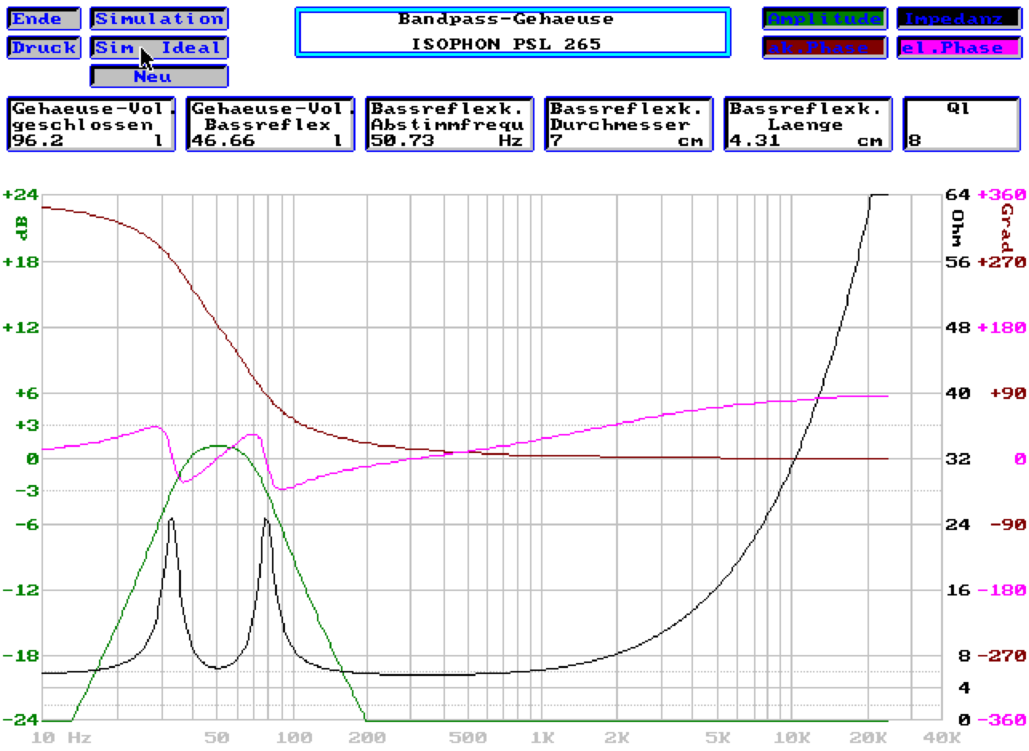
AudioCad

Pro 8.0 & ECNC 8.0

A Computer Aided
Loudspeaker Design Software



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1 General

1.1 Licensing conditions

 These instructions and this program have been created with the greatest of care. However, it is not possible to completely exclude the existence of some errors. Therefore the author would like to point out that he can neither guarantee the absence of any errors, nor is he responsible for any consequences which may occur, whether caused through mistakes in the manual or in the software. The author is however grateful for the notification of any faults which may arise.

Allowed: The installation of the program on only one computer, or on a computer which will be used for a time instead of the original one, or on a computer which will replace the initial one. You are allowed to hand over the whole program to a third party as long as they agree to abide by these conditions. Such a transaction requires you to either destroy all copies of the program or hand them over to the third party. The same goes for the version of the program installed on your hard disk.

Not allowed: It is forbidden to copy, modify, sub-licence, rent out, lend, transfer, translate and disassemble the program and its documentation.

1.2 Supplements to the documentation

Since the first printing of this manual changes or additions to the README.DOC have been documented. With the DOS commands
C:
CD\AC8
COPY README.DOC PRN
the file can be printed out. The command **SHOW README.DOC** displays the file on the screen.

2 Problems and solutions

A few typical support problems will be worked through here.

The mouse arrow cannot be moved over the whole screen: You should use an original microsoft mouse driver (mouse.com or mouse.sys), as found in many microsoft programs.

The program sometimes behaves strangely: Determine how much free memory for the user program (....free bytes) of your PC exists, using the MS DOS command CHKDSK. When there is less than about 530 KB then the program may not run correctly. Obtain more free memory through appropriate configuration measures in the **autoxec.bat** and the **config.sys** files.

The print out from the graphics does not work: You have probably forgotten to enter your printer in the installation mask.

I always get two copies of the graphics when I print: You must not allocate a value larger than 0 to more than one field **Printer Port** in the installation mask.

The graphics are distorted. The circles are never round. You can adjust the graphic by using the X-zoom and the Y-zoom of the installation mask, in the second graphic hardcopy routine.

The program cannot find my audio measurement system files: Enter the data directory of the convertion program C:\AC8\MESS\ in the installation mask and in the interface program.

How large main memory does the program need? Refer to the above problem - **The program sometimes behaves strangely**

When I call the database I always get error messages and the program breaks down: Activate the database reorganization in the main menu.

How can I read the absolute values (dB) off the graphics? For both the loudspeaker database and the enclosure simulation the **0 dB line** corresponds to the **SPL** of the respective driver, however the 0 dB line in the crossover simulation corresponds to the SPL of the woofer.

3 Program installation

3.1 Hardware requirements

An IBM compatible PC-AT (80286) or a more advanced version with at least 640 KB of main memory, a VGA graphic card, a microsoft (compatible) mouse and hard disk and MS-DOS from version 3.2 onwards or any compatible operating system is required, in order to install this software.

During installation about 5 megabytes of memory will be temporarily required on your hard disk. After installation the program will only occupy about 3 megabytes of hard disk space.

Optional: CoProcessor (80287, 80387, 80487SX), EPSON-FX80 (9 dot), NEC-P6 (24 dot), HP-Deskjet (inkjet), HP-Laserjet (laser printer) or any corresponding compatible printer.

3.2 Installation

3.2.1 General

Place the CD-ROM into your CD-ROM-Drive and start the installation process for example by typing in
D:
INSTALL

Take into account exactly the notes you given by the installation program.

The installation routine now installs the program onto your hard disk. During installation the index files of the databases will first be created and then the program will be called up. The installation screen will automatically appear during the first installation. If you install the program for a second time you will have to call the installation mask as follows: Activate the **Alt key** in the main menu and select the **Utility** menu with the cursor control key and the return key. By selecting the menu item **Install** and pressing the return key the installation screen will appear.

All menus are also operable using your microsoft (compatible) mouse. This is naturally more convenient than using the keyboard.

When using a VGA colour monitor, with the exception of the printer adjustment, all parameters in the installation mask are pre-allocated in a way so that the program is optimally configured for standard use. Therefore all that remains for you to do is to define your printer, as described below, and you can immediately work with AudioCad. You can alter the installation at any time.

The various configuration possibilities will be described below. Now start the menu item **INPUT**.

Language: Should your diskette contain the international version of the program you can choose between German (**0**) and English (**1**).

Colour-monitor: The program only runs on PCs with VGA graphic cards. At this point you must type in whether your PC has a colour screen (**j**: Yes) or a monochrome screen (**n**: No).

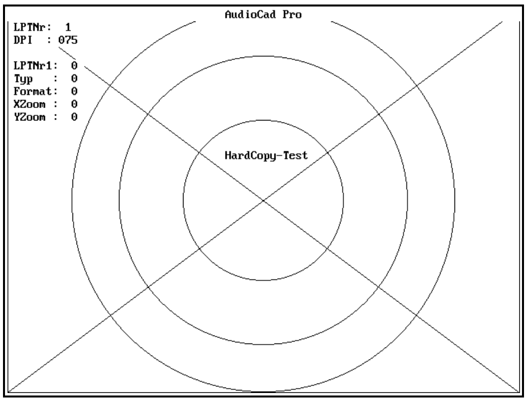
3.2.2 Printer Adjustment

Graph-plot routine 1 & 2: Two different hardcopy routines have been built into the program.

The first only requires the parameters **Printer port** and **Dots per inch**. It is only suitable for HP-Laserjet, HP-Deskjet and EPSON-FX80 compatible printers. This routine works faster than the second one.

The second routine has the parameters **printer-port, printer-type**, **format**, **X-zoom** and **Y-zoom**. It is suitable for EPSON LQ500 and NEC P6+- (24 dot) as well as EPSON-FX80 compatible printers. You can select the desired hardcopy routine through the **printer port** parameter. If you give in both **printer port** fields values greater than zero then you will always get two print outs!

By using the menu item **Test Graphic Hardcopy** you can test the adjustments.



Printer port: Enter the number **1** here. If there is no printer connected to your PC, or if you want to use the other hardcopy routine then enter the number **0**.

Dots per inch: The type of printer will be defined through the **Printer port** parameter and the **Dots per inch** adjusted for a laser and ink printer. If the field **Dots per inch** is left blank then the program will automatically select a dot matrix printer (9 dot, EPSON-FX 80 - compatible). If you are using an HP-laserjet compatible laser printer or an HP Deskjet you can type in one of the adjustments **075**, **100**, **150** or **300** dpi afterwards. The resolution 075 dpi prints large graphics in a landscape layout. The adjustments 100, 150 and 300 dpi print smaller graphics in a portrait layout.

Printer port: Should you wish to use the second hardcopy routine enter the number **1** here. If no printer is connected to your PC or you wish to use the other hardcopy routine enter a **0**.

Printer type: Enter **1** for a 24 dot and **2** for 9 dot.

Format: Enter **1** for a portrait layout or **2** for a landscape layout.

X-zoom and **Y-zoom:** The printout can be influenced through these fields. The following recommendations are given:

*** EPSON FX80** and **IBM GRAPHICS**
- format : 2
- X-zoom : 1
- Y-zoom : 3

*** NEC P70**
- format : 1
-X-zoom : 4
-Y-zoom : 2

Form feed: Type in **1** here if the program should select a new page after it has printed out graphics. If **0** is entered then no new page will be selected and it is possible that you will have more than one graphic per page.

3.2.3 The General group

Graphic-dB-scale: By entering **0** the scale for the amplitude of the graphics ranges from ±48 dB, it is reduced to ±24 dB when **1** is typed in.

dB-scale: If you enter a **1** here the display will show the amplitude relative to the SPL for the respective driver, an input of **0** will display the amplitude in absolute values (dB).

Graphik-Ω-scale: The following scale ranges for the impedance display are possible:

Graphik-Ω-Scale	Ohms
0	0 - 16
1	0 - 32
2	0 - 64
3	
4	0 - 256

Graphik-lines: In order to make the graphics more legible help lines can be blended in. By inputting **0** no help lines will be blended in, inputting **1** blends in 8 lines and inputting **2** blends in 24 lines.

Lower Sim-Frequ: This gives the frequency from which point the simulations work. This frequency depends on your audio measurement system, and should be adjusted accordingly. If your measurement system can measure, for example between 20 - 20000 Hz, then enter 20 as the **Lower Sim-Frequ** and 20000 as the **Upper Sim-Frequ**.

Upper Sim-Frequ: This gives the frequency up to which point the simulations will function.

Show Function Keys: As a standard a list of function keys (**1**) will be overlayed on the main menu. When you are familiar with the functions you can remove the list by entering **0**.

AutoActivate menu: When **1** is entered here the pull down menu will automatically operate on calling a mask. If **0** is entered then the menu must be activated using the mouse or Alt key. The automatic activation (**1**) is not recommended, as when coming out of a graphic to the mask it is possible that the graphic may unintentionally be recalled.

