \pi \cdot f_G \cdot Z}$

In the appendix you will find a table with element values, for frequencies' from 50Hz to 20kHz.

Block circuit, Drain circuit and Inductance correction

Block circuit
The aim of these circuits is to flatten off amplitude rises, by cutting of certain areas of the frequency with the filter. This does though lead to a rise in the overall impedance of the circuit. In this application the rise and fall of the flanks are not as steep as a drain circuit.

Function:
The blocking circuit is made of a parallel circuit of coil, capacitor and resistor in row to the speaker. Figure 8.17 on the page over next.
In the chapter w8.2 the function of the crossover elements are explained. The condensor has high resistance for low frequencies, so that the current searches its way of the coil that has a low resistance for low frequencies. The same thing happens at high frequencies but vice versa. Now the coil has a high resistance and the current searches its way over the condensor that then has a low resistance. The area where the capacitor and coil have the same resistance is called resonance frequency. Here the resistance values cancel each other. The swinging circuit has in case of resonance a resistance going on to zero. Because of that a parallel resistor is necessary to limit the swinging circuit. With that you can manipulate the area between and set the insensitivity of the swinging circuit by choice of the resistor.



blue = original curve
violet = smoothed curve

Figure 8.16

Figure 8.16 shows how the blocking circuit works. The blue curve shows the original frequency sweep that has a rise at 3.7kHz. With use of the blocking circuit this rise is smoothed out, represented by the violet line.

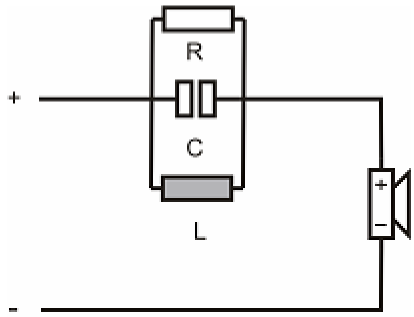


Figure 8.17

Calculation:
Needed values:

L = Inductivity (coil)
C = Capacity (condensor)
R = Resistor
Z = Virtual resistance of speaker (usually 8 or 4Ohm)
f_r = Resonance frequency (middle frequency area of spoken figure 8.16 approx. 3.7kHz)
f_1 = high boundary of rise
f_2 = lower boundary of rise
B = Bandwidth of rise (high frequency – low frequency of rise. Figure 8.16 approx 1KHz)

Formulas:

$$C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_2} \qquad L = \frac{Z}{2 \cdot \pi \cdot f_1} \qquad B = f_1 - f_2 \qquad R = \frac{1}{2 \cdot \pi \cdot C \cdot B} \qquad f_r = \frac{1}{2 \cdot \pi \cdot \sqrt{L \cdot C}}$$

The calculation usually only gets close, so that the values have to be determined from those values and corrected accordingly.

Drain circuits

Drain circuits are especially well suited to compensation of membrane resonances in the higher mid range. As they short circuit the electrical current to earth the elements used are not as critical as those of blocking circuits. In basic terms a draining circuit works the same way as a blocking circuit except that the flanks are steeper and they decrease the overall circuit impedance. Using a drain circuit can make the use of a impedance correction not necessary, as in the best case it does that job at the same time. All the same that doesn't always work satisfactorily in which case an impedance correction circuit can be connected parallel.
The drain circuit is made of a coil, condensor and resistor in a row connected parallel to the speaker figure 8.18 on the next page. The area where the coil has the same resistance value as the condensor is called resonance frequency.
Here the resistance values cancel each other and the resistance goes toward zero during resonance. This makes a resistor in row necessary to limit the swinging circuit.

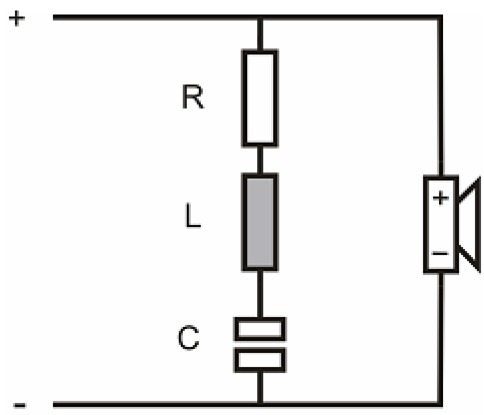


Figure 8.18

Calculation:
Needed values:

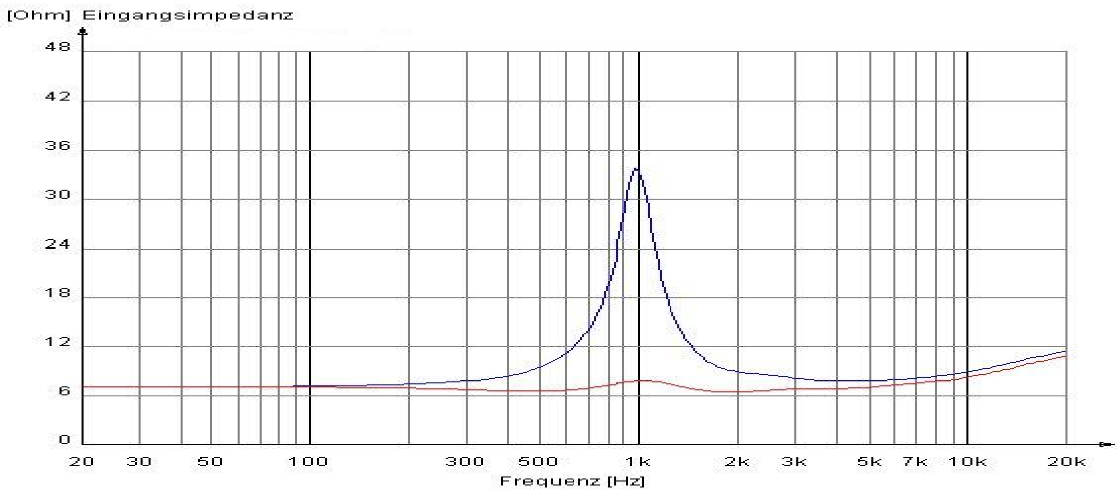
L = Inductivity (coil)
C = Capacity (condensor)
R = Resistor
Z = Virtual resistance of speaker (as a rule 8 or 4 Ohm)
f_r = Resonance frequency (mid frequency of rise)
f_1 = Higher frequency of rise
f_2 = Lower frequency of rise
B = Bandwidth of rise (high frequency – low frequency)

Formulas:

$$C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_2} \quad L = \frac{Z}{2 \cdot \pi \cdot f_1} \quad B = f_1 - f_2 \quad R = \frac{B}{2 \cdot \pi \cdot fr^2 \cdot C} \quad f_r = \frac{1}{2 \cdot \pi \cdot \sqrt{L \cdot C}}$$

The calculation usually gets only close, so that the calculated values need to be corrected.
Attention: The drain should as mentioned only compensate the membrane resonance in the higher mid range area. That is why it is hard to read the bandwidth B from the frequency sweep.
A drain circuit only works properly when the bandwidth used to calculate matches the bandwidth of the mechanical resonances. But this is if at all very difficult to access. The bandwidth is important for the calculation of the resistor, in practical terms this is then done by ear.
For a first try for the in row resistor R values of 4.7 Ohm for 8 Ohm and 2.2 Ohm for 4 Ohm speakers (13cm diameter) have proved to work. An example calculation can be found in *chapter W8.6 The nugget crossover (Development).*

Impedance correction
A crossover only works properly when the connection impedance of the crossover matches that of which it was calculated for. In the most cases that would be 4 or 8 Ohm. The impedance of a speaker though rises at higher frequencies' due to its voice coil inductivity. The impedance correction circuit compensates this rise and consists of condensor and resistor in a row connected parallel to the speaker. Figure and calculation on the next page(Figure 16.5).



blue = original impedance
violet = corrected impedance

Figure 8.19

Figure 8.19 shows the affect of the impedance correction. The blue curve shows the original impedance sweep, that has a rise at about 1kHz. Using the impedance correction the rise in amplitude is smoothed, shown by the violet curve.

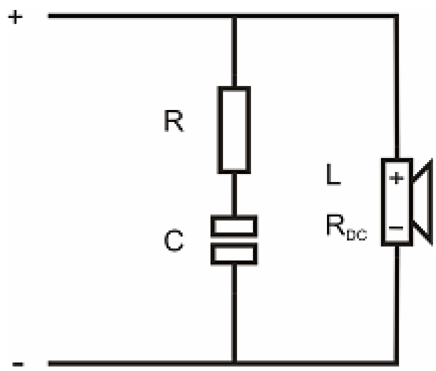


Figure 8.20

Calculation:
Needed values:

L = Speaker inductivity (manufacturer details)
C = Capacity (Condensor)
R_{DC} = Direct current resistance of speaker (manufacturer details)
R = Resistance

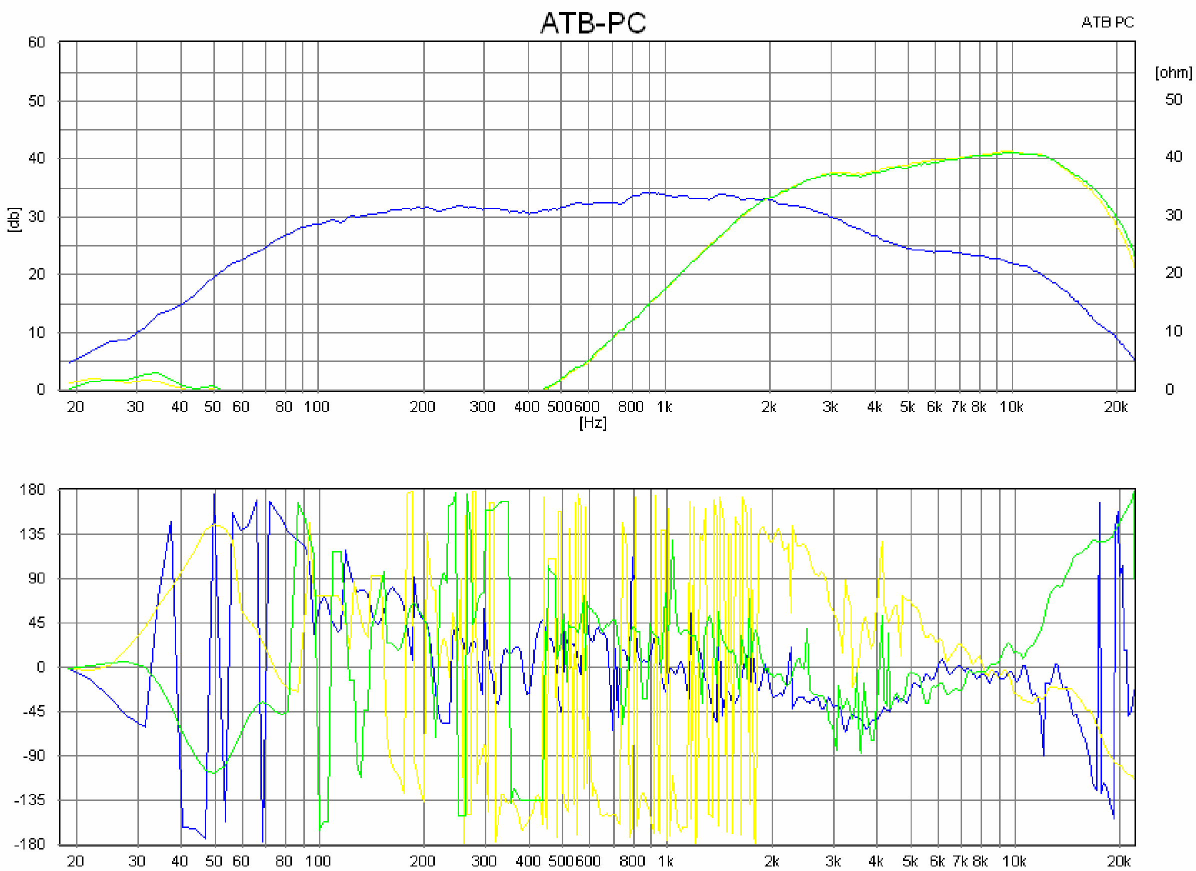
Formulas: $R = R_{DC} + R_{DC} \cdot 0{,}5$ $\quad C = \frac{L}{R^2}$

The calculation usually is only close to and has to be corrected in the practise.

W8.3 Running time, acoustic phase

A loudspeaker is a swinging system, as is consists of springs and mass. This system is permanently accelerated and braked down. As a woofer has a different mass than a tweeter, the running times are also different. This shows up in the fact that the sound transmitted from the speakers do not reach the human ear simultaneously. This leads to an unclean sound and in the worst case even to frequency deletion. In other words the speaker does not run in phase to each other.
The phase can be improved immensely by polling the tweeter around or by regarding the phase position when constructing the housing.

Figure 8.21 shows the frequency sweep and phase position of the woofer and tweeter in the Nugget. For illustration the tweeter is shown correctly and incorrectly poled.



blue = Woofer
yellow = Tweeter correctly poled
green = Tweeter incorrectly poled

Figure 8.21

It is clearly recognisable that the tweeter (yellow curve in the lower diagram) is not in phase with the woofer (blue curve in the lower diagram).
Important: The measurement system can only determine the phase by a measurable signal. If no signal is measured, then there is no defined phase so that the displayed values are useless. As an example the yellow curve in the lower diagram in the area from 0Hz to 2kHz, can be seen as such.

Through poling the tweeter around better results are achieved (green curve in the lower diagram). Another important point of knowledge can be read from the diagram too, the fact that the frequency sweeps of the tweeter doesn't change by poling around. This means that a linear frequency curve doesn't guarantee good sound quality. Then only speaker combination that play in phase to one another offer a maximum of clarity, reality and transparency. This shows in the reproduction of the tiniest nuances. As the radiated frequency spectrum from high and low speaker are heard at the same time.

W8.4 Electrical phase

Where as the acoustical phase refers to the pure running time of the sent tones and measured by a microphone, the electrical phase is a pure electrically measurable value.
Frequency dependant elements have a frequency dependant resistance, the impedance. The impedance consists of amplitude and phase.
Or more simply, when connected to a voltage only with a resistor does the current flow simultaneously. With condensers and coils there is a time difference.

Figure 8.22 shows the impedance and phase change of resistor condensor and coil.

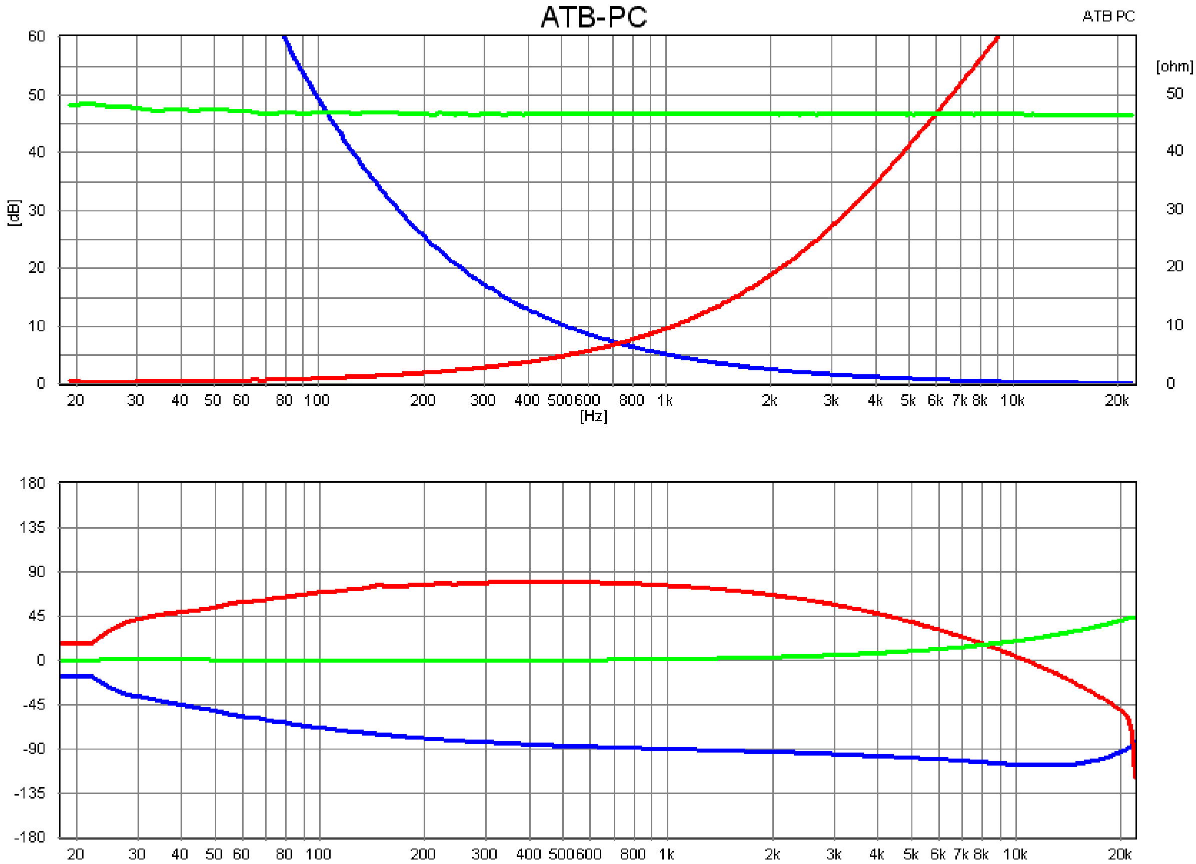
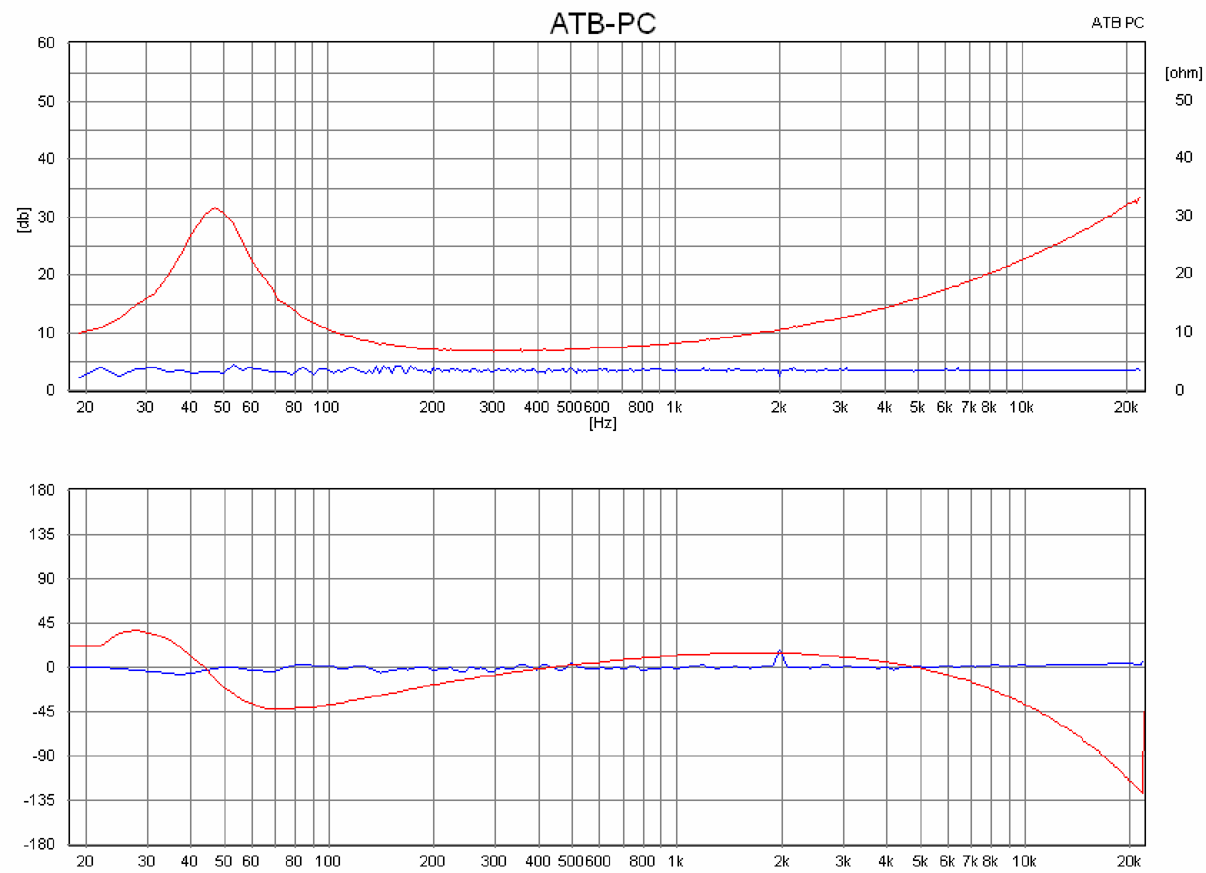


Figure 8.22

green =Resistor
red =Coil
blue =Capacitor

It is to be seen that the resistor as explained in chapter W8.2 , is frequency independent. So there is no time delay between voltage and current. The Figure 8.22 shows the impedance measurement of the frequency dependant resistance of resistor condensor and coil. During measurement the voltage on the elements are measured. There for the phase angle relates to the voltage. The red coil frequency sweep of the coil shows the near to school book +90°(degree) phase displacement. The voltage precedes the current. This can also be looked at as if the current recedes the voltage. The elapse of the blue condensor frequency sweep shows a -90° (degree) phase displacement, but in this case the voltage recedes and the current precedes.
In this measurement only the middle frequency range is shown correctly. For the high and low frequencies' is either because of the low resistance of the measured elements the measured voltage too low, or the resistance of the elements are too high, so that the measurement range is exceeded.



red = Woofer
blue = Tweeter

Figure 8.23
Figure 8.23 shows the impedance and phase elapse of high and low speaker. As the blue frequency sweep shows, the tweeter behaves very frequency independent that is although not always so. The red frequency sweep shows the impedance and phase sweep you would expect from a woofer. Where the impedance peak is to be seen in the higher diagram, a phase step can be found in the lower diagram, this is the resonance frequency of the woofer at about 45 Hz.
To compensate the impedance peak at app. 2kHz a correction as described in chapter W8.2.can be used.

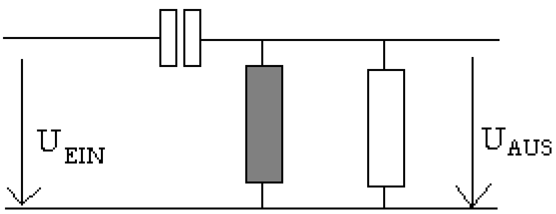
W8.5 Dynamic-Measurement

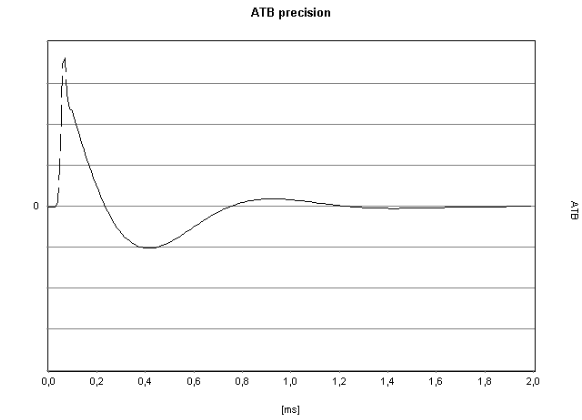
In the preadent chapters the frequency and phase sweeps were looked into.
To show the time response behaviour of a transmission path the Dynamic-Measurement Program was developed for science. The program runs under the ATB precision measurement system. The sound cards do not have the time resolution for the measurement. The behaviour across time is best shown on hand of the impulse response. The impulse response shows the amplitude and phase clearly. In the normal impulse response there are no frequencies' to be seen. That's why the impulse response is shown in the Dynamic-Measurement Program in a 3D chart with frequencies'. The chart consists of a y axis for the amplitude, a x axis for the frequency and a y axis for the time.
More information at:

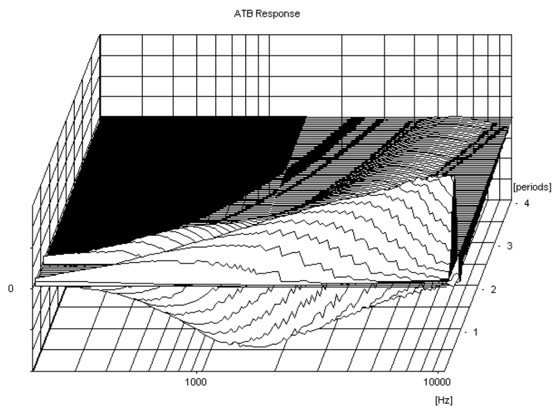
www.dynamic-measurement.de

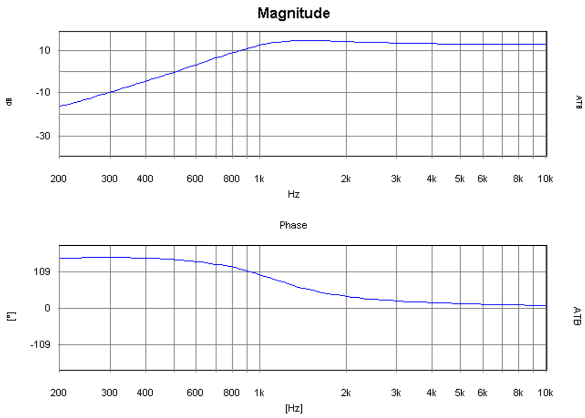
Using the program frequency, phase, impulse response and Dynamic-Measurement illustration of the forgoing circuit are shown.