Abort Key (F5): You can leave the program immediately by using the function key **F5**. This can however be very troublesome when the key is accidentally activated, Therefore the key is normally deactivated (**0**). If however you want to use the function key **F5** then give **1** in here.

3.2.4 The cabinet draw group

The cabinet height, width and depth: The parameters entered are the standard proportions of a enclosure which will be shown in the **cabinet-draw** mask. Experiment a bit in this area. The configuration can be changed at any time.

Wood thickness: Enter in millimetres the thickness of the board you normally use. This parameter will be necessary for the **cabinet-draw** mask, but can be changed at any time from within the mask.

3.2.5 The enclosure simulation group

Sim-precision: Through this parameter you can adjust the calculation precision and thus also the calculation time of the processing speed of your PC. The values **1**, **2**, **4** and **8** are allowed here. The lower the number the more exact the simulation and therefore the longer the calculation time.

Sim-precision	Number of sim-points/curve (10 - 40.000 Hz)
1	512
2	256
4	128
8	64

Double speaker: The program makes the simulation of several parallel or serial connected drivers into one crossover network possible. Such a construction would usually be used for a double bass system, as one can have slimmer enclosures by replacing one large woofer by two smaller ones. For this purpose with every new initialisation of a driver (when leaving the database mask) follows a corresponding check. As this in general is not necessary for mid-range speakers or tweeters, the check remains hidden (parameter **0**). Through the double driver parameter you can also install the check for mid-range speakers and tweeters so that two drivers can operate in parallel. To achieve this enter the parameter **1** (the check for **Woofer** and **Mid**), **2** (the check follows for **Woofer, Mid** and **Mid/Tw.**) or **3** (the check for all drivers).

Cone Scale: The y-axis scaling of the membrane excursion simulation will be automatically adjusted for each curve when the parameter has the value **0**. However, you can also define a field of values. If the parameter is given a value of 10 then the diagram of the membrane excursion will have a scale of 0-10 millimetres.

3.2.6 The crossover group

Sim-precision: This value determines the number of frequency points to be simulated in the crossover network simulation and optimization. For a more exact description see above.

Coil-Resistance: When this parameter is chosen to be greater than 0 (ohm), the crossover network connection diagram will show an ohm resistance switched for each inductance (coil). The sense behind this is to include the DC-resistance of the coil in the crossover network simulation. The value of the resistance can be altered later in the crossover network diagram.

Acoustical centre: Through this parameter the function for considering the acoustical centres of the single chassis in the crossover network simulation can be switched off (**0**). The standard presetting is **1**, that is to say that the acoustical centres will be taken into consideration.

Crossover-Type: Please only choose **0**.

3.2.7 The crossover optimization group

MinValue Component: When the value of a constructional element (resistance, condenser or coil) falls short of this stipulated amount it will no longer be changed by the optimizer.

MaxValue Component: When the value of a constructional element exceeds this stipulated value it will no longer be changed by the optimizer.

The **opt. start divisor**, the **opt. multiplicator** and the **opt. end divisor:**
The optimizer changes the constructional part values which are to be optimized respectively to a fraction of the initial value. If, for example, the initial value is 3.3 microfarad an opt. start divisor of 10 will vary the initial value by 0.33 microfarad. If further changes in the constructional element values of 1/10th steps do not cause substantial improvements then the divisor (in this example, 10) will be multiplied with the opt. multiplicator (likewise, 10) and the optimising procedure will continue. This step will be repeated until the opt. end divisor no longer brings about a substantial improvement.
Because of this input structure any arbitrary optimization divisors can be pre-allocated. The standard value is 10, 10, 100 which produces the divisor sequence 10, 100. The values 10, 10, 1000 would also be another possible example, which by producing the divisor sequence 10, 100, 1000 would lead to very accurate optimizing results. Normally, however this would result in an unnecessarily long processing time with no practical use. You could achieve, for example, the sequence 5, 25, 125 by selecting the initial values 5, 5, 125. The configuration could possibly lead to a faster optimization time. This depends on the respective constructional part values at the start of the optimization.

3.2.8 Miscellaneous

Graphic Headline: The text entered here will appear as the title of the graphic when printed out (for example your company name).

Project directory: AudioCad will save your project files in this directory. The standard directory is **C:\AC8\PRO**. Should you enter an alternative directory you must first create it with the DOS command **MD** (make directory) before you can use it.

Measurement-directory: Here you must enter the complete path of the directory in which the measurement system interface program can store the converted measured value files. Normally this is **C:\AC8\MESS**. Should you enter an alternative directory you must first create it with the DOS command **MD** (make directory) before you can use it. Please keep in mind that in this case you must make appropriate adjustments to the **AudioCad directory** field in the Audio measurement system interface program. (It is best to keep the standard preallocation and everything will work out).

The menu item Test Graphic-Hardcopy: You can test the configuration of your hardcopy routine with this function.

The menu item Graphic colour: Here you can adjust the background colour for graphics. The complete VGA colour palette (262144 colours) is available for this purpose. After selecting this menu item a test graphic appears. Now you can create the desired background by mixing the colours blue, green and red. The colour intensity can be regulated by using a key. By selecting **Shift B** (capital B), for example, the intensity of blue will be enhanced. A **lower case b** will reduce the intensity of blue. The procedure for changing red and green is analogous. Once you have obtained the desired colour you can leave the function by pushing the space bar.

Now you can leave the mask by selecting **Exit**. The program will ask you if the configuration is to be saved. Select **<Yes>**. Leave the program by selecting **Exit** in the database menu. Now the installation is complete.

Should you make a "total mess" of the configuration delete the file **AC.CFG** from the DOS command level and call up the main program. This will recreate the standard presettings.

3.3 Deinstallation

Now all that remains is for you to save the databases (files **BOX.DBF** and **ACMESS.DB**) and also your projects (files ***.PRO**) onto another diskette. Install the program in the normal way onto your new hard disk, and after completing the installation process copy your databases and projects onto the new hard disk. Finally you start the **database reorganization** from the Main menu.

4 General program description

A short description on how to construct a box will be given below.

Enclousure Construction

1) Finding the Thiele/Small-Parameters
The best to measure Thiele/Small-parameters is using an audio-measurement-system. If none is available you can also consult the manufacturer data.

2) Creating a new AudioCad-Loudspeaker-database
This step isn't absolutely necessary, contributes however fundamentally to the general view. You can deposit the loudspeaker data of course too in an already existing database. The author manages every project in a separate database.

3) Creating an new loudspeaker in the data base.
For the enclosure-construction only the Thiele/Small- and mechanical parameters are needed. Measuring data (amplitude, impedance,...) aren't necessary.

(4) Initialization of the loudspeakers for the calculation
see chapter „the menu database“ and „the menu End“.

5) Enclosure Construction
Construct as explained below more nearly for the single drivers of the system enclosures (closed, vented, ...)

6) Mechanical enclosure construction
Construct the box using the menu item „cabinet draw“.

7) Create and save a project for the enclosure construction

Crossover network construction

If you don't have an audio measuring system you can directly develop a crossover network using AudioCad now. But realize that this procedure leads as a rule to rather large errors at the crossover network construction.

If you have an audio measuring system proceed as following:

1) Build the Enclosure
Build the enclosure including the Loudspeakers but without crossover network. The loudspeakers are connected so that they are accessible singly form the outside.

The author uses a modified connection terminal, which makes possible the connection of up to 4 loudspeakers. After completion of the Crossover network it is replaced by a normal one.

2) Measuring the loudspeakers
The amplitude, the acoustical phase, the impedance and the electrical phase of the single drivers are measured in the enclosure without crossover network.

The measurings must be absolutely subject to the following conditions:
Measure the amplitude, impedance and phase progression of all drivers in the cabinet. The position of the microphone must not be altered during the measurement otherwise the acoustical phase information of the drivers will be incorrect in relation to each other. Don't put the microphone in front of the chassis to measure! The microphone location always must be with respect to the complete box in the same place. Finally the

human ear is , when listening to a box, also only in a point and don't jump to and fro in the chassis levels. The measuring point should be in typical ear height (approximate 100 centimetre above the floor). A microphone distance is favorable of approx. 1 - 2,5s meters.

3) Konverting of the audio measuring system files with the interface program

4) Creating new loudspeakers in the data base.
The loudspeakers are created in the database with a new loudspeaker name to indicate that it is in an enclosure measured data.
E.g.: **ISOPHON PSL 155 ALU** or **ISOPHON PSL 155 ALU *G**
The same value must be entered in the field **SPL** for all drivers of the system.

5) Importing the measured data into AudioCad
Feed in the same value into the field **0-dB point** for all drivers of the system!

With the value of the field **0-dB point** the amplitude curves can be moved up or below (try out) on the Ys axis. The **0-dB point** of the woofer should correspond to the 0-dB line of the AudioCad diagram. Have a look at the amplitude graphic to check this after data import. The levels of midrange and tweeters lie as a rule over this line, because they usually have a bigger SPL as woofers. Midrange and tweeters with lower SPL than the used woofer are unsuitable in combination with this woofer! You then initialize the single loudspeakers for the crossover network construction.

6) Crossover network construction using AudioCad
Don’t construct enclosures in this case! The drivers already were measured in the enclosure! Develop the crossover network as described in the appendixes.

7) Scrutinizing structure and measuring the box including the crossover network
Build the crossover network, measure the results and perform the hearing test. The microphone position must still be the same as described above **(2)**! If the expected results are not obtained restart from further up.

Measure the components (resistances, inductances, condensers) of the scrutinizing structure before. At differing component values you think otherwise perhaps, this program would simulate rongly. The usual 5 per cent component tolerance altogether can lead to a large difference!

8) Save your crossover network construction as a project

4.1 Calling up the program

In order to call up the program type in the following commands at the DOS level:

C:
CD\AC8
AC

To call the program please do not apply a user interface, for example, MS-DOS shell, as in this case valuable main memory, which is necessary for AudioCad, would be lost.

4.2 Using AudioCad with MS-Windows 3.1

Under MS-Windows AudioCad is executable as a DOS application in the full screen mode. Therefore it is possible to load, for example, the Audio measurement system into one window, the AudioCad audio measurement system interface into another and the AudioCad main program into a third window. Thus, during the development of a loudspeaker combination all necessary tools are simultaneously available. A disadvantage in this is however the loss of performance caused by MS-Windows. Especially the optimizer (this part of the program needs the most performance) runs slower than under DOS. Depending on the configuration (contents of the files autoexec.bat and config.sys) it is possible that even in the DOS window there will not be enough main memory available (see **Questions and Answers**). The procedure for using AudioCad under MS-Windows will be described briefly below.

4.2.1 Instructions for users unfamiliar with MS-Windows

Start MS-Windows and double-click to call the MS-DOS window. Call AudioCad as described in the program call-up section.

Hint: Before calling AudioCad for the 1st time under MS-Windows you should check the memory available under DOS. To do this enter the command **MEM** in the MS-DOS Window. You will find the necessary memory requirements for AudioCad under **Questions and Answers**. If your configuration does not give you enough memory and you are using MS-DOS 6.0 you can get your memory optimized by MS-DOS. The program **MEMMAKER** within MS-DOS 6.0 is responsible for this. It is simple to use, just consult your DOS manual.