12 dB / Oktave hightpass

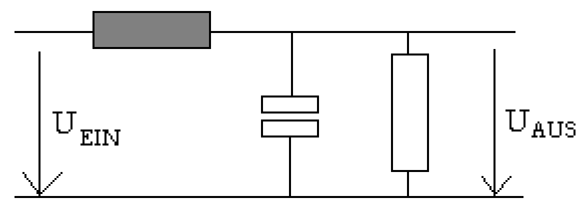


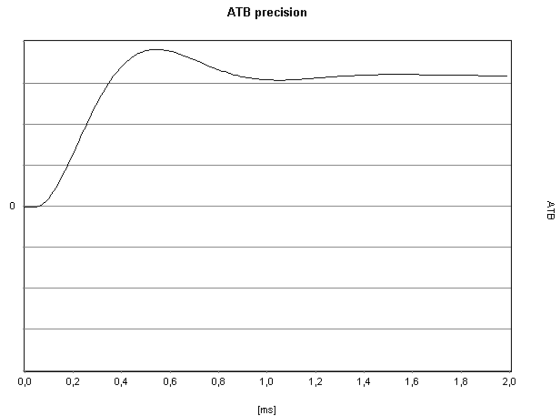


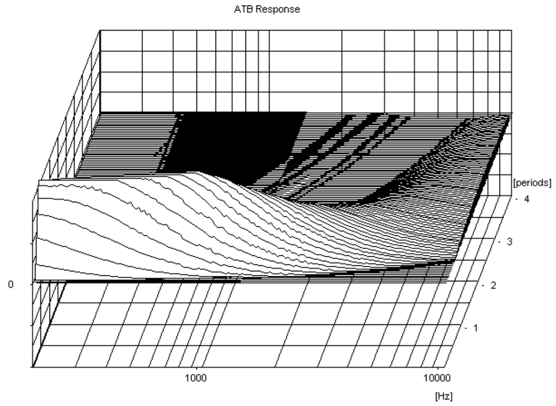


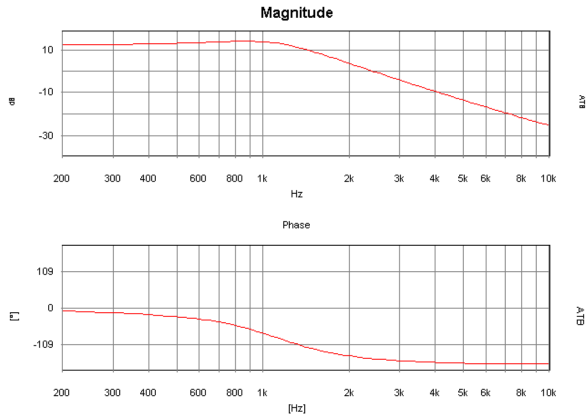


12 dB / Oktave lowpass

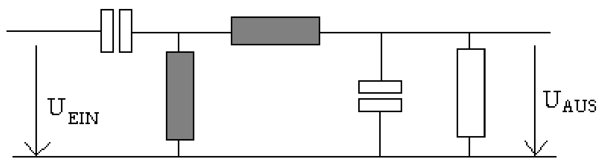


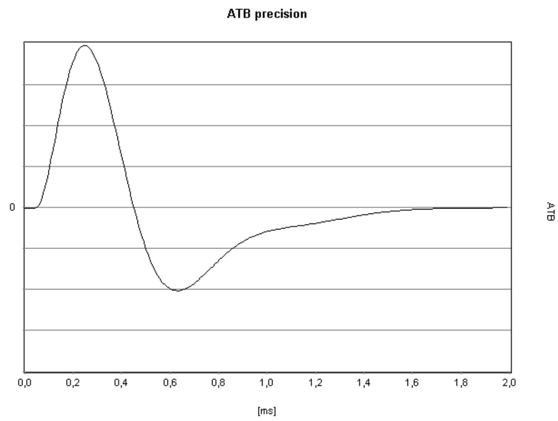


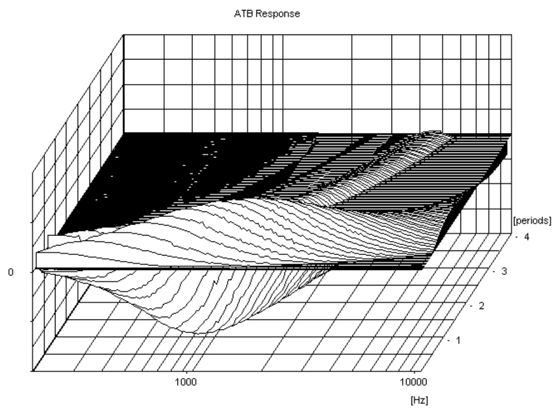


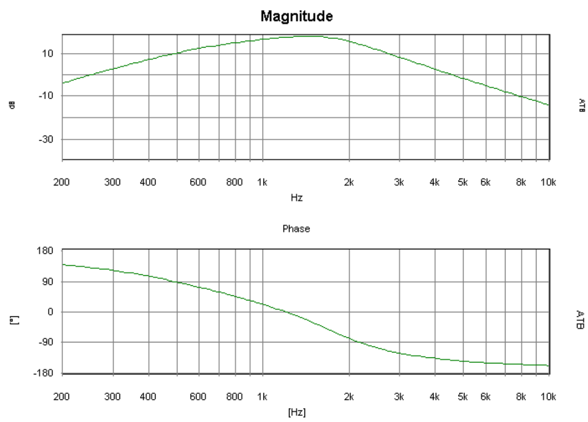


12dB / Oktave Bandpass









The Dynamic-Measurement Program is of great importance also for acoustic measurements, especially when developing time corrected loudspeakers.

W8.6 The Nugget crossover (Development)

The speaker was built according to the enclosure calculations, the speakers mounted with long connection wires. The aim is the construction of a 1 degree crossover, to achieve the best phase response. The position of the tweeter in the casing was determined before hand using pink noise.
All measurements were carried out using 2,83 Volt (1 Watt), the distance to the microphone was 1 meter figure 8.24. The absolute dB value is not correct as the microphone sensitivity was not adjusted.
As measurement location an old garage was chosen, just to demonstrate that no particular measurement environment is needed using the ATB PC Pro. The measurement setup is shown in figure 8.25.

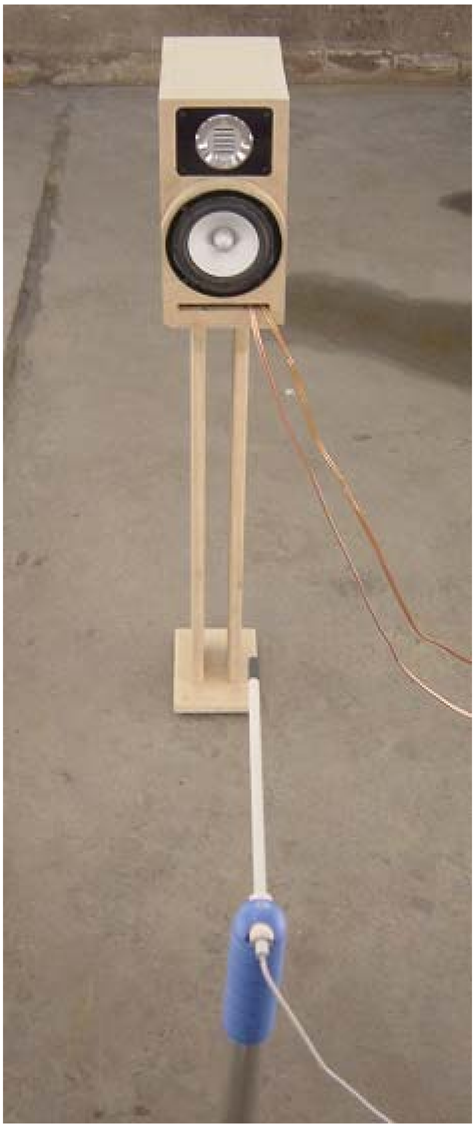
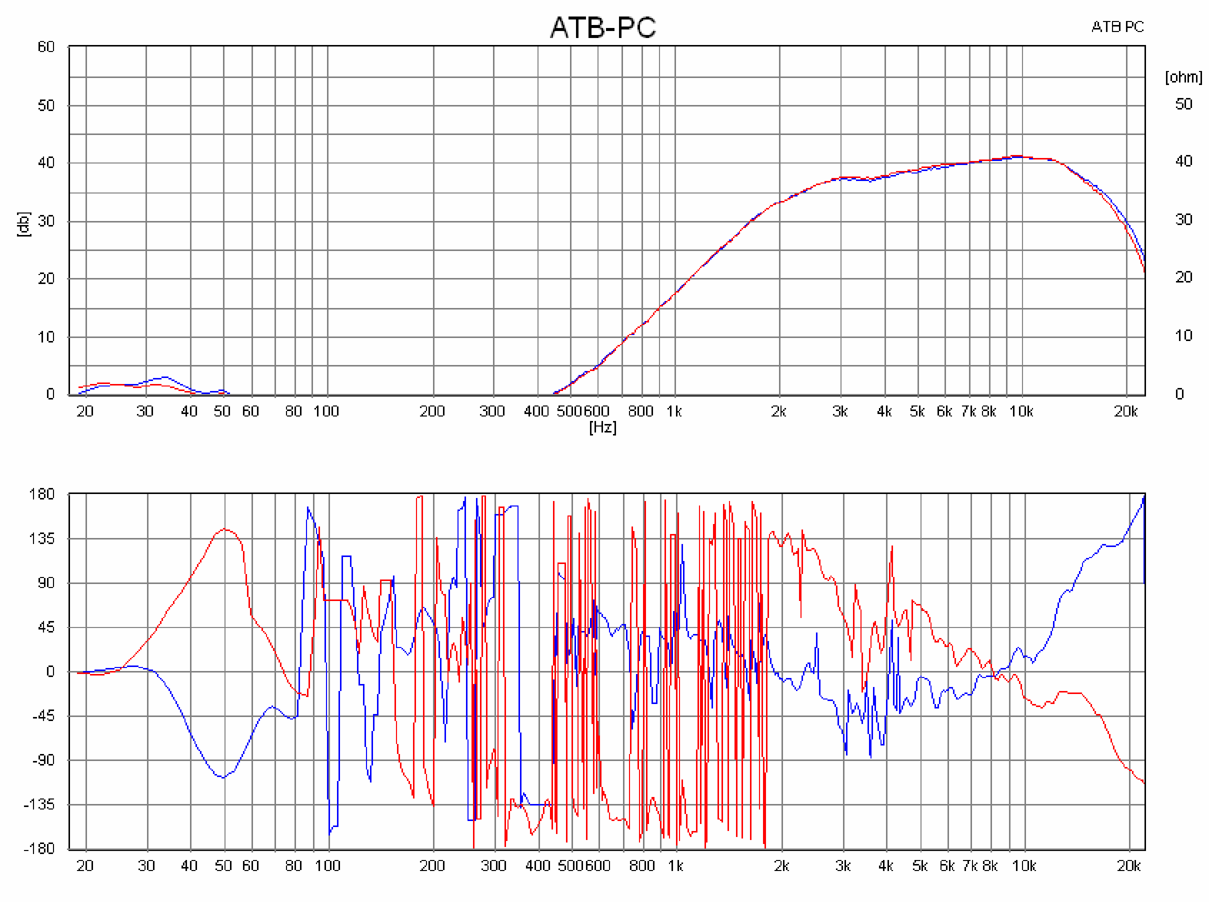


Figure 8.24

Figure 8.25

Figure 8.26 shows the frequency sweep and phase of the tweeter. The speaker was connected with out crossover elements.

attention: When reproducing the measurement protect the tweeter by starting with the lowest possible volume!!!



red = correctly poled
blue = incorrectly poled

Figure 8.26

Result: This measurement shows the phase difference of 180° due to false poling. From 5kHz the diversion, on the blue curve, is caused by the insufficient time resolution of the sound card. The steps below 500Hz come to be by the too low amplitude of the speaker. The steps on the red curve below 2kHz are caused by the phase angle around 180°.

Figure 8.27 shows the frequency and phase sweep of the woofer. The speaker was connected without crossover elements.

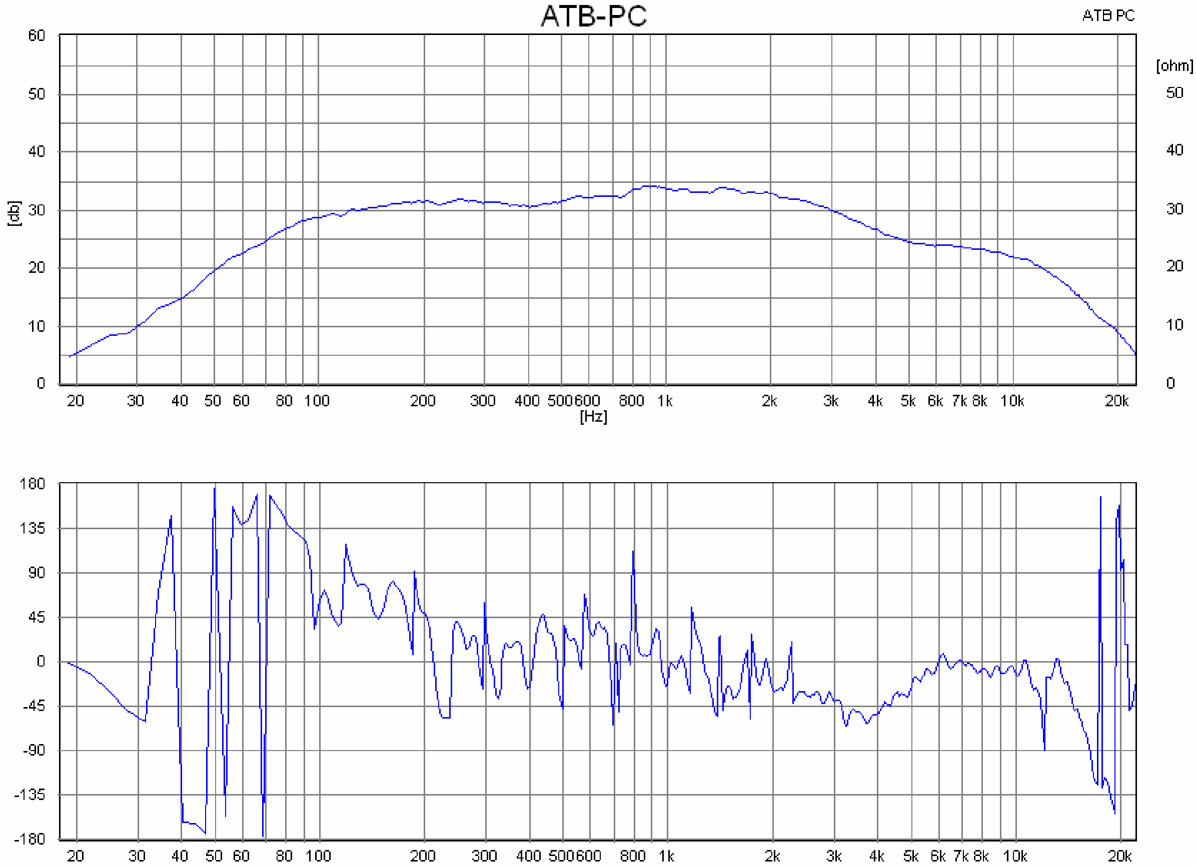


Figure 8.27

Result: This measurement shows that the woofer has a very linear phase response and is used correctly poled.

Figure 8.28 shows the frequency sweep and phase of the high low speaker, both speakers are connected parallel and without crossover elements.

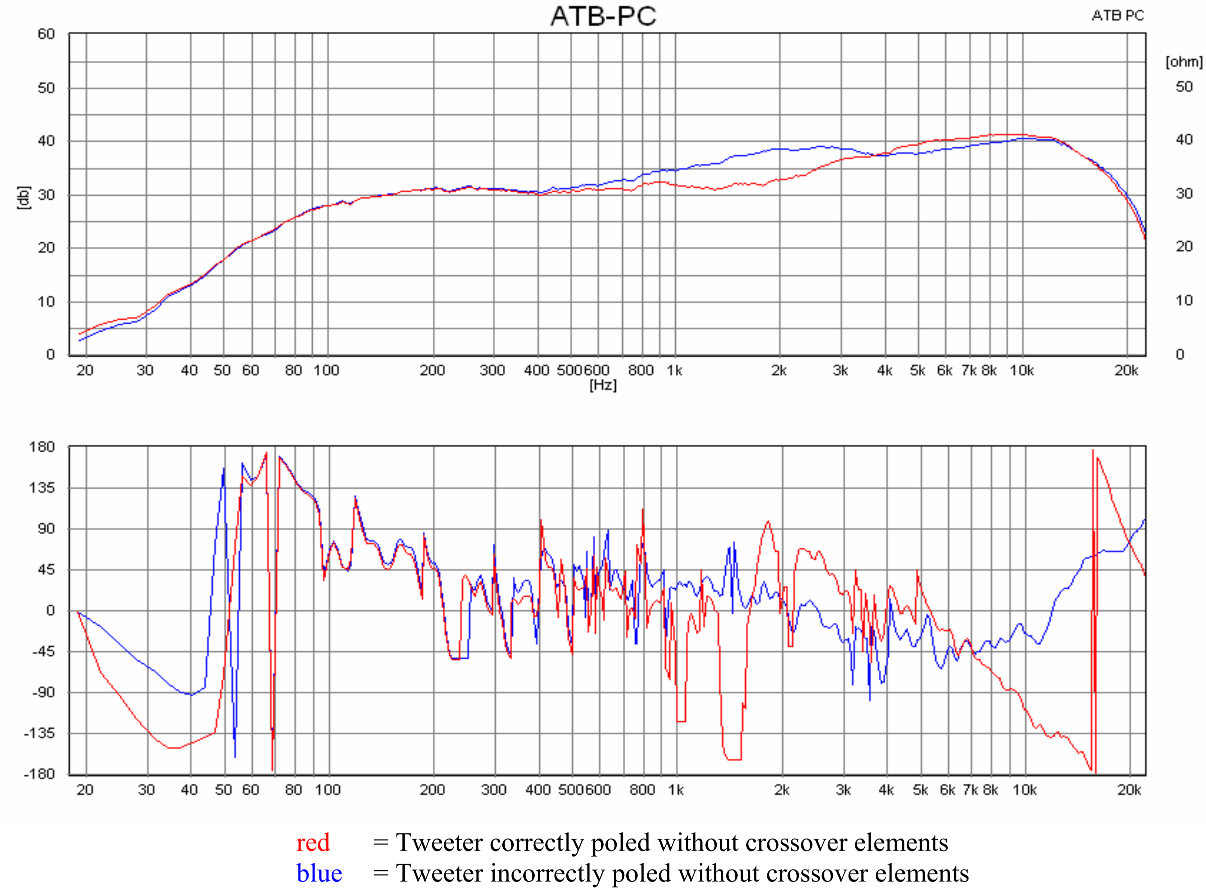
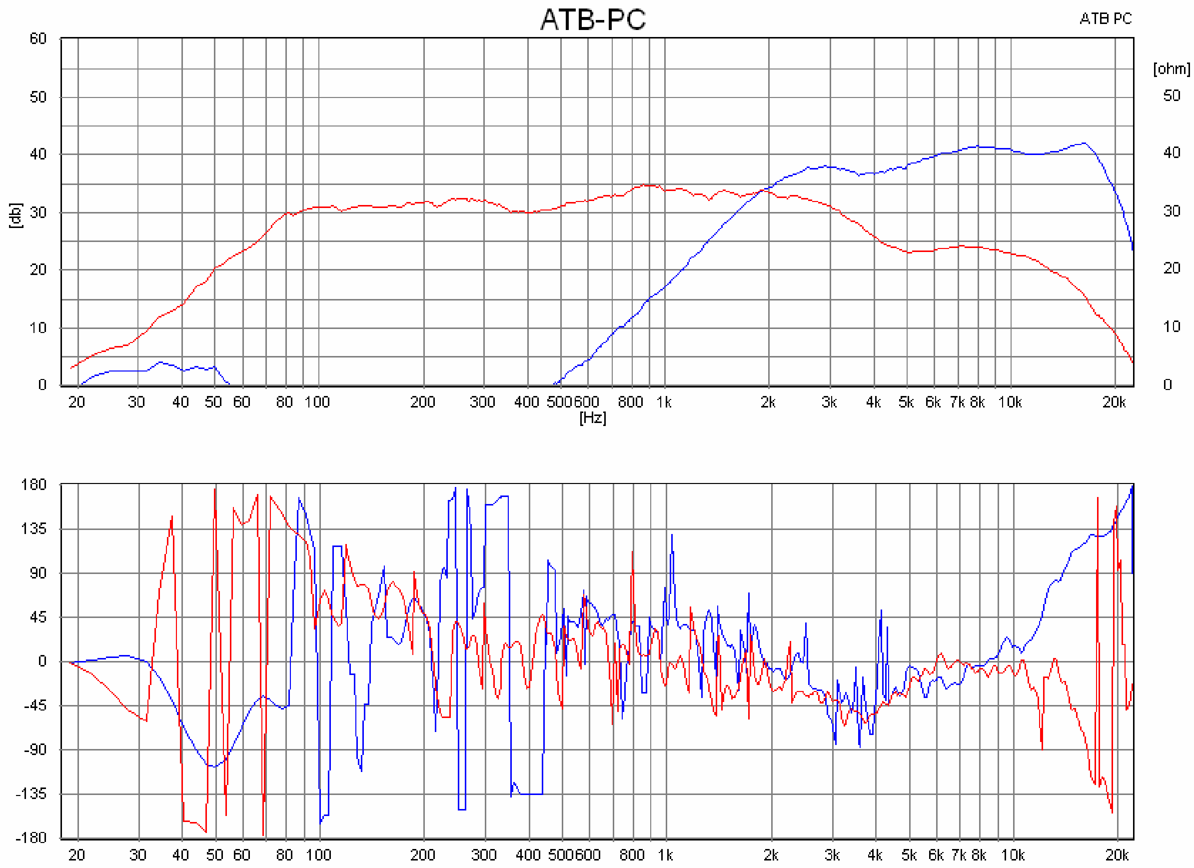


Figure 8.28

Result: The correct poling of the correct poling of the speaker is found. The woofer is correctly poled, the tweeter incorrectly. The frequency sweep shows this, but the phase step does so more clearly.

Figure 8.29 shows overlapping the single frequency sweeps of the high and low speaker.



red = Woofer without crossover elements
blue = Tweeter without crossover elements

Figure 8.29

Result: The frequency sweep shows that the tweeter crossover frequency should be about 2.3kHz, for the woofer a frequency of about 2.2 seems to be reasonable. As the crossover voltages add up, here can and should be let a gap. In this case the gap is about 100Hz. If that is enough will show up as we go on.

Calculation

From the speakers following values are known:

	Tweeter	Woofer
f_r	= 2kHz	= 45Hz
R_{DC}	= 4,5Ω	= 6,4Ω
Z	= 6Ω	= 8Ω

Low pass for the woofer:

Coil: $$L = \frac{Z}{2 \cdot \pi \cdot f_G} = \frac{8}{2 \cdot \pi \cdot 2200\,Hz} = 0{,}58\,mH$$ Chosen 0,56mH.

High pass for the tweeter:
Important: In the blue curve figure 8.29 you can see, that the tweeter is on average 4db louder than the woofer, this is compensated with a resistor. The preresistor has to be determined before calculating the condensor. Its value is added to the power of 2 to the to the impedance of the tweeter to get the correct results. This is important as the exact impedance of the tweeter branch is needed calculating the capasitor.
Determining the preresistor:
As the impedance Z consists of the addition of the direct current resistance R_{DC} to the power of 2 and the blind resistance X_L of the tweeter, the value of X_L must be calculated, as it is not given.

Known is though the DC resistance R_{DC} and the overall impedance Z, with that X_L can be calculated as follows:

$$X_L = \sqrt{Z^2 - R_{DC}^2} = \sqrt{6\Omega^2 - 4{,}5\Omega^2} = 3{,}97\Omega$$

Now the relationship of the suppression is needed, to determine the preresistor. For this the logarithmic value of 4dB has to be transformed into a value basis 10:

$$V_{UD-} = 10^{\left(V_{ULog}/20\right)} = 10^{\left(\frac{4}{20}\right)} = 1{,}585$$

The calculated relationship of 1.585 is to be put into relationship to the preresistor and R_{DC} of the tweeter:

$$R_V = \frac{R_{DC}}{V_{UD-}} = \frac{4{,}5\Omega}{1{,}585} = 2{,}84\Omega$$ Chosen 2,8Ω out of 2x parallel 5,6Ω resistors.