4.2.2 Instructions for experts in MS-Windows

Create two PIF-files for AudioCad Pro using the Pif-editor. One PIF-file is for the Audio measurement system interface, the other one is for the main program. To do this copy the MS-DOS Pif-file and only call up the COMMAND.COM from the Pif-file. You can also call up the program names (ACPRO.EXE and KONVERT.EXE) directly from the Pif-file, however the program will run substantially slower than if you first start COMMAND.COM and then load the application. Icons can be allocated to the Pif-files. AudioCad comes with an icon library **ACICONS.ICO** which contains some icons that function under MS-Windows.

Calling up the program ACPRO.EXE: Start the main program under MS-Windows as follows:
CD\AC8
SET DBPOOL=16
ACPRO
The set command is very important. By omitting this command the program will use 48 KB more conventional memory!

Calling up the program KONVERT.EXE: Start the interface under MS-Windows as follows:
CD\AC8
KONVERT
In addition please read the hints in the section **Instructions for Users Unfamiliar with MS-Windows**.

4.3 General program operation

The program can be operated either with the keyboard or with a microsoft compatible mouse. Using the mouse is naturally the easiest way. All menus can be selected with the left mouse button. The right button closes a pull-down menu. Once you have worked a bit longer with the program, you will inevitably find quicker ways to operate. The so called HOTKEYS are for this purpose. A hotkey is the highlighted figure (either letter or number) of a menu item, by use of which the menu item will be selected and carried out. Once you are familiar with the program you will know the individual menu items by heart. With time the hotkeys will imprint themselves in your memory on their own accord. Then, handling the program will run substantially quicker when using the keyboard rather than the mouse. The switch from mouse to hotkeys can happen smoothly as the mouse is still at your disposal. The following will give a short description of the program operation using the keyboard which generally corresponds to SAA-standards.

Program operation by keyboard:

Alt-key : Displays hotkeys for the menu
key : Selects and carries out menu item (hotkey)
cursor keys : Select menu item
return key : Activates menu item
Escape key : Closes pull-down menu

Operation of the selection-boxes using the keyboard:

TAB key : Selects an alternative
Return key : Confirms the choice

Allocation of function keys:

F3: Removes active speaker from the main memory
F4: Selects active speaker
F5: Aborts program
F8: Prints a graphic screen
F9: Aborts optimizing process

Warning: The abortion of a process concerning a loudspeaker database (everthing contained in the main menu Database) by using the **F5** key can, in the worst case, lead to a destruction of the database, and should therefore not be used. If such an error should occur with the database then restart the database reorganization from the main menu.

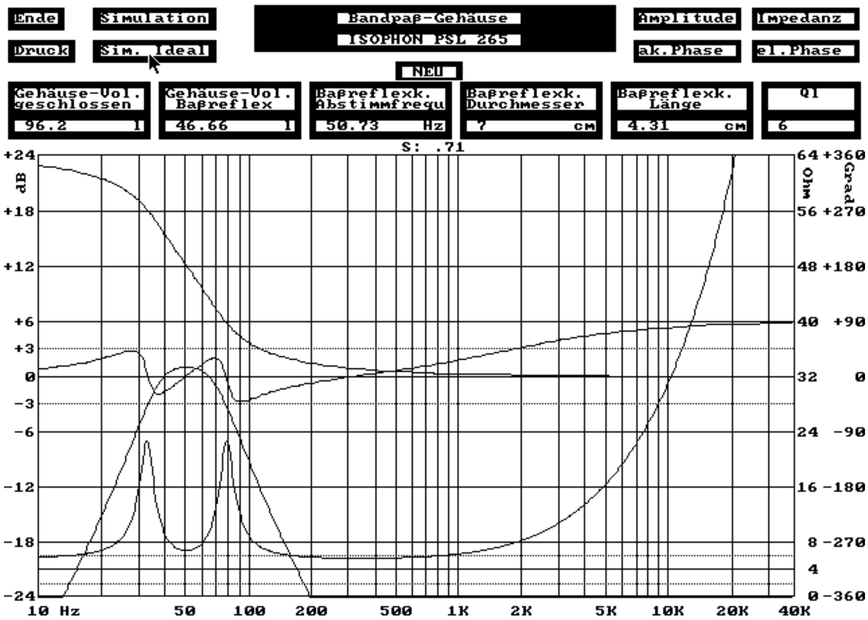
Cursor up or down: This takes you to either the next or the previous input field or positions you on either the next or the previous menu item.

Page down key: This allows you to leave the input field of a mask and return to the menu.

The program is to a large extent self-explanatory if you have experience with loudspeaker construction and the THIELE-SMALL - parameters. Dialogue messages in the bottom part of the screen help the user with the input. False or incomplete inputs will be rejected with the corresponding error message. However keep in mind that not every error is recognizable as such. Wrong inputs can also lead to incorrect calculation results. It is unlikely that the program will crash through incorrect usage, as all parts are secured. If a "fatal" error occurs, which cannot be countered, the program will, in the worst case, return to the main menu. If this happens simply recall the mask and correct your input data.

4.3.1 Program operation in graphics

In graphics, functions are available which are partly only attainable using the mouse. Various graphics offer comfortable program operation by using so called switch fields. These are the "keys" which can be selected in the upper part of the graphic. These switch fields, which will be described below, can be activated by clicking on the left button of the mouse.



EXIT: The graphic will end and the program will return to the menu. Alternatively you could use any other key. In the part of the program for the cabinet construction you may eventually have to press the key twice.

Print: The graphic will be printed. You could also use the function key **F8** if you want to print a graphic.

Plot: The graphic will be newly drawn or a new simulation will be carried out.

Simulation: A new simulation will follow with the inclusion of the measured data (amplitude, impedance etc).

Sim. Ideal: A new simulation will follow without taking the measured data (amplitude, impedance; phase) into account.

NEW: If the text is showing in this box then the screen will be cleared before each new simulation, so that only the newly derived curves will be displayed. By clicking on this switch field you can get rid of the text "new", and then in a new simulation the old curve will appear in grey with the newly simulated curve overlayed. In this way one can compare several tuning results on one graphic.

Closed cabinet volume or bass reflex cabinet volume: By clicking on this switch field you can enter another cabinet volume.

Vent frequency: Here you can alter the Helmholtz resonance. The vent-length will be newly calculated.

Vent diameter: The vent-diameter can be changed. The vent length will be newly calculated.

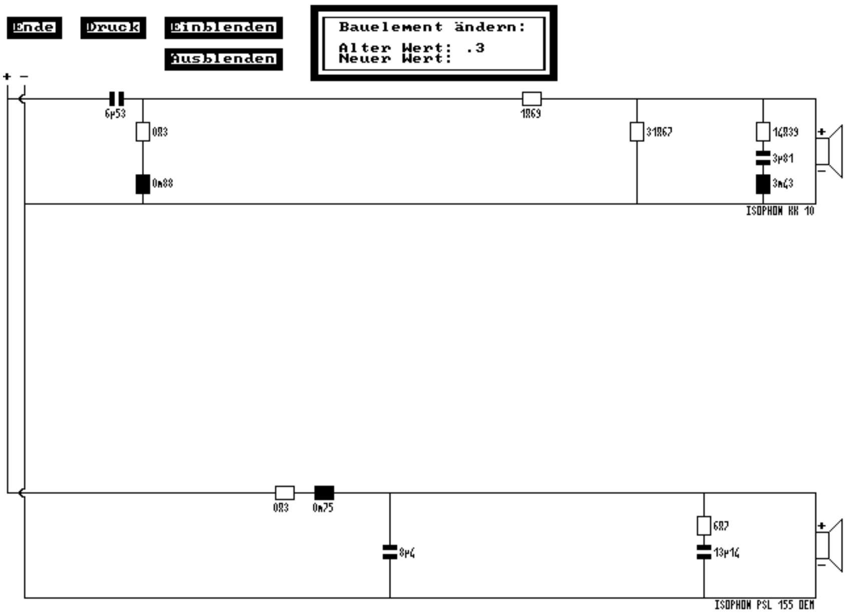
Vent length: The vent length can be altered. The tuning frequency will be newly calulated. By doing this the alteration of the tuning frequency due to short vents can be judged, if the required vent length does not fit in the cabinet.

QL: The dissipation quality of the cabinet can be altered.

Amplitude, ac. phase, impedance, el. phase: With the help of the switch fields you can select which curve should apppear in a graphic. If a switch field appears in the same colour as the affiliated curve then the curve will be pictured. If a switch area is not coloured, then the specific curve will not be shown.

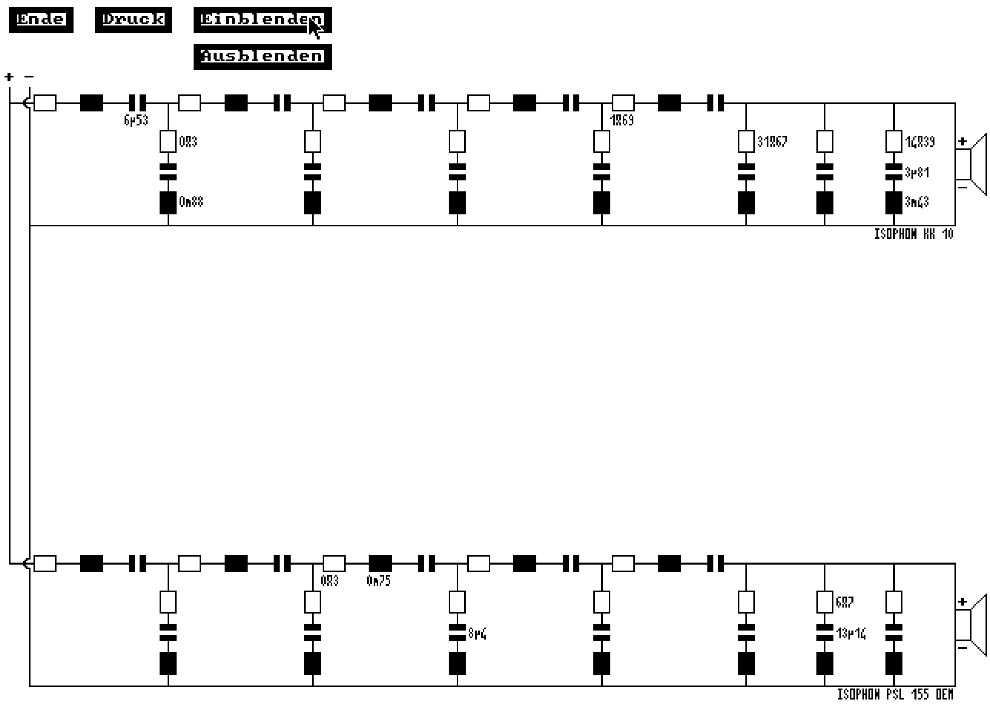
Special features of the crossover network circuit diagram:

Changing the constructional part values: Click on the respective constructional element with the left mouse button. The constructional element will turn red and an input box will appear in which you can enter the new value. If you do not wish to change the value simply press the return key or enter 0 and press RETURN. Answer the question "Delete Component?", which will follow, by clicking on **No**.



Loudspeaker polarity: By clicking on the loudspeakers you can exchange the polarity.

Show all: By clicking on the box **Show all** the maximum equipment of the circuit diagramm will be shown. By using the function Change Values of Constructional Elements, as described above, you can insert further construction elements onto the screen.



Hide: Only the constructional elements in use will be displayed.

Removing a constructional part from the circuit diagramm: Click on the desired constructional element and enter the value **0** in the input box which appears or simply use the return key. Answer the question **"Delete Component?"** which will now appear, by clicking on **Yes** with the left mouse button.

4.3.2 Printing out mask-contents and graphics

To print out the contents of a mask simply use the **print** key. Each graphic can be printed either by using the function key **F8** or by clicking on the **print** field within the graphic. You must however define your printer beforehand in the program (see installation)

If you are not satisfied with the printing routine in AudioCad you can also use the graphic print routine of the operating system (program **Graphics**). This is suitable for VGA-monitors with MS-DOS Version 5.0 upwards, and if you have MS-DOS release 6.0 or higher then there are numerous print drivers, one of which is certain to be your printer. The program Graphics produces very fine printouts but unfortunately it is quite slow.

If you operate AudioCad under MS-Windows (see above) then you can also print under MS-Windows. Proceed as follows:
- Call up the graphic in AudioCad and use the print key.
- Use the **Alt-** and **TAB-** (on the left of q) keys simultaneously to switch back to the program manager.
- Start the MS-Windows program **PAINTBRUSH**
- Activate the menu item **Picture Attribute** in the options menu and put in the unit **PIXEL**, the breadth **640**, the height **480** and **colours**.
- Select the menu item **Insert** in the menu **Edit**. By doing this you have copied the AudioCad graphic into **PAINTBRUSH**.
- Now chose the menu item **Invert** in the menu **Trick Box**.
- Finally activate the menu item **Print** in the menu **File**.
- Use the **Alt-** and **TAB-**keys simultaneously to switch back to AudioCad.

4.4 Technical information about the program

As the program is bigger then 640 KB and therefore cannot normally run under MS-DOS, it has been constructed in the overlay-technique, that is parts of the program will be loaded later from the hard disk, when needed. If your PC is equipped with an expanded memory (after LIM-EMS version 4.0-standard) the overlays will be held resident in the EMS. This is more time saving than any reloading from the hard disk. An EMS memory manager (program **EMM386.EXE** or **EMM386.SYS**) is for example included in the MS-DOS release 5.0 and above.

If your PC has more than 1 megabyte of main memory you should definately install a hard disk cache. This will speed up the database access considerably. Use the program **SMARTDRV.SYS** or **SMARTDRV.EXE**, for example, which come together with MS-WINDOWS and MS-DOS.

5 The Main program (ACPRO.EXE)

5.1 The Main menu

In the Main menu you will at first notice four lines with the text, **Woofer**, **Mid**, **Mid/Tw**. and **Tweeter**. The active driver will be chosen through these. This can be done simply by clicking on the field between "(" and ":" with the left mouse button, or by using the cursor key and return key after activating the function key **F4**. All calculations now refer to this driver. To clarify the combination possibilities here are a few examples.

- 2-way speaker

Tweeeter : tweeter
Mid/Tw : not allocated
Mid : not allocated
Woofer : woofer

- 3-way speaker

Tweeter : tweeter
Mid/Tw : not allocated
Mid : mid-range speaker
Woofer : woofer

- 4-way speaker

Tweeter : tweeter
Mid/Tw : mid-tweeter
Mid : low mid-range speaker
Woofer : woofer

By clicking on the box **Delete Speaker** or by using the **F3** function key you can remove the actual marked driver out of the main memory again.

5.2 The menu Database

5.2.1 The menu item Database

In this mask you will gather all the necessary loudspeaker data for the construction of loudspeaker systems. An input in the respective calculation or simulation mask is not possible. In order to use all of the possibilities offered by the program you must deposit at least the data fields Speaker, Speaker-Type, Re, Pe, fs, Qms, Qes, Vas, Le, Mms, Chassis diameter, Membrane stroke, Acoustical centre and SPL next to the measuring data (amplitude, impedance and phase). The rest of the information you can give in for its own sake.

Hint: When a project is loaded and you intend to load a woofer from another database, you first have to delete the previous speaker from the desktop by clicking at the box **delete speaker**, as the screen **database**, disregarding the actual adjustment (see **the menu item Load database**), automatically assigns the database, the previous speaker was taken from.

5.2.1.1 The menu Find

Find: Enter the loudspeaker sign. You need only enter part of the information (for example only one letter). An example of this is that by entering the term **ISO** all loudspeakers, beginning with the first loudspeaker, which has a name starting with the ISO, will be shown. You can make your selection by using the left mouse key and clicking on **<OK>** or by selecting it with the cursor and using the return key.

Next, Previous, Fist, Last: These menu items allow you to leaf through a data record forwards or backwards or display the first or last record in the order of sorting of the database. **Note:** The first record in the sorting order has in general as much (minimal) chance of having the number one as the last record of having the highest number. This is caused by the kind of processing of databases, in which the indices (sorting criterion is the field **Speaker**) define the logical order. The physical record numbers are dependent on the order in which the record is input.

5.2.1.2 The menu Edit

Create: The screen contents will be deleted and the marker positioned on the first input field in the mask. Locate your desired data and after inputting use the **page down key** or the **return key** until you have passed the last input field.

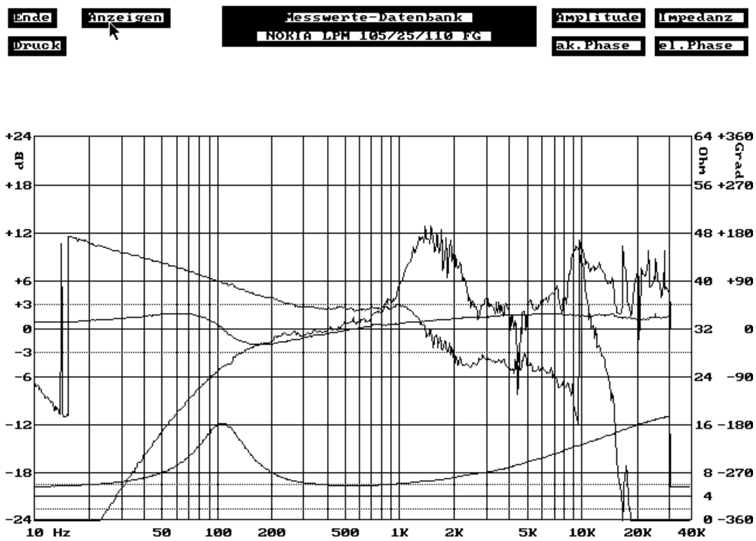
Edit: The contents of the screen still remains and the marker will be positioned in the first input field. With the help of the **cursor keys** move to the desired data field and change the value.

Save and Cancel: The newly retrieved or the changed values above have not yet been saved. You have to select **Save** or **Cancel** to decide what will happen to the data. At the beginning this procedure seems to be a bit complicated but essentially it supports data security. For example if 'a file drops on your key board', by selecting **Cancel** you can restore the old state without knowing what the actual content of a particular data field was.

Delete/Undelete: Through this function a delete mark will be installed or removed again. All marked records will be physically deleted by carrying out a database reorganization (menu item in the main menu). As long as the record is still displayed on the screen it can be undeleted.
After setting a delete mark you will be asked whether the measured data (Amplitude, Impedance, ...) for the loudspeaker should also be deleted with them. Answer this question with **<Yes>** if you are completely sure, as the undeleting of the measured data is not possible.

DAAS + DSA: Import Thiele/Small:
If you measured Thiele/Small-parameters using one of the audio measuring systems **DAAS 3.L**, **DAAS 3 NT** or **DSA** you can export them in ASCII-format and import in AudioCad with this menu item.

5.2.1.3 The menu Measuring-Data



Graph: The measured data of a loudspeaker will be graphically displayed. If no measured data have been saved with the loud- speaker then they will be simulated. Thus a minimum amount of Thiele/Small Parameters is necessary. In cases where there are one or more parameters missing the program will give a corresponding message.

Import: Here amplitude-, impedance- and phase- measured data which have been set with the loudspeaker measuring system or with the measuring data editor (see below), can be fed in. After selecting the menu item a file selection box with existing measured values will appear. Select one of these with the cursor key and Return or with the mouse. With amplitude measured data first the input file will be displayed in a selection window. Here choose an amplitude which corresponds with the 0 dB level (for example you can assume that with woofers they have reached their 0 dB level at about 150 Hz at the latest). Mark the desired value with the mouse or the cursor keys and click **<OK>** or use the return key. If you want you can change the value again in the input box which will now appear. If you have already used this function for a driver the program will first delete all "old" measured values for this driver and measured value type. After that the new data will be read. Repeat this procedure for all measured value types (amplitude, impedance and electric as well as acoustic phase.)

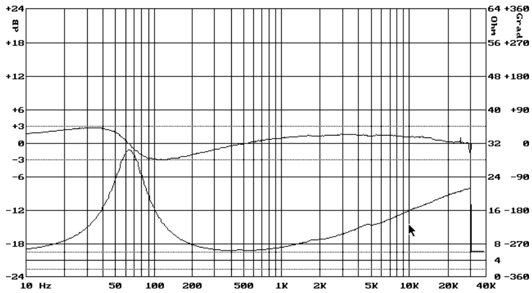
Hint: If you want to read all the measured values of a loudspeaker at once they must have the same file names (e.g. TEST.AMP, TEST.IMP, TEST.PHE, TEST.PHA)

Export: Here the amplitude, impedance- and phase measured data which are saved in the measured database (ACMESS.DB) can be exported in the AudioCad data format. Using this function they can be, for example, edited with the measured data editor and then again read into AudioCad Pro.

2 Files -> 1 File:
Use this menu item to make one output file out of two input files measured with different sampling rates. The results are exact measurings in the bass-range (e.g.: 20 Hz - 2 Khz) and by the second measuring (e.g.: 20 Hz - 20 Khz) nevertheless the complete frequency response. One exports the two measurings in the AudioCad format. The use of the menu item **2 Files -> 1 File** results in one File, containig the complete frequency response. The complete frequency response now can be imported into the AudioCad measuring database. This procedure is applicable for all audio measuring system basing FFT (DAAS, DAAS 3L, DSA, MLSSA, IMP, ...).

5.2.1.4 The menu Utility

Calculate parameters: Various parameters will be checked. Thus you can test whether the Thiele/Small parameters of the loudspeaker are plausible. This function depends on the number of your input data. You can tell from the corresponding message which values have been checked. This message will appear on your screen for six seconds. Those of you who are impatient can shorten the waiting time by pressing the space bar. If you want to save the calculated data in the database select the menu item **Save** in The menu Edit.



Calculate Le: This menu item calculates the moving coil inductance Le, presupposing that you have already saved the impedance path of the driver in the measured value database and direct current resistance **Re** in the database. If one of the values is missing an error message will be given and Le will retain the old value. After calculating Le you can save the value in the database by selecting the menu item **Save** in The menu Edit of the database. You should give in the desired crossover network cut off frequency as the frequency for calculating Le, by clicking on the right mouse button.