The 2.8 Ohm is now brought together with X_L and R_{DC} a new Z :

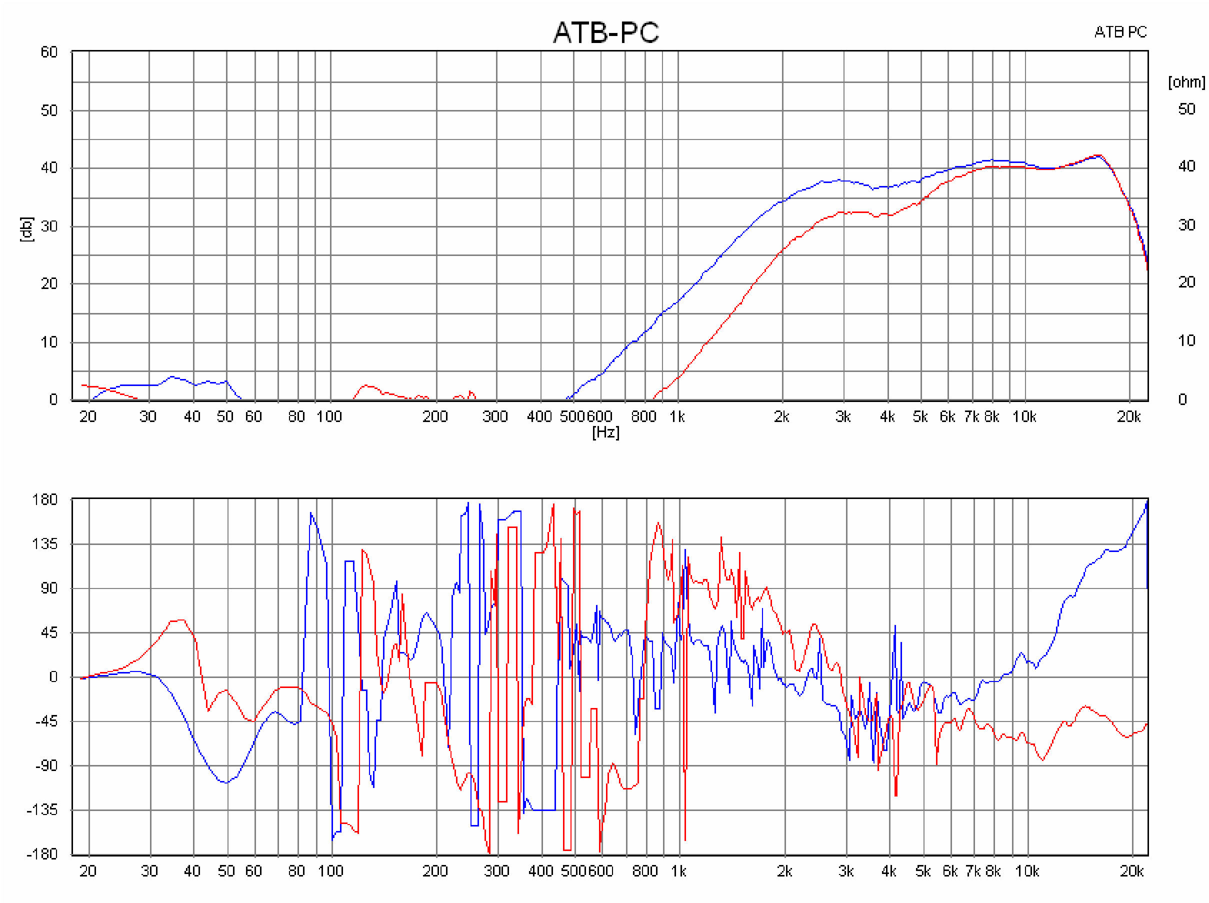
$$Z = \sqrt{(R_{DC} + R_V)^2 + X_L^2} = \sqrt{(4{,}5\Omega + 2{,}8\Omega)^2 + 3{,}97\Omega^2} = 8{,}3\Omega$$

It would have been possible, to use a voltage splitter, making the more complicated calculation redundant. All the same this way we have raised the overall resistance of the circuit to 8 Ohm, which makes sense with a tweeter of 6 Ohm impedance.
With the new calculated Z we can calculate the condensor:

$$C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_G} = \frac{1}{2 \cdot \pi \cdot 8{,}3\Omega \cdot 2300\,Hz} = 8{,}3\mu F$$

We chose 8,2µF.

The calculation is at good close, however should it be, as things go on that the values do not fit properly, they can still be changed.
Figure 8.30 shows the influence of the capacitor on the tweeter.

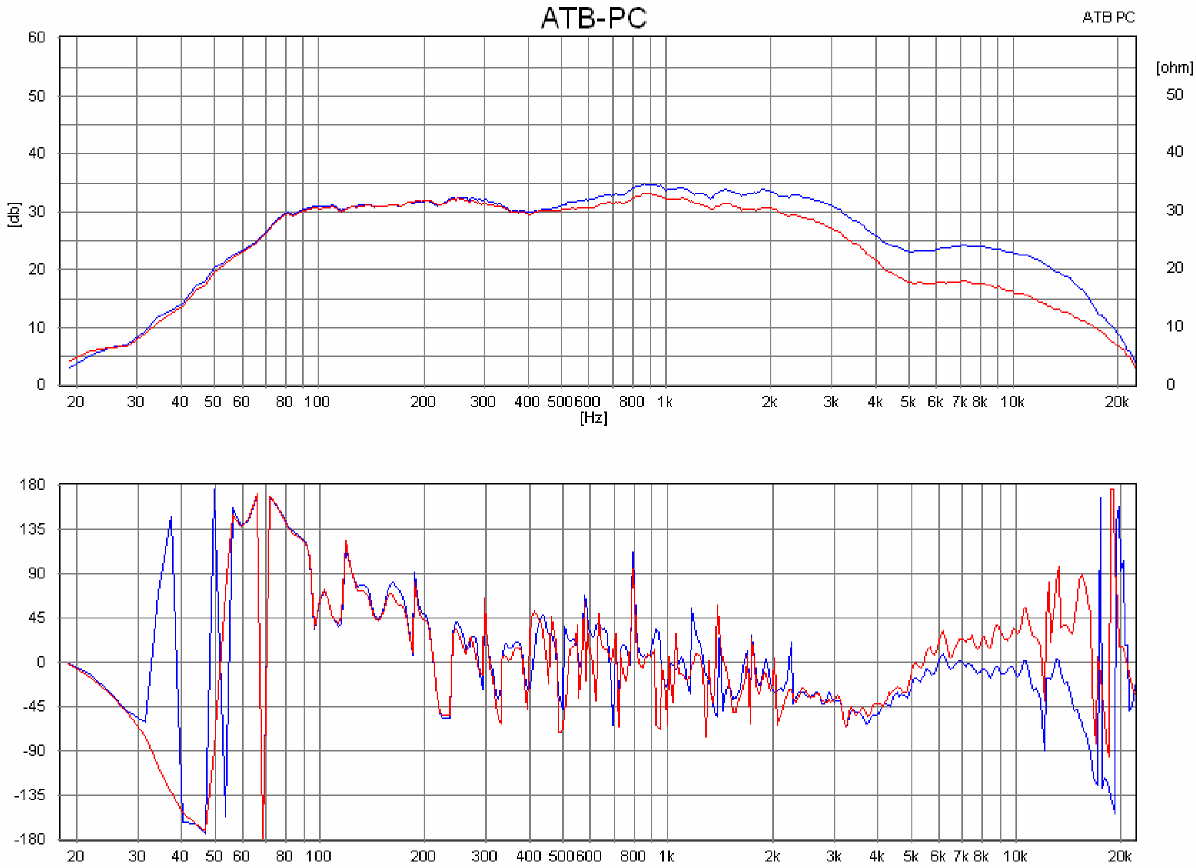


red =Tweeter with condensor
blue = Tweeter without crossover elements

Figure 8.30

Result: The top diagram shows that the lower frequency spectrum is suppressed by the condensor. The lower diagram shows the influence of the condensor on the phase. At about 900Hz the condensor starts to take influence and turns the phase at the start up to nearly 90°.

Figure 8.31 shows the influence of the coil on the woofer.



red = Woofer with coil
blue = Woofer without crossover elements

Figure 8.31

Result: The upper diagram shows that the coil suppresses the high frequency spectrum. The lower diagram shows the influence of the coil on the phase. Here the phase is influenced so much as the speaker itself has a coil that has already turned the phase. A coil in row to a coil doesn't change the phase noticeably further.

Figure 8.32 shows the frequency sweep and phase of the speaker with the crossover developed so far. The relating circuit diagram and measurement setup figure 8.34 are shown on the next page.

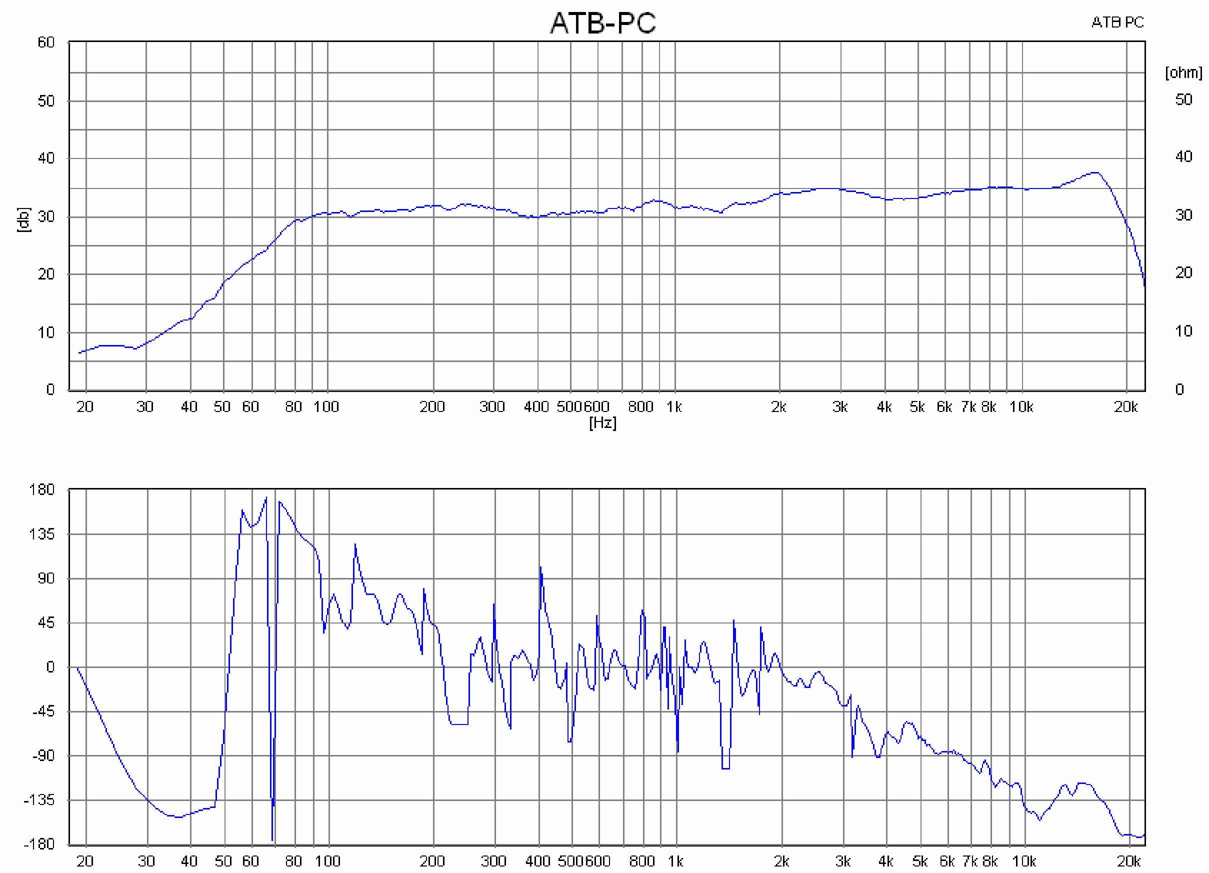
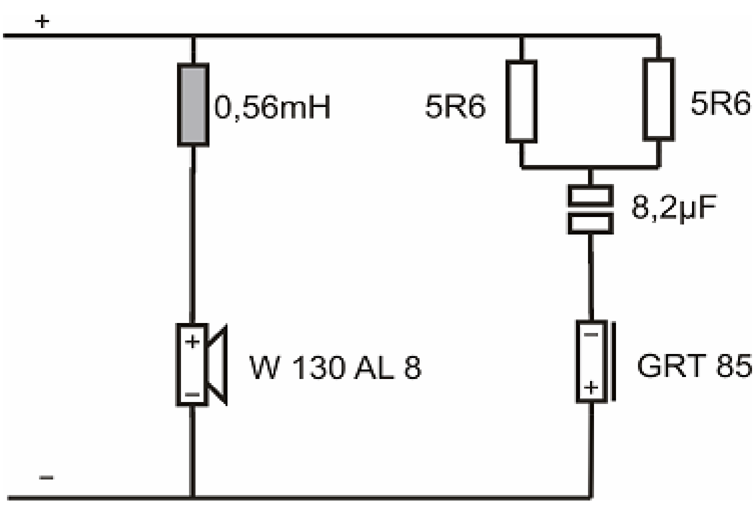


Figure 8.32

Result: The upper diagram shows an already very good frequency sweep, the phase in the lower diagram is also well done. The hearing test enclosed though that there were distortions in the mid frequency range that would not fit into the overall picture.