Calculate Q factors: This menu item determines fs, Qms, Qes and Qts, presupposing that you have already saved the impedance path of the driver in the measured value database and the direct current resistance Re in the database. If one of the values is missing an error message will be given and the calculation will be stopped. After the calculation you can save the values in the database by choosing the menu item **Save** in The menu Edit. Because of the logarithmic frequency raster used in AudioCad this function is mostly suitable for woofers and midrange speakers, the logarithmic frequency raster is too large for tweeters. If you save the impedance curve for a driver in the free field and in the cabinet you can determine the comparative volume with the function **Vas calculation** (in the main menu **Utility**).

Calculate Compound speaker: A possible way to reduce the required enclosure size of a speaker by about 50% is to use two woofers in a so called compound case. This can be done by using two drivers linked by a volume of air to create a new driver with different parameters. The essential difference between a compound driver and a single driver is the reduction, to about half, of the equivalent air volume Vas. Thus the case can be constructed half as big as when using a single driver. In addition you of course have the coupled volume for the two loudspeakers. This is the reason why it is only worth the trouble for considerably large speakers or loudspeakers with a large Vas. Apart from being expensive (4 instead of 2 loudspeakers are required) there is also the disadvantage of having about a 3 dB reduced reference degree of effectiveness. That is the price you pay for the appealingly small box. The reduced degree of efficiency caused by the compound construction has to be accounted for by the selection of the mid-range speakers and the tweeters or by the alterable switch circuit selection. Here you can either chose mid-range speakers or tweeters with a small degree of efficiency or you correct the volume via the voltage dividers in the crossover network. The program will calculate only compound speakers made out of two identical drivers.
Select the menu item **Calculate Compound Speaker** and enter the coupled volume in litres. By selecting a relatively large coupled volume you can lower the resonance frequency. However by doing so the degree of efficiency (SPL) will drop lower than through selecting a small coupled volume. The program will create a new record for the compound driver in the database, will simulate the new measured data and will save these into the measured value database. For a single loudspeaker up to nine compound drivers with differing coupled volumes can be calculated. If you have to cross this border you must first delete a compound driver which is no longer being used.

5.2.1.5 The menu End

Initialising loudspeaker data for calculation: Select the driver type (Woofer, Mid, ...) in the main menu as described above and then call up the loudspeaker database. Select the desired driver through the menu item **Find** in the **Find menu** and leave the mask by clicking on the menu item **End**. The program will now ask you whether the data should be used for the calculation. Confirm the question with **<Yes>.** The program then retrieves all the necessary loudspeaker data, for the construction, from the database and leaves the mask. Repeat the process for the other driver types. If no measured data for a driver are saved in the database it will be simulated, provided that the necessary Thiele/Small parameters are available.

Calculating a double bass system: Wanting slim speakers often conflicts with the usage of a large bass driver. If you insist on achieving a high sound pressure then often this will only be made possible through the use of two smaller bass drivers rather than one large one. The bass speakers will be connected in parallel. Only double bass systems with identical drivers will be calculated. If you want to use two basses connected in parallel then selected menu item **<No>.**

Tip for the impedance simulation:

Maximum impedance with basic resonance: The impedance simulation of the driver base resonance will happen with sufficient precision. In this case, with the correct presupposed Thiele/Small parameters, there will be an undetectable difference to the measured data.

Inductive impedance increase: The problem here with the simulation is, that for this, the moving coil inductance is the basis. The inductance of a real driver is dependent on the frequency. The **Le** given by the loudspeaker producer is measured in most cases at 20 kHz. If the impedance simulation is checked with an Le measured at 20 KHz you will find that the impedance at 20 kHz is correct. However it will increase improportionally towards higher frequencies. The results will be reciprocal if the **Le** is measured at 40 kHz. The impedance value will then be correct, however the impedance path below this frequency will be too flat. From the above it will be clear that the simulation of the impedance increase caused by the Le will, despite correct formulas, only be an emergency solution. (Other programs do not do this differently, however this problem will not often be mentioned in their manuals). Therefore if possible the real impedance path of the driver should be brought in the measured value database. Should this not be possible due to non-existent data the author recommends determining the inductance using the desired crossover network cut-off frequency and to put these into the loudspeaker database.

5.2.2 The menu item Load database

With this menu item you are in the position to load another database. Leaving the program, the last loaded databse will be stored in the configuration file. When starting the program again, the last loaded database will be initiated.

5.2.3 The menu item Create new database

This menu item creates a new, empty database. So you are able to keep the loudspeakers, measured by yourself, in a separate database. When keeping several databases, datas of other producers, like the HiFiSound-database, can be used without a time-consuming loading. To create a new database you have to feed in a file-name up to seven characters, without a point an without file-extension.

Example:

File-name: **Test**

The program creates the following files:

Name of loudspeaker-database	: **TEST.DBF**
Name of Index-file	: **TEST.NDX**
Name of measurement-data database	: **TESTM.DB**
Name of index-file	: **TESM.NDX**

Foloowing the above example the name of the measument-data - database wil be formed out of the loudspeaker-database file-name; but don't forget to add an **M**.

5.2.4 The menu item Delete database

This menu item deletes a databse from yout fixed disk, after having confirmed twice. The case being such, **the data is irrecoverably lost!**

5.2.5 The menu item Query database

This function allows an automatic search for loudspeakers with certain parameters. The concept of the loudspeaker database assumes that you do not look for a case for a certain loud- speaker, but that you have certain given targets (case size, lower frequency limit, etc.) according to which you choose a suitable loudspeaker.

The menu item Input: Only enter the values according to which you want to make your choice. For example if you want all the loudspeakers displayed with values between 0,3 and 0,4 Qts only enter these two values. The selection criteria are connected with the logically defined AND, i.e. by entering more selection criteria you continuously restrict the set of results. If you want to see all the records saved in the database simply use the default settings. If for example you wish to select all the loudspeakers with double moving coils, then enter the text **DOP** in the remark field.

By selecting the menu item **List** you start the output on the screen. You can print out the values by using the print key or by selecting **Print**.

The menu item Copy selected records to other database: By using this menu item you are able to copy the selected records (loudspeakers) into another database. In case it do not yet exist, you have first to create the new database with the menu item **Create database**.

5.2.6 The menu item Reorganize database

The program will physically remove all data records marked to be deleted from the database, frees the occupied memory on the hard disk and creates new index files. A reorganization can produce an increase in performance, if you have previously deleted many records (loudspeaker, measured values). This function will be helpful in cases of system malfunctions (e.g. Computer crash due to power cut during database processing), as by using it a defect database can be repaired in almost all cases.

5.2.7 The menu item Export

Here all amplitude-, impedance- and phase paths which have been saved by the program in the **database** or simulated in the **enclosure** part can be exported in AudioCad format. In order to work on these curves further they can be retrieved again into the database or for example be used as a target function for the crossover network optimisation. The curve paths can also be loaded into and changed in the measured data editor. In the crossover network mask there is a single menu item for the export of the crossover network simulation.

5.2.8 The menu item Exit

After an affirmative to the security check question you will leave the program and return to the main menu (**Acmenu.bat**).

5.3 The menu Enclosure

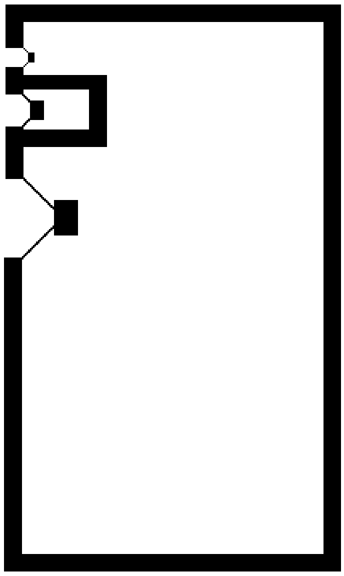
Here the determination of the fine tuning of the loudspeaker case will happen separately for each driver in the system. You can for example construct a bassreflex case for a woofer and a closed case for a mid-range speaker. Further more by using the **Cabinet Draw** function a case with up to 4 loudspeakers and bass reflex channels can be sketched. This menu item also calculates the wood dimensions (cuts).

For closed, bass reflex and bandpass filter cases there will be a simulation of the amplitude, impedance and phase behaviour of the drivers built into the case. The membrane excursion can also be graphically displayed.

Transmission line speakers, and horns can be calculated by the program, but first they must be built and measured with an audio measuring system in order to use the data in the crossover network simulation.

As already mentioned there is a possibility to leave out the case construction and to build a crossover network for a loudspeaker already installed in a case. To do this measure the loudspeakers installed in the case individually with an audio measuring system and enter the data into the measured value database. Then initialise the individual paths of the speaker as described in the section **Database** and move on directly to crossover network construction without detouring through the case simulation.

5.3.1 Closed boxes

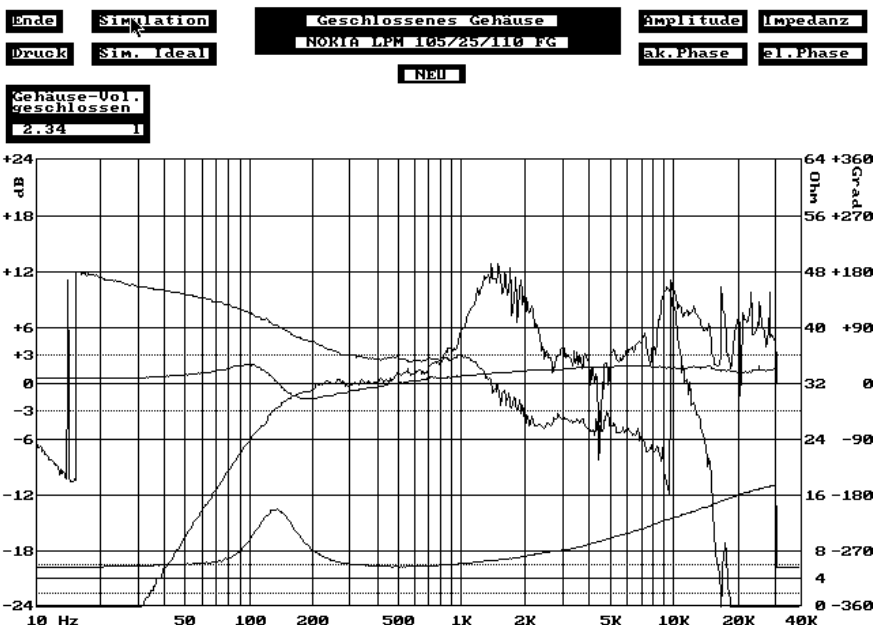


This part of the program calculates closed speakers according to Thiele-Small. These distinguish on having an ideal impulse behaviour when they have been correctly fine tuned. The bass comes over as being clearer and less fuzzier than with a bass reflex construction. A disadvantage however is that the low bass of a closed speaker is less characteristic than in the case of a bass reflex speaker.

The menu Edit

The menu item Input: The **serial resistance,** which consists of the internal resistance of the amplifier and the losses of loudspeaker cable and crossover network, should be taken into account in the construction as it will have an essential influence on the quality of the chassis. Enter the value in ohms.
To calculate the enclosure you can give as an initial either the system quality (Qtc), the case volume (Vb), or the resonance frequency (fc) or the other initial values are to be set to 0. If you do not value a Qtc of 0.707 (Butterworth-Fall) will be assumed. After leaving the input mask (e.g. by using the page down key) the output values will be calculated and displayed.

The menu item Graph: This function will show the amplitude, phase and impedance path for the driver installed in the calculated case. Concerning the further opportunities offered by this graphical display please read the **Program operation In Graphics**.



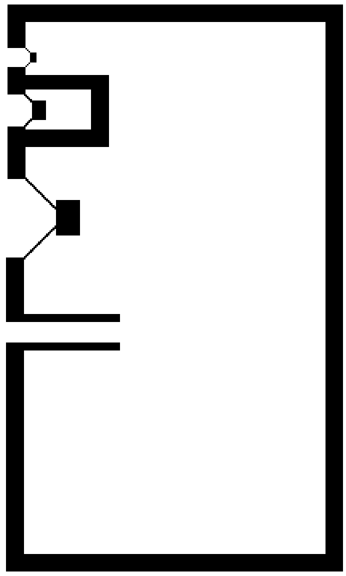
The menu Exit

The menu item Exit: After leaving the mask the values of the last case simulation will be used for the crossover network construction. Please make sure that you use **Simulation** and not **Sim.ideal** for the last simulation in the graphics, in order to obtain realistic values.

5.3.1.1 Influence of damping-material

Through the inclusion of damping material the case can be smaller than the dimensions calculated. You can obtain the best efficiency with 6 cm thick Pritex (knobbly foam material). If you completely cover the inner walls of the case with pritex then you can reckon with an increase in volume of between 10 and 15%. An increase of about 30% can be obtained by completely filling the case.

5.3.2 Vented boxes



The bass reflex displays the best compromise between enclosure size and feasible bass renderings. A bass reflex casement will, in general, be larger than a closed case for an identical driver, however the lowest possible frequency limit (f3) is considerably lower than that of a closed case.

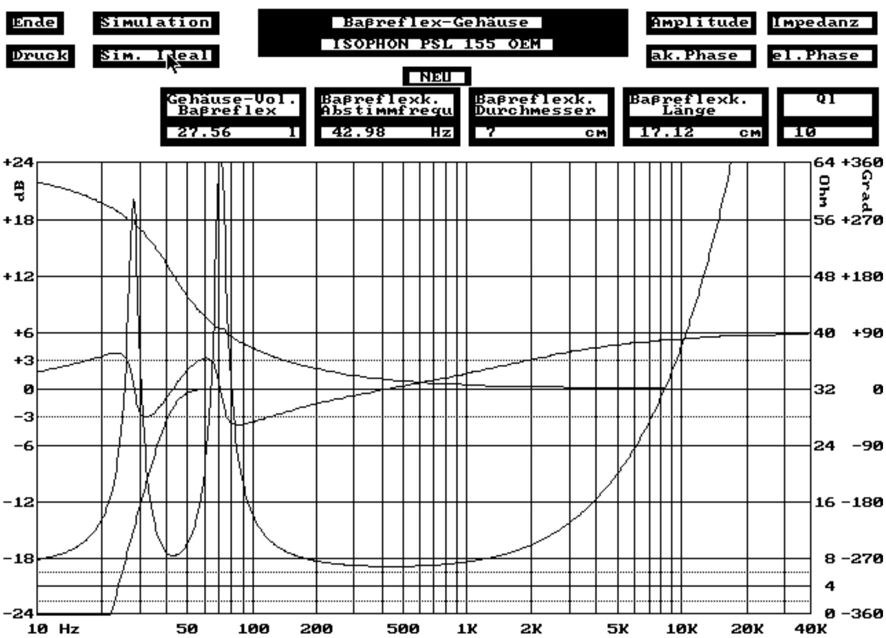
The menu Edit:

The menu item Input: Here the necessary initial values for the construction will be entered. The **Serial resistance** which consists of the internal resistance of the amplifier and the losses of loudspeaker cable and crossover network (-coils), should be taken into account in the construction as it will have an essential influence on the quality of the chassis. Please enter the value in ohm.

As the next value enter the **quality loss** (Ql). This value depends on the case volume. For a case up to 30 litres choose 10, up to 70 litres choose 7 and over 70 litres choose 5. Just try it with 10. The program will adjust the Ql value automatically according to the case size.

The bass reflex channel diameter can be defined by the user. By inputting **0** the surface of the bass reflex channel can be entered if a rectangular channel is to be built. When no figures are entered the program will assume a round channel with a diameter of 7 cms. After leaving the input mask (e.g. page down key) the output values will be calculated and displayed.

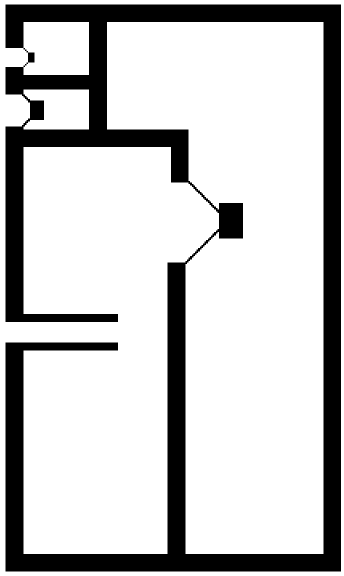
The menu item Graph: This function will show the amplitude, phase and impedance path for the driver installed in the calculated case. Concerning the further possibilities offered by this graphical representation please read the **program operation in graphics**.



The menu Exit:

The menu item Exit: After leaving the mask the values of the last casement simulation will be used for the crossover network construction. Please make sure that you use **Simulation** and not **Sim.ideal** for the last simulation in the graphics, in order to obtain realistic values.

5.3.3 Band pass filter case, one-side ventilation (Band pass 1)



The band-pass filter case is mostly suited for usage as a subwoofer with a lower cut-off frequency. The construction comprises a steep mechanical low pass (the bass reflex channel), which makes the construction of simple alterable switch circuits possible. Theoretically you could do without the electrical low pass (e.g. coil). In practice however you should connect at least a 6 dB low pass (a coil) in front of the band pass filter sub woofer, through which the amplitude and the impedance restraint of the box in the mid-range and tweeter area can be improved.

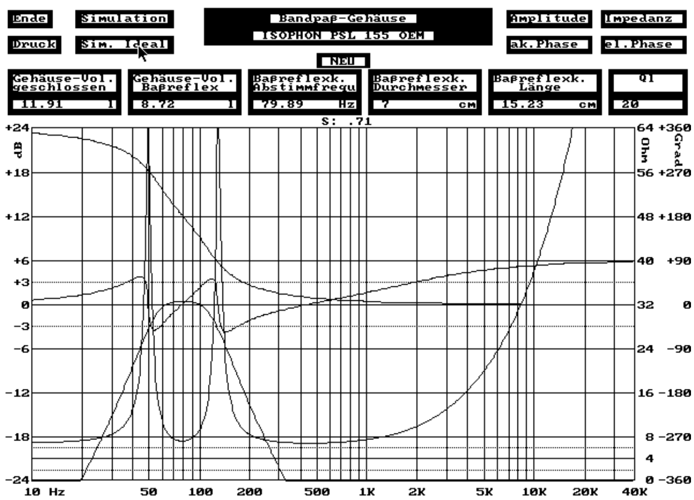
The menu Edit
The menu item Input: Here the necessary default parameters for the construction will be given in. The **Serial resistance** which consists of the internal resistance of the amplifier and the losses of loudspeaker cable and crossover network (-coils), should be taken into account in the construction as it will have an essential influence on the quality of the chassis. Please enter the value in ohm.

The next value to be entered is that of the quality of the closed case part. If nothing is entered here the value 0.707 will be allocated.

Then enter the value for the quality of the ventilated case parts. Again the value 0.707 will be used if nothing is entered here.

The bass reflex channel diameter can be defined by the user. By inputting **0** the surface of the bass reflex channel can be entered if a rectangular channel is to be built. When no figures are entered the program will assume a round channel with a diameter of 7 cms. After leaving the input mask (e.g. page down key) the output values will be calculated and displayed.

The menu item Graph: This function will show the amplitude, phase and impedance path for the driver installed in the calculated case. Concerning the further possibilities offered by this graphical representation please read the **program operation in graphics**.

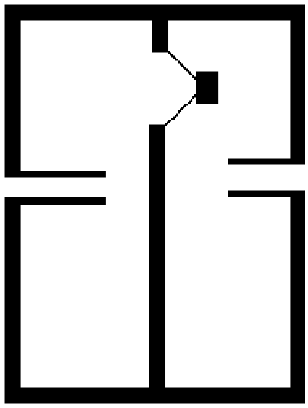


The simulation will be carried out for symmetric band pass filter cases. Here the resonance frequency of the bass reflex channel (fb) will be attuned to the resonance frequency of the closed case. As very large phase angles occur it can happen that phase jumps in the diagram arise when there are large errors in tuning the brc-resonance frequency. This is due to the principle and is not a program fault.

The menu Exit:

The menu item Exit: After leaving the mask the values of the last casement simulation will be used for the crossover network construction. Please make sure that you use **Simulation** and not **Sim.ideal** for the last simulation in the graphics, in order to obtain realistic values.

5.3.4 Band pass filter case, two-side ventilation (Band pass 2)



The band pass filter case with double sided ventilation is especially suitable for use in car hi-fi-systems as a subwoofer with lower cut-off freqeuncy.. Apart from horn speakers this type of case has the best degree of efficiency in the woofer range. The disadvantage will however be generally inferior dynamic behaviour of the system compared to bass reflex systems or band pass filters with one sided ventilation. Despite the band pass filter characteristic you should connect at least a 6 dB low pass filter in front of the subwoofer (a coil), in order to counteract the changes of the mid-tone range.

Using AudioCad to tune a band pass filter with double sided ventilation is only empirically possible. Calculation of the case according to specific filter characteristics is not yet practical due to the complexity of the procedure on a PC because of insufficient calculation power. Tuning according to the rule of thumb:

Case 1: Conventional bass reflex tuning
Case 2: Half case volume of 1., double vent tuning frequency of 1

This method produces good results and can be optimised in a short time with the help of graphic simulation by trial and error. However beware of

- too large deviations in the case volumes
- too large deviations in the resonance frequencies of both chambers
- approximately identical case volumes
- approximately identical resonance frequencies of both chambers and
- too large bandwidths of the system (2 to at most 3 octaves should be sufficient!).