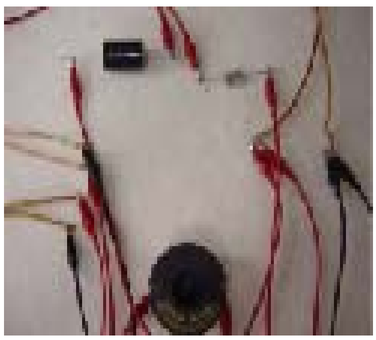
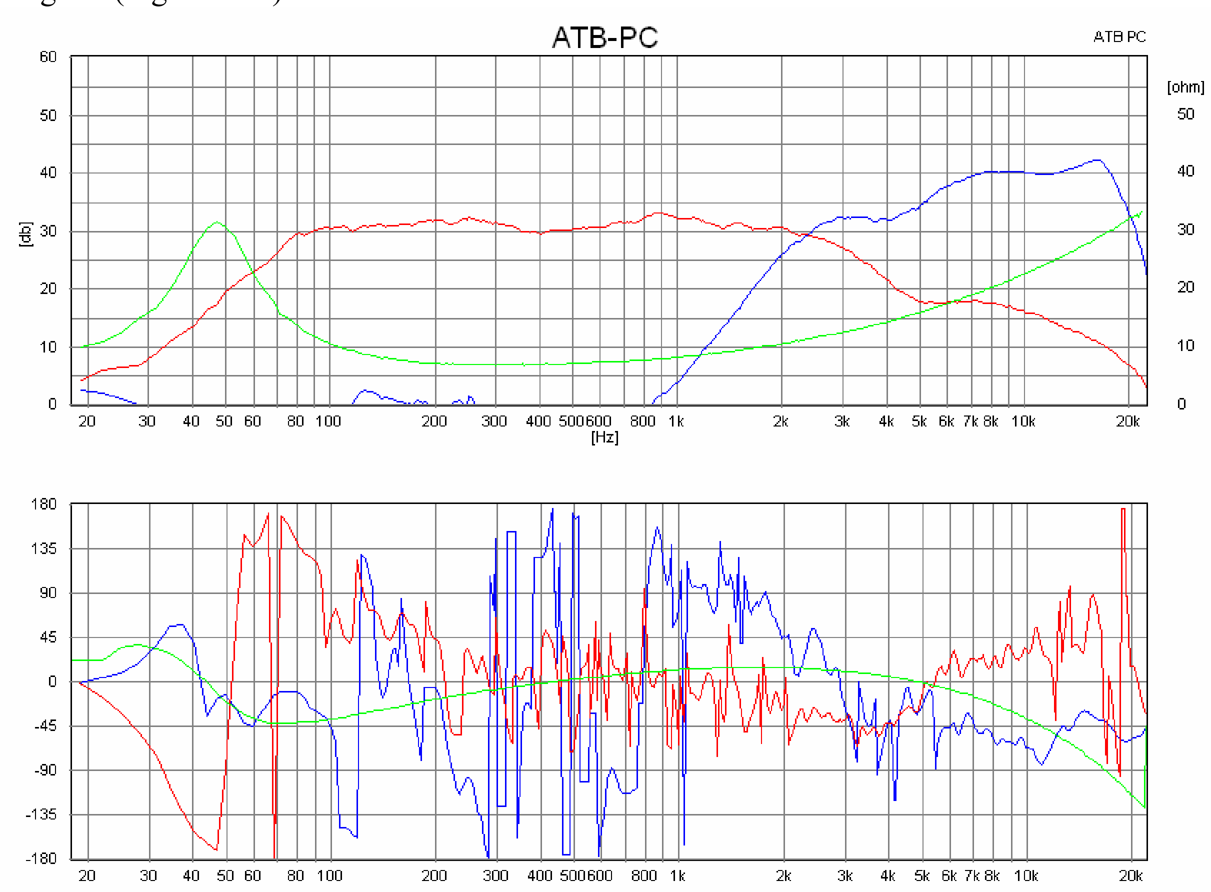


Figure 8.33

Figure 8.34

To find out where the unwanted distortions in the mid range were coming from, the single measurements of the tweeter and woofer with coil were laid over each other in a single diagram (Figure 8.35).



red = Woofer with coil
blue = Tweeter with condensor
green =Impedance of woofer

Figure 8.35

Result: The left hump of the tweeter (blue curve in the upper diagram) at range from app. 1,95kHz until 3,5kHz disturbs the overall sound reproduction. The peak at about 5kHz gets suppressed through the preresistor.
The woofer also has a hump with midpoint at about 6.2 kHz. The green impedance curve shows here indecently the mid value of its impedance maximum. To smooth out these bumps, two drain circuits are used.

Calculation of the tweeter drains circuit:
Condensor:

$$C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_2} = \frac{1}{2 \cdot \pi \cdot 6\Omega \cdot 1950\,Hz} = 13{,}6\mu F$$

Chosen 13.3µF out of a 10µF and a 3.3µF parallel.

The coil:

$$L = \frac{Z}{2 \cdot \pi \cdot f_1} = \frac{6\Omega}{2 \cdot \pi \cdot 3500\,Hz} = 0{,}27mH$$

Chosen 0.27mH.

The resistor R is not necessary here as the resistance of the elements are sufficient and the pre circuited condensor in connection with the preresistor have a limiting effect.

Calculation of the woofer drains circuit:
For the calculation of the drain circuit the boundaries of the impedance peak f_1 = 20kHz und f_2 = 1,9kHz are used.
Condensor:

$$C = \frac{1}{2 \cdot \pi \cdot Z \cdot f_2} = \frac{1}{2 \cdot \pi \cdot 8\Omega \cdot 20000\,Hz} = 1\mu F$$

Chosen 1µF.

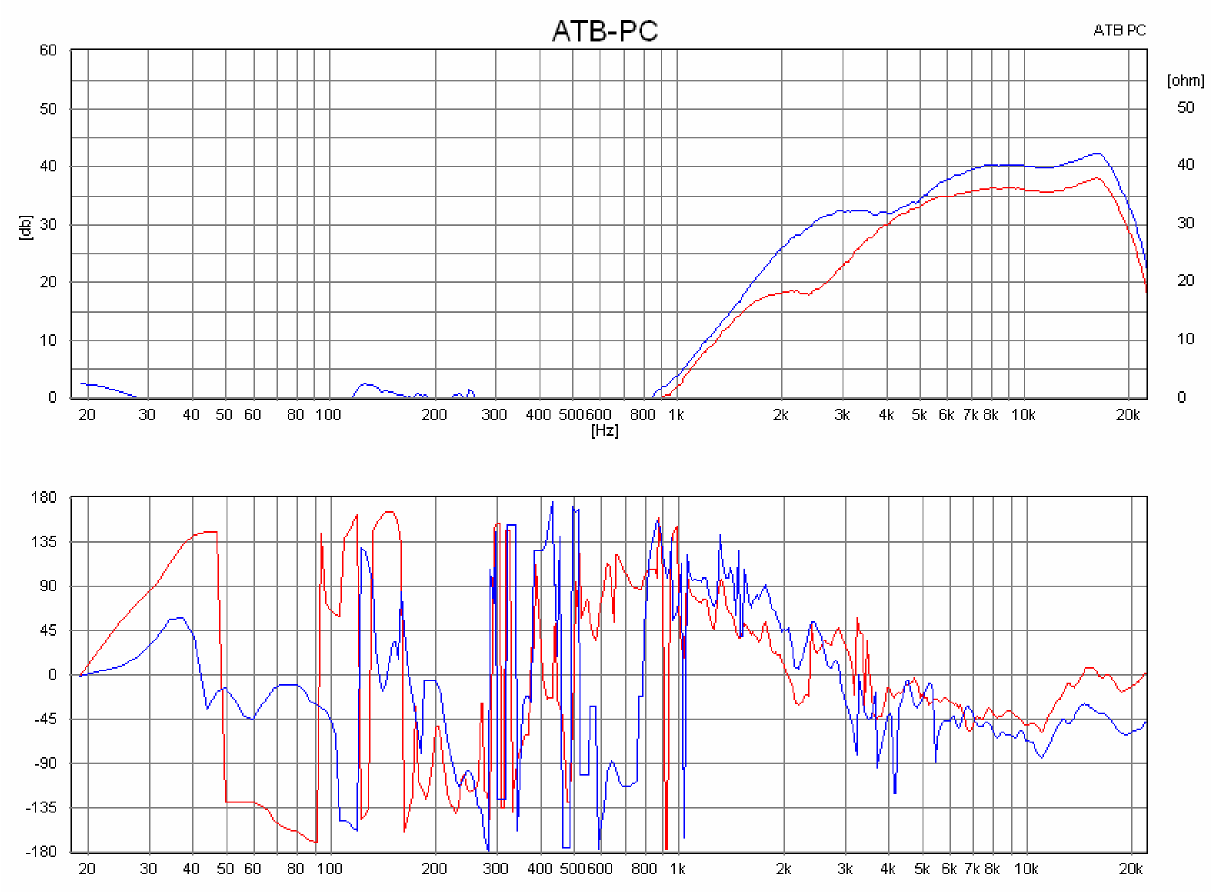
Spool:

$$L = \frac{Z}{2 \cdot \pi \cdot f_1} = \frac{8\Omega}{2 \cdot \pi \cdot 1900\,Hz} = 0{,}67\,mH$$

Chosen 0.68mH.

For the resistor the practice value of 4.7Ω was chosen.

Figure 8.36 shows the influence of the drain circuit on the frequency spectrum and phase of the tweeter. The relating circuit diagram (Figure 8.37) and the measurement setup (Figure 8.38) are shown on the next page.



red =Tweeter with drain circuit, condensor and preresistor
blue = Tweeter with capacitor

Figure 8.36

Result: the upper diagram shows, the influence of the modification has pushed the bump between 1.95kHz and 3.5kHz down lower, i.e. its was drained off. The bump at app. 2kHz is so weak in intensity, that it has no disturbing influence on the sound impression.
Further more the phase has changed positively and is now close to the 0° line.

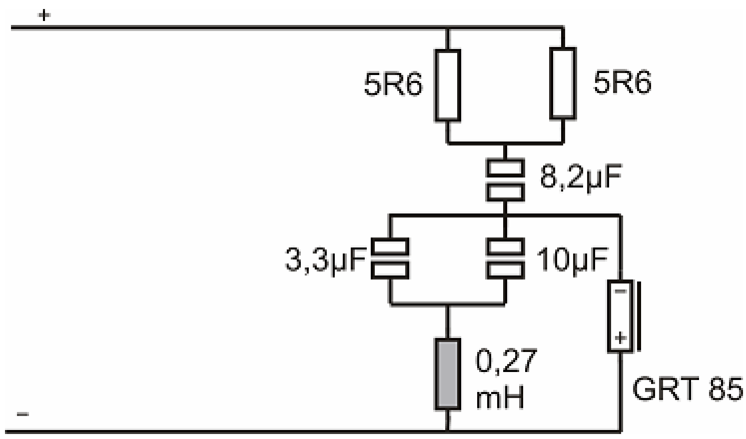
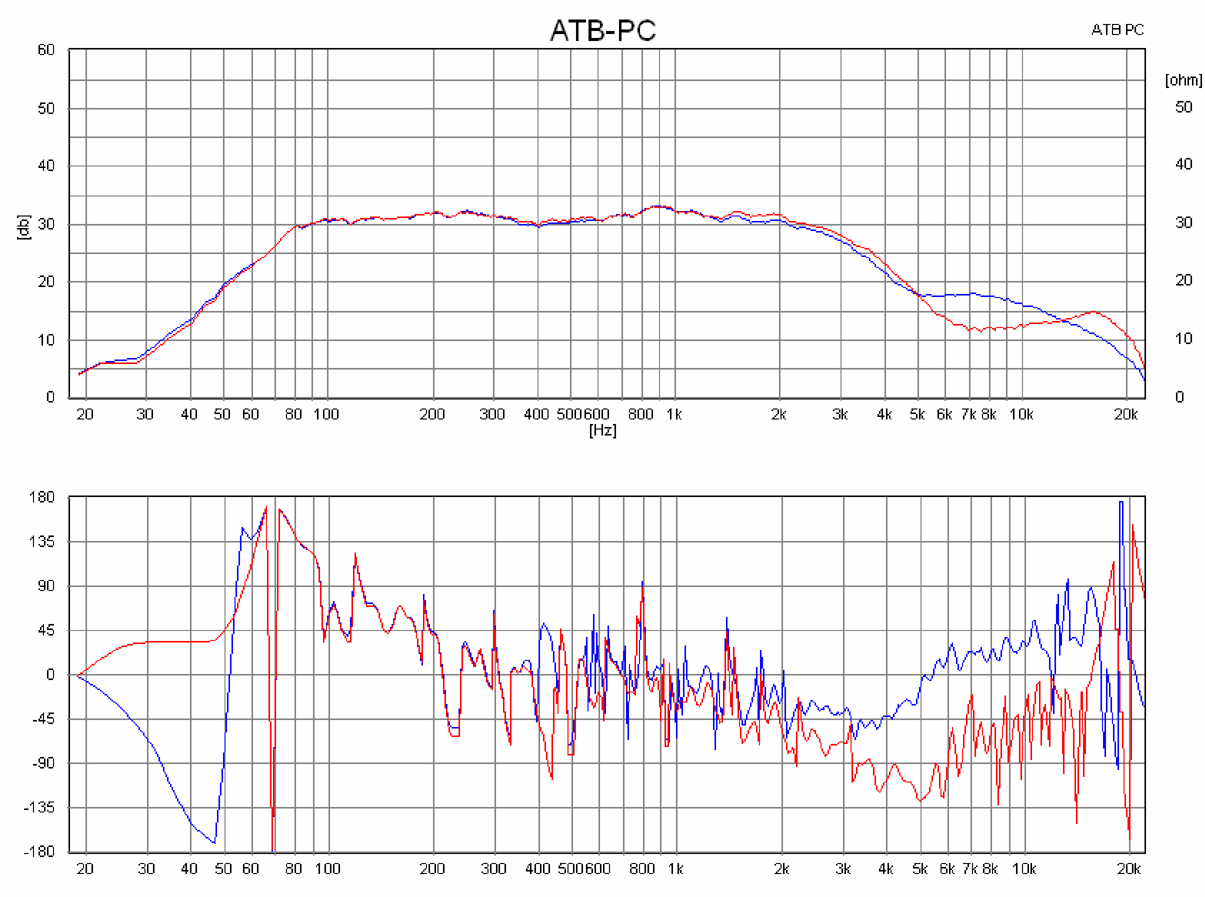


Figure 8.37

Figure 8.38 shows the influence of the drain circuit on the frequency spectrum and phase of the woofer. The related circuit diagram (Figure 8.40) and the measurement setup (Figure 8.41) are shown on the next page.



red = Tweeter with drain circuit, condensor and preresistor
blue = Tweeter with capacitor

Figure 8.38

Result: the upper diagram shows that the modification the pushed hump at app. 6.2kHz down, i.e. it was drained off. The rest of the peak from 7kHz onwards is so weak in intensity, that doesn't have a disturbing effect on the sound impression. Further more also here the phase has changed positively.