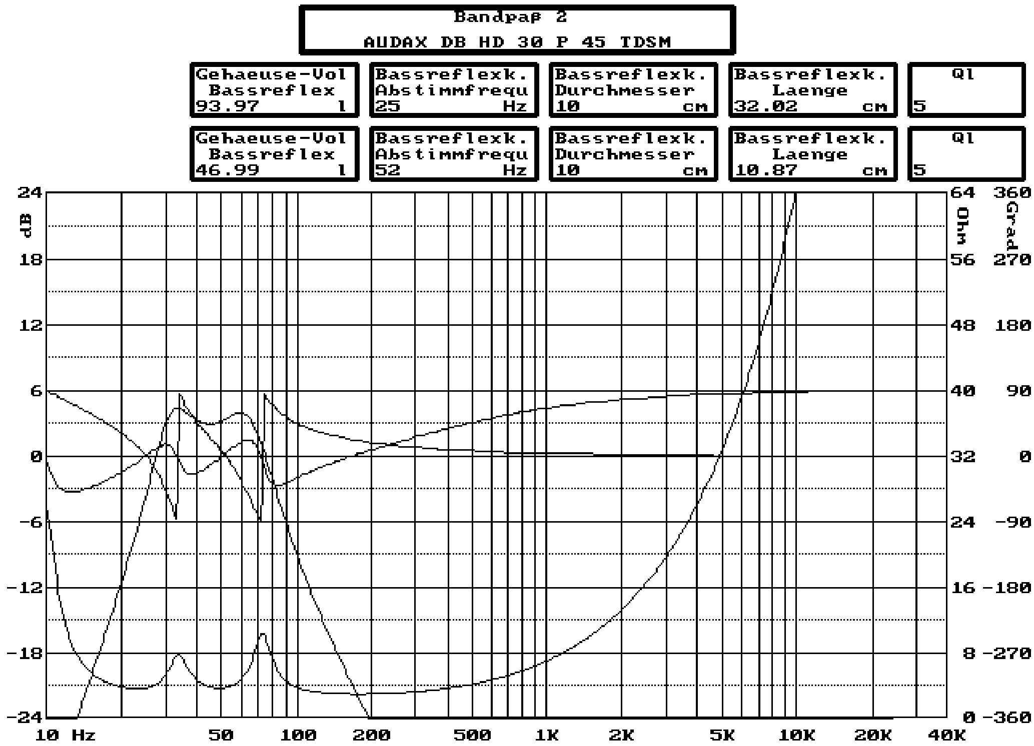
Uncontrolled oscillation of the whole system could result. These influences are not contained in the model which is the basis of the simulation, they however are extraordinarily efficient!

The menu Edit

The menu item Input: Here necessary pre-allocations for the construction will be done. The **Serial resistance** which consists of the internal resistance of the amplifier and the losses through the loudspeaker cables and crossover networks (coils) should definitely be taken into account, as it has a considerable influence on the quality of the chassis. Enter the value in ohms.

The diameter of the two the bass reflex channels can be pre-allocated. By inputting 0 you can enter the area of the respective bass reflex channel, if you wish to construct a rectangular channel. If you leave out all inputs the program will assume round channels with a 10cm diameter. After leaving the input mask (possibly by hiting the **page-down** key) the output values will be calculated and displayed.

The menu item Graph: This function will show the amplitude, phase and impedance path for the driver installed in the calculated case. Concerning the further possibilities offered by this graphical representation please read the **program operation in graphics**.



The End menu:

The menu item End: After leaving the mask the values of the last casement simulation will be used for the crossover network construction. Please make sure that you use **Simulation** and not **Sim.ideal** for the last simulation in the graphics, in order to obtain realistic values.

Practical case construction:

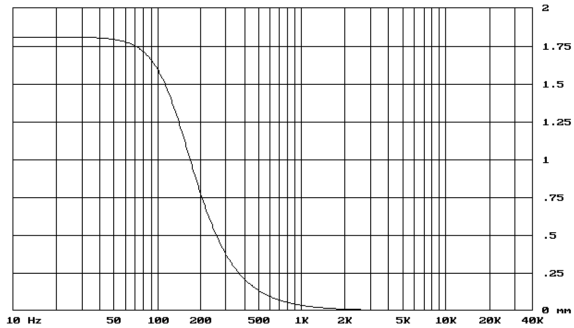
As the two bass reflex channels emit the sound at 180 degrees phase turns, they should terminate at the opposite walls of the case, as shown in the sketch above. By doing this you will get an equal phase emission.

5.3.5 The menu item Membrane stroke

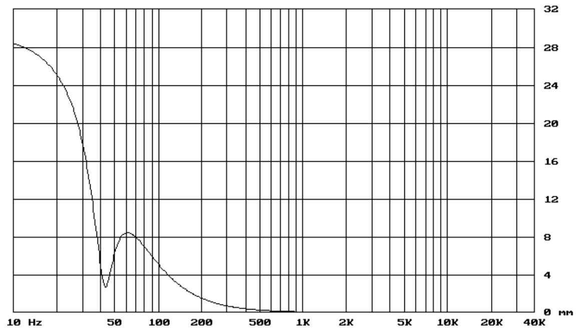
The menu item **Membrane stroke:** will graphically display the membrane deflection (available for closed, bassreflex an onde-side ventilated bandpass boxes) after inputting the electrical power supplied for the connected amplifier. The scale will automatically fit to the respective curve, if you have allocated other instructions in the installation mask. Do not only observe the curve path but also pay attention to the maximum value of the y-axis (membrane excursion in mm).

Tip for the simulation of mid-range speakers: A load of 50, 100 or more watts is often given for a mid-range speaker. In fact a mid-range speaker can only cope with such a power through a crossover network connected before it, i.e. it will only receive a few watts. If you enter 100 watts as the initial power then you will naturally get completely irrelevant values in the membrane excursion simulation. A reasonable value is about 15 watts for a mid-range speaker.

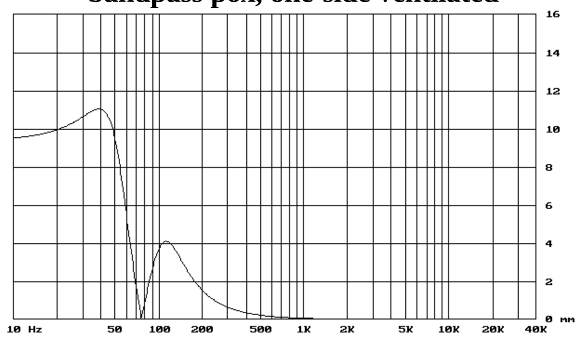
closed box



bass reflex box



bandpass pox, one-side ventilated

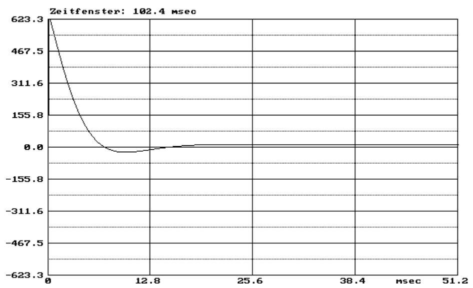


5.3.6 The inverse fast Fourier transformation

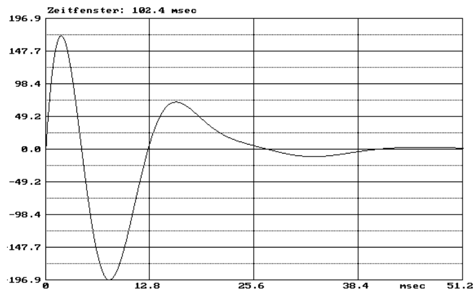
The fast Fourier transformation, which will be hereafter denoted as FFT, is a very precise mathematical method for the transformation of the behaviour of the loudspeaker from time range to frequency range. This means that the impulse behaviour can be converted into a frequency path. In AudioCad the inverse FFT (transformation from frequency to time range) will be used with 1024 support points to represent the dynamic behaviour of drivers in the case. As the 1024 points are not representable with the corresponding resolution on a standard VGA monitor only the 512 points will be displayed. However all important effects will occur within this range so this display is fully satisfactory and efficient. The size of the frequency steps can be varied between 1 and 20 Hz. The most informative display is obtained between 5 and 10 Hz.

In order for you to get a feeling for a "good" or a "bad" dynamic behaviour some examples are provided here. As it is already known closed speakers have the best impulse behaviour. As a comparison the behaviour of bassreflex cases and band pass filters with one- and two-side ventilation will be shown. All the simulations were carried out with a Butterworth tuning (Qtc 0.707) with the same driver and time period.

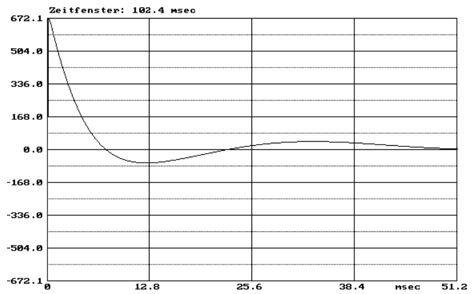
Closed box



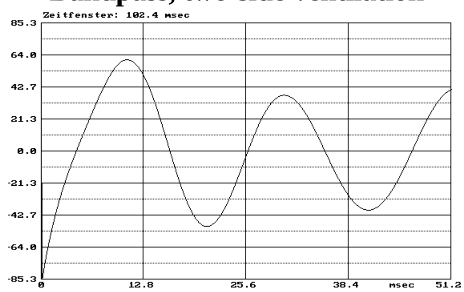
Bandpass, onde-side ventilation



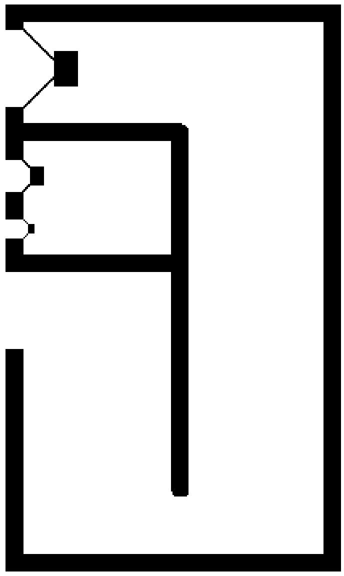
Bassreflex case



Bandpass, two-side ventilation



5.3.7 Transmission line boxes



The transmission line case (TML) is the biggest challenge for a speaker builder. With this kind of enclosure it is possible to get a -3 dB point down to the free air-resonance frequency (fs) of the woofer. However this will result in larger effort for the case building and especially for the damping. But let us just start with the selection of the drivers. The scholars are undecided with regard to the optimal Qts value. Most of them are of the opinion that an optimal TML-driver should have a Qts > 0.9. However this should not be taken too seriously. The most important selection criteria are a large membrane plane, the largest possible membrane hub and a low resonance frequency (fs).

The menu Edit

The menu item Input: Input the inner width of the cabinet. This will be taken into account when the calculation of the line-dimensions is done.

Program outputs:

Line opening cross-section (LS-Line): The cross-section of the line at the location where the loudspeaker is built in (line begin); if you design the line as a square you can take the "line dimensions" as a basis.

Line opening cross-section (to the outside): The cross-section of the line at the speaker wall (the exit opening); if you design the line as a square you can take the "line dimensions" as a basis.

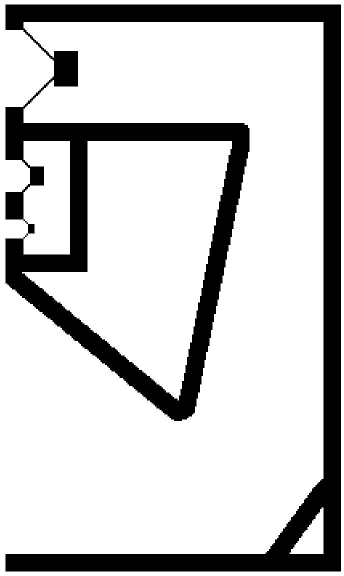
Internal cabinet volume: The net inner volume represents the line inner volume. It is only meant to be used as a scale to estimate the size of the case. In order to predefine the real size of the case of course you have to add the volumes of the chip boards of the inner and outer walls.