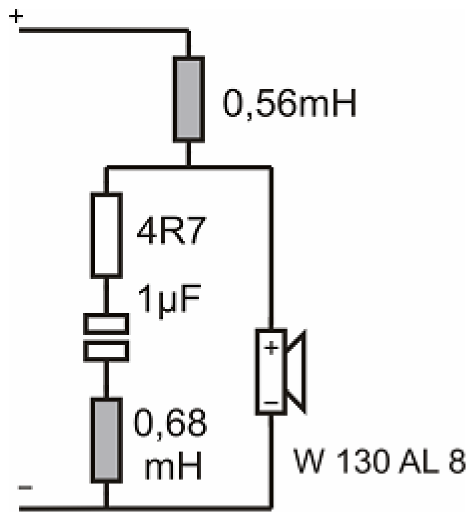
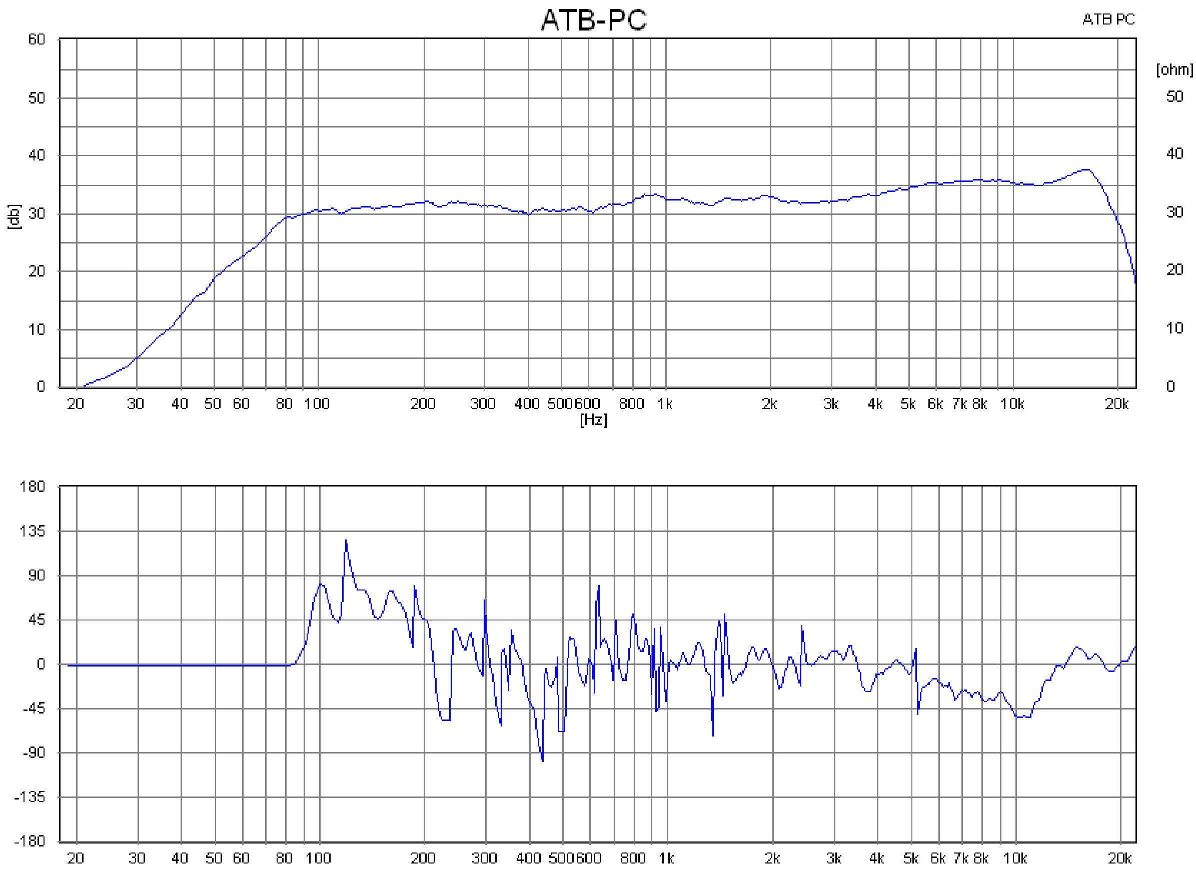


Figure 8.49

Figure 8.40 shows the frequency spectrum and phase of the complete loudspeaker with the final crossover circuit. The related circuit diagram (Figure 8.41) is shown on the next page.



red = Tweeter with drain circuit, condensor and preresistor
bue = Tweeter with condensor

Figure 8.40

Result: The blue curve shows the linear frequency sweep (above) and the perfect phase (below). The aim to create a correct phase loudspeaker with a linear frequency sweep was successful.

Note: Those that propagate that the phase is only something you can measure with instruments and has no effect on the sound quality and experience, are very welcome to come and have a listen to this speaker.

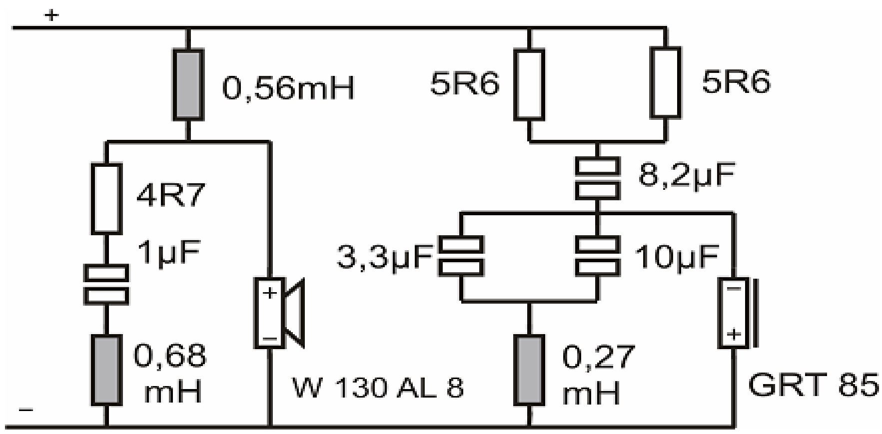


Figure 8.41

Figure 8.42 shows the phase of the finished speaker.

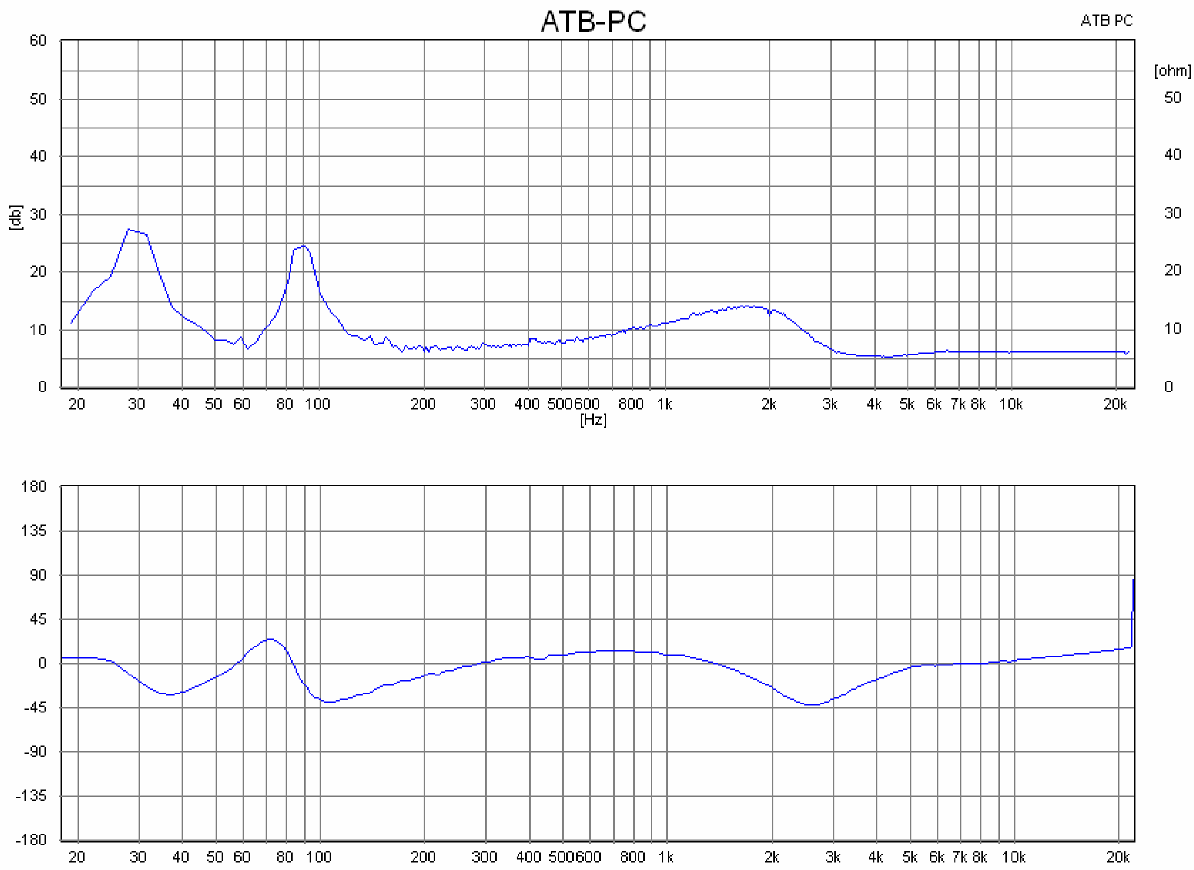
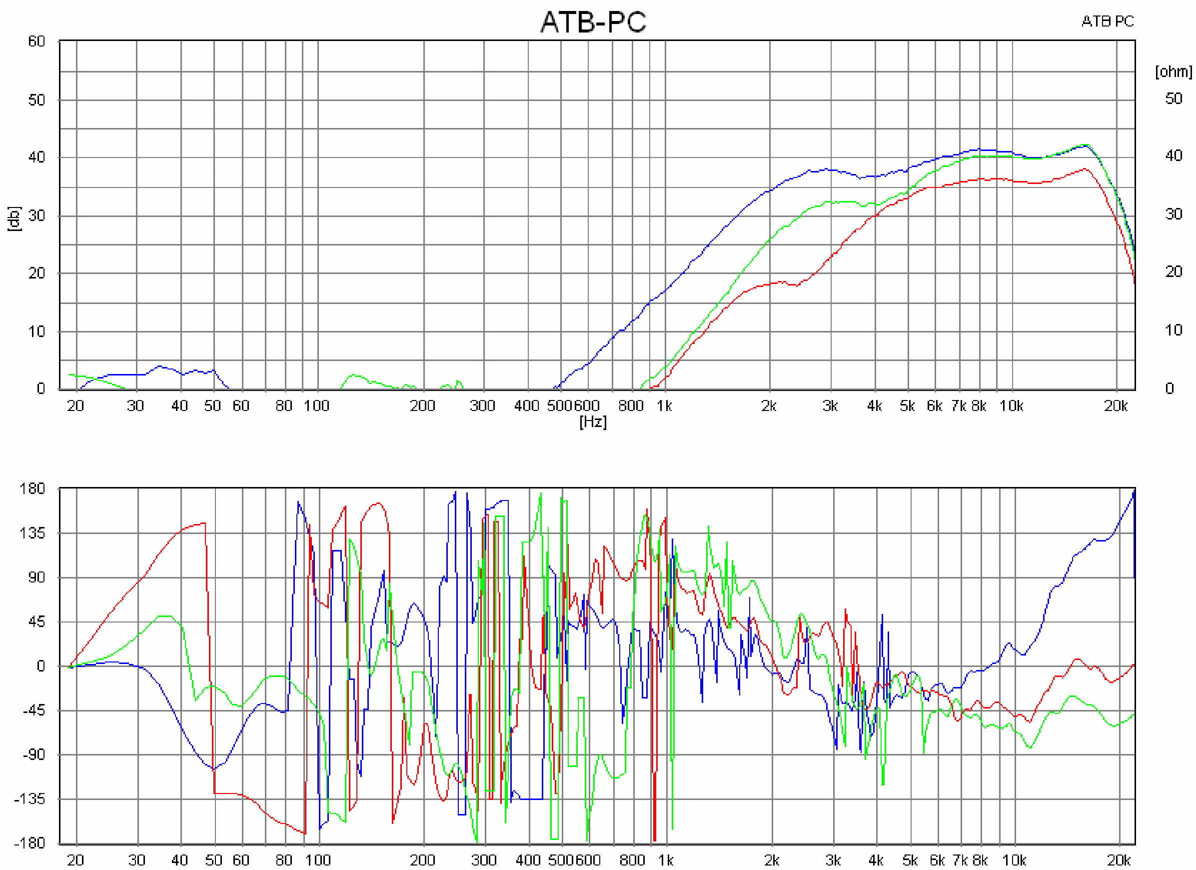


Figure 8.42

Result: The impedance peak at 30Hz and 90Hz in the upper diagram is due to the bass reflex opening. The further elapse of the impedance curve is to be seen as noncritical for amplifiers. It would have been possible to smooth the impedance at 1.8 kHz, but this would only have had a cosmetic effect and would have ruined the achieved frequency sweep results.

Adding up, the frequency sweeps of all development phases of the tweeter crossover are illustrated in figure 8.43.

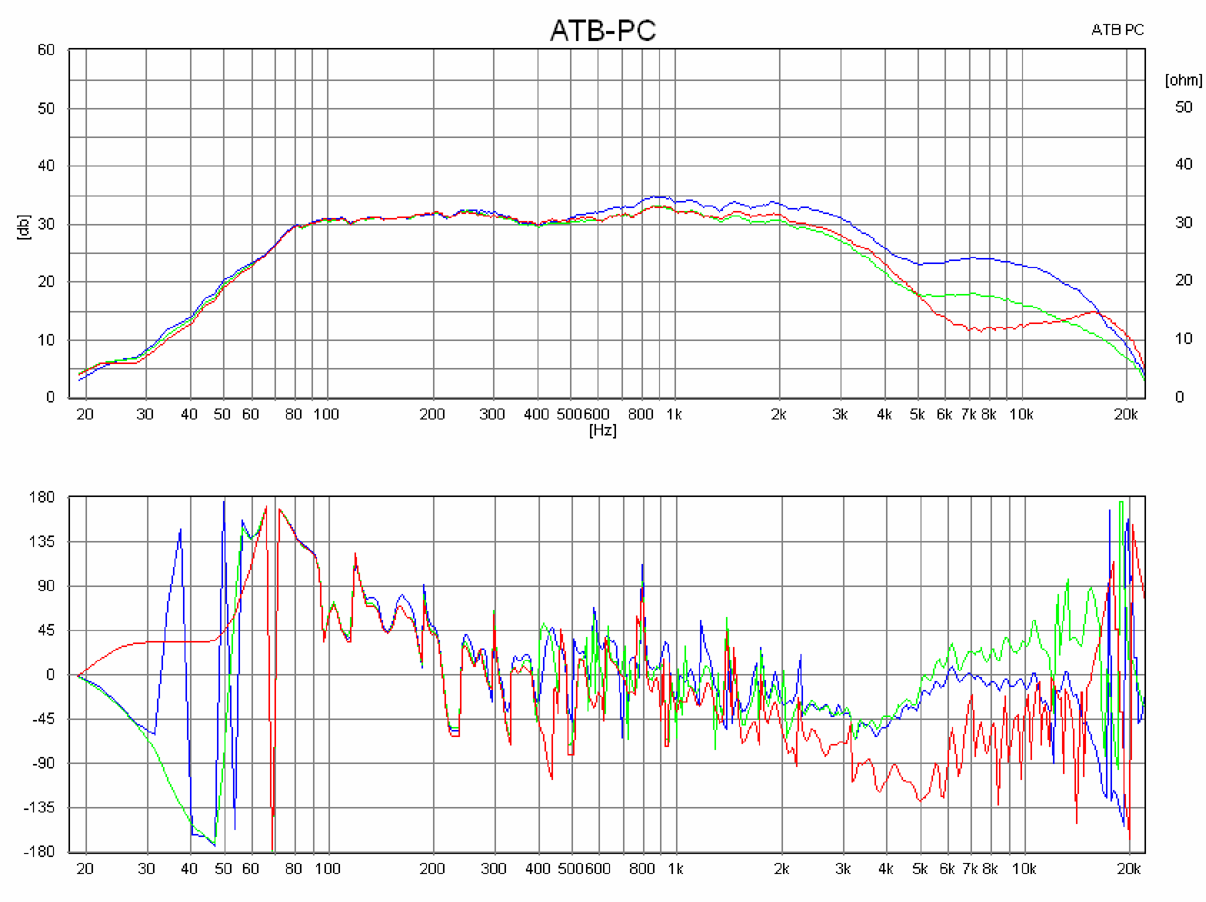


blue = Tweeter without crossover elements
green = Tweeter with condensor
red = Tweeter with condensor, drain circuit and preresistor

Figure 8.43

Result: The influence of the crossover on the frequency spectrum of the tweeter is easily recognisable. Important in this context is the fact that the phase is almost right back to where it started at the end of construction!

Adding up, the single frequency sweeps of the woofer during construction are put together in a single diagram Figure 8.44.

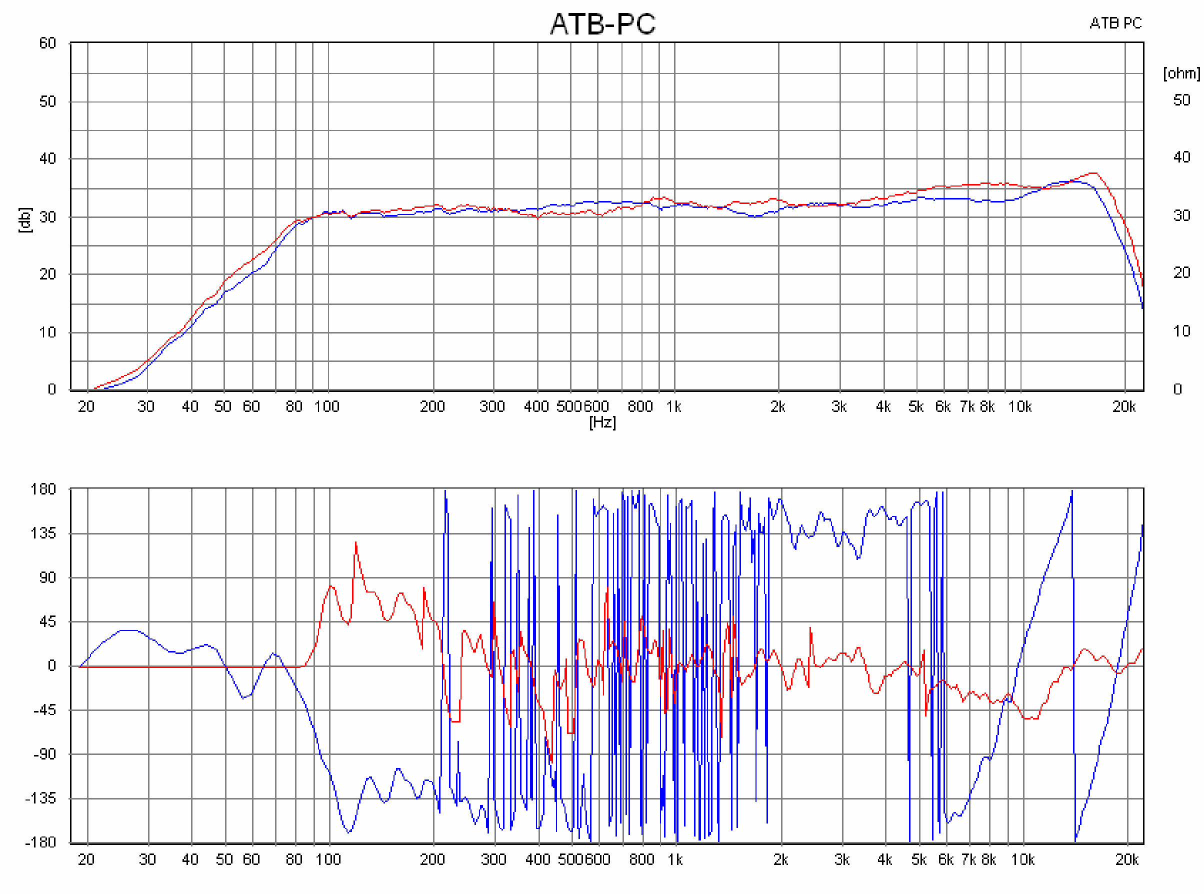


blue = Woofer without crossover elements
green = Woofer with coil
red = Woofer with coil and drain circuit

Figure 8.44

Result: Here also has the crossover has a noticeable effect on the frequency spectrum of the woofer. The phase hardly changes during the development phases at all. Like the overall phase in figure 8.46 the slight turn of phase at 5kHz doesn't have a disturbing influence.

Closing up in figure 8.45 the frequency sweep and phase of the Nugget is compared with a loudspeaker from a renowned manufacturer.



red = Nugget
blue = Reference loudspeaker

Figure 8.45

Result: Clearly recognisable, other manufactures achieve a linear frequency sweep, but do not give the acoustic phase any attention.

W8.7 Appendix

Resistor table for voltage splitters
Circuit diagram capital W8.2, Figure 8.2
All values in Ohm

Suppression in dB	R1 R2	4 Ohm Chassis	8 Ohm Chassis	16 Ohm Chassis
- 1	R1	0,4	0,9	1,7
	R2	33	66	131
- 2	R1	0,8	1,5	3,3
	R2	15	33	62
- 3	R1	1,2	2,2	4,7
	R2	10	22	39
- 4	R1	1,5	3	5,9
	R2	6,8	13,7	2,7
- 5	R1	1,8	3,5	7
	R2	5,1	10,3	20,6
- 6	R1	2	4	8
	R2	4	8	16,1
- 7	R1	2,2	4,4	8,9
	R2	3,2	6,5	12,9
- 8	R1	2,4	4,8	9,6
	R2	2,7	5,3	10,6
- 9	R1	2,6	5,2	10,3
	R2	2,2	4,4	8,8
- 10	R1	2,7	5,5	10,9
	R2	1,9	3,7	7,4
- 11	R1	2,9	5,8	11,5
	R2	1,6	3,1	6,3
- 12	R1	3	6	12
	R2	1,3	2,7	5,4
- 13	R1	3,1	6,2	12,4
	R2	1,2	2,3	4,6
- 14	R1	3,2	6,4	12,8
	R2	1	2	4
- 15	R1	3,3	6,6	13,2
	R2	0,9	1,7	3,5
- 16	R1	3,4	6,7	13,5
	R2	0,8	1,5	3
- 17	R1	3,4	6,9	13,7
	R2	0,7	1,3	2,6
- 18	R1	3,5	7	14
	R2	0,6	1,2	2,3
- 19	R1	3,6	7,1	14,2
	R2	0,5	1	2
- 20	R1	3,6	7,2	14,4
	R2	0,4	0,9	1,8

Table for Butterworth filter 1. degree
for 4 Ohm and 8 Ohm Loudspeaker
Flank steepness : 6bB pro Octave

Frequency	Z = 4Ω		Z = 8Ω	
in Hz	L in mH	C in µF	L in mH	C in µF
50	12,7	800	25,5	400
100	6,4	400	12,7	200
150	4,2	260	8,5	133
200	3,2	200	6,4	100
250	2,5	160	5,1	80
300	2,1	133	4,2	66
400	1,6	100	3,2	50
500	1,3	80	2,5	40
600	1,1	66	2,1	33
700	0,9	57	1,8	29
800	0,8	50	1,6	25
900	0,7	44	1,4	22
1000	0,6	40	1,3	20
1200	0,5	33	1,1	17
1500	0,4	27	0,8	13
2000	0,3	20	0,6	10
2500	0,25	16	0,5	8
3000	0,2	13	0,4	6,6
3500	0,18	11	0,36	5,7
4000	0,15	10	0,3	5
4500	0,14	8,7	0,28	4,4
5000	0,12	8	0,25	4
6000	0,1	6,7	0,21	3,3
7000	0,1	5,7	0,2	2,7
8000	0,08	5	0,16	2,6
9000	0,07	4,4	0,14	2,2
10000	0,06	4	0,12	2
12000	0,05	3,3	0,1	1,7
15000	0,04	2,7	0,08	1,3
20000	0,03	2	0,06	1

Table for Butterworth Filter 2.degree
for 4 Ohm and 8 Ohm Loudspeaker
Flank steepness : 12bB pro Oktave

Frequenz	Z = 4Ω		Z = 8Ω	
in Hz	L in mH	C in µF	L in mH	C in µF
50	18	560	36	280
100	9	280	18	140
150	6	190	12	93
200	4,5	140	9	70
250	3,6	110	7,2	56
300	3	90	6	46
400	2,2	70	4,5	35
500	1,8	56	3,6	28
600	1,5	47	3	23
700	1,3	40	2,6	20
800	1,1	35	2,2	18
900	1,0	31	2	16
1000	0,9	28	1,8	14
1200	0,7	23	1,5	12
1500	0,6	19	1,2	9,3
2000	0,4	14	0,9	7
2500	0,36	11	0,7	5,6
3000	0,3	9,3	0,6	4,6
3500	0,25	8	0,5	4
4000	0,22	7	0,45	3,5
4500	0,2	6,2	0,4	3,1
5000	0,16	5,6	0,36	2,8
6000	0,15	4,7	0,3	2,3
7000	0,12	4	0,25	2
8000	0,11	3,5	0,22	1,8
9000	0,1	3,1	0,2	1,6
10000	0,09	2,8	0,18	1,4
12000	0,07	2,3	0,16	1,2
15000	0,06	1,9	0,12	0,9
20000	0,04	1,4	0,08	0,7

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