Line length: The length of the folded line from the loudspeaker to the line opening surface on the outside of the case. If you want to build an unfolded line add 15cm to the line length.

Damping: The damping material should have a certain density of minimum 30kg/m3. You can use for example pritex (knobbly foam stuff 50 mm), stone wool or long fibre natural wool (also called Bailey wool). Polyester wadding (BAF-Wadding) and glass wool should not be used in the line! Their damping qualities are not good enough by far. The line should be damped with about 6kg damping material per cubic metre (1000 litres). The material has to be prevented from moving about by appropriate means, so that it will not gather at the bottom of the case.

Tip for the position of the line opening: The line opening for large cabinets should in any case run into the sound wall (facing forward). If the line opening runs into the upper side of the case this will generally lead to problems in normal living rooms with regards to the bass transmission as the distance to the next wall (ceiling) is too small for the transmission of the lowest frequencies.

5.3.8 Horn loudspeakers



Why horn loudspeakers? Horn loudspeakers with correct dimensions have the following advantages:

-High degree of efficiency: up to 50% in contrast to 1-5% for closed or bass reflex speakers.
-Excellent impulse reproduction
-Small membrane excursions
-No compression of dynamics
-Small harmonic distortion factor

Considering these advantages, why are there only a very few horn combinations in the ready made or self made speakers market? To answer this question we will go into details on the basics of horn loudspeakers. The trick in the construction of horn speakers is the increase of radiation resistance which the loudspeaker sees. With a direct radiating loudspeaker the air in front of the membrane can swerve to the side. Due to this the degree of efficiency is extremely lowered as this air is not radiated by the driver membrane in the direction of the listener. For a better explanation we now assume that a loudspeaker is not built into a case. Such a loudspeaker chassis can only create reasonable reproduction up to a frequency area which has a wave length corresponding to the membrane circumference. This frequency can be calculated using the following formula:

Lower limit frequency = sound speed/membrane circumference

where the sound speed = 343m/sec at 20 degrees Celsius and the membrane circumference is entered in metres. With a 300mm woofer this formula will result in a lower limit frequency of 364Hz. The idea behind the construction of horns is now the virtual increase of the membrane area of the loudspeaker, to the size required for the lower limit frequency. After reshaping the above formula the result gives the required horn opening area (AM) for the desired lower limit frequency, which will be refered to as horn mouth. If you calculate this for a horn with a lower limit frequency of 30 Hz, the result will be a diameter of 364cm for a round horn. This equals 10.4 m2. And there lies the root of the problem, these dimensions would never fit into a normal living room! Nevertheless there are possibilities to decrease the size of the horn mouth. The horn opening area can be reduced by half for every adjoining part of the hearing room. This means in practice the following:

- Installation on the floor	: horn mouth / 2
- Installation on the floor and on one wall	: horn mouth / 4
- Installation in a corner of a room	: horn mouth / 8

If installed in a corner you will get an opening area of 1.3 m2 for a 30 Hz horn. This corresponds to a diameter of 64 cm. This opening area can just be placed in a large living room, which makes the whole thing interesting again. This simple calculation example on the other hand makes clear that you cannot place a horn system just anywhere! Already during the development the installation of the completed speaker will be fixed. An incorrect installation can therefore only lead to poor results. But at this point the author does not want to conceal the biggest disadvantage apart from the need for space (as enough advantages have been mentioned). The practical building of horns turns out to be extraordinarily difficult. The existing program covers all the required calculation work for you but the construction and building of a case with corresponding horn contours can not be carried out for you by even the most powerful program. It is perhaps still possible with exponential horns but with Tractrix (ball wave-) horns it is close to being purely unattainable for private users. The only reasonble way that the author can see to produce Tractrix horns is, for example, to build a form out of wood or plaster and to laminatate the horns by using glass fibre.

5.3.8.1 Horn

With this part of the program it can be decided whether a speaker is suited for the use of a horn. Moreover, the relevant chamber and fore-chamber volumes will be calculated. To that let's next have a look at the sketch of a horn.

At this point the author does not want to go further into the theoretical foundations of the horn loudspeakers, instead he will, in the following, concentrate on describing the program..

Output data: The calculation will be carried out even when not all input data are available. In this case it is of course not possible to calculate different output values.

If the desired horn is defined by the specifications

-Chamber volume,
-Fore-chamber volume
-Neck area
-Horn length
-Mouth area

you can start to determine the horn contour with the help of the next part of the program. The determination of the length of the horn will only be executed for exponential horns. The horn length for tractrix horns can be seen in the program part Horn Calculation.

5.3.8.2 Horn calculation

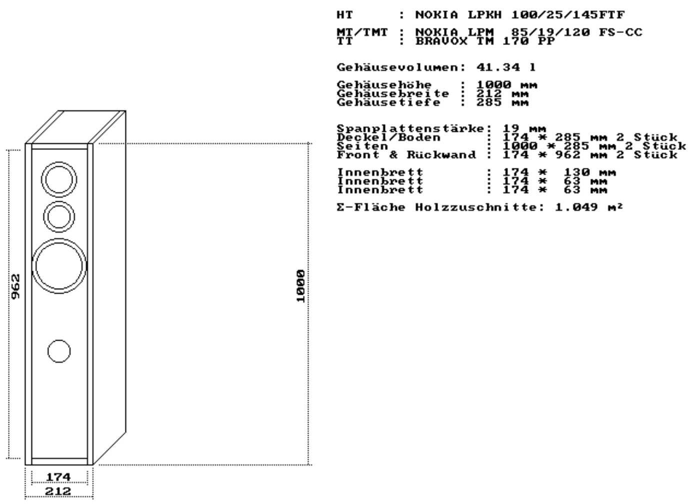
With this part of the program you can determine the contour of exponential or tractrix horns. As in general many single values are required for this, the output will only be executed on your printer. Most of the input data will be taken from the mask **Horn**, if you worked on this mask immediately beforehand.

The menu item Input: Most input data will be taken from the mask Horn Driver Selection.

The menu item Print: The calculation and output of data on your printer will be started when selecting this item.

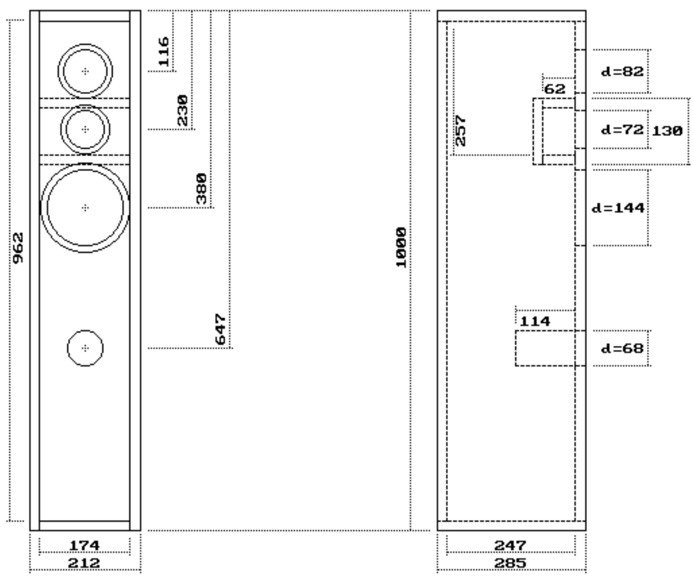
The menu item Horn Type: Here you should select the type of horn to be calculated (Exponential or Tractrix).

5.3.9 The menu item Cabinet draw



This part of the program displays a box including loudspeakers and bass reflex channel on the screen and will output the chipboard dimensions. Thus you can have a preview of how the speaker will look like in proportion. The data of the loudspeaker (diameter of the chassis) will be automatically retrieved from the database. The size of the case results from the sum of the volumes of the previously constructed cases. The virtual volume-increasing by dams material in closed systems is taken into account.

The menu Edit



The menu item Input: Through this menu item you come to the mask and here you can enter the required information for all sketches. If you do not want to go through all the input fields you leave the mask by using the page down key. The sketch will immediately be displayed.

You can feed in the external hight and external width. The depth is calculated.

Board-thickness): Here enter the strength of the chip board (19mm, 20mm, ...). This value will be taken from the installation mask.

Das in der Maske für jeden einzelnen Lautsprecher angegebene **Gehäusevolumen** enthält das jeweils zugehörige Zusatzvolumen.

The additional volume calculates himself from the volume of the driver and the vent. These volumes were estimated by the program. You also can calculate manually exactly and feed in this values.

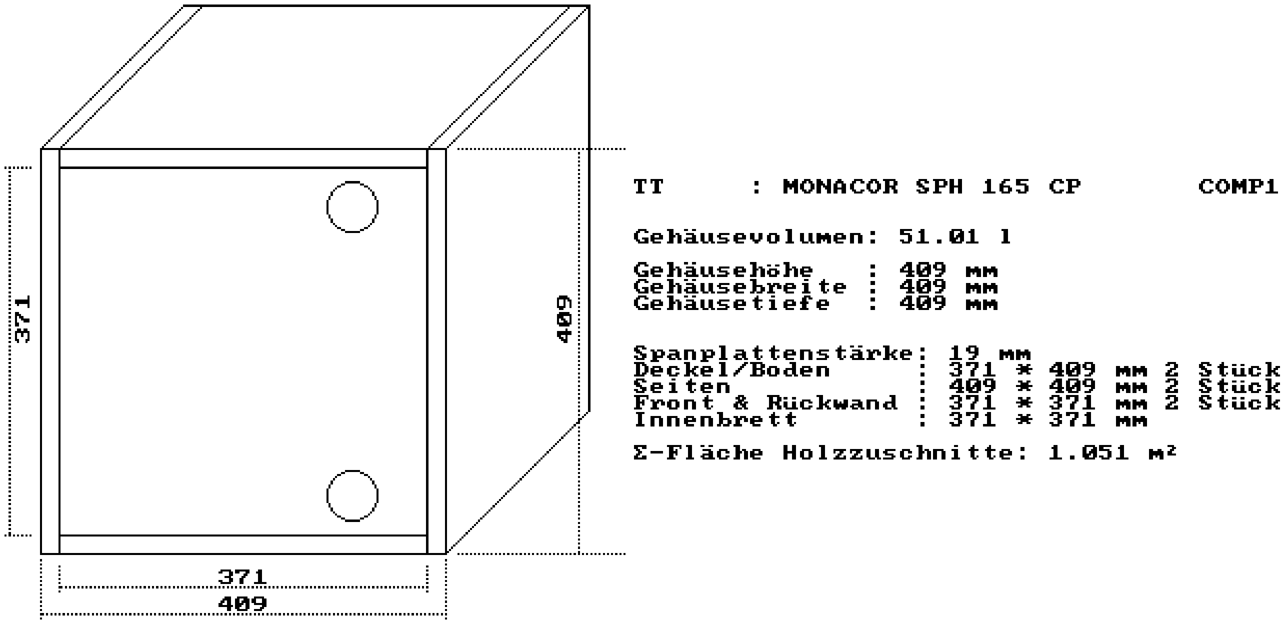
Distances: The distances of the tweeter from the upper case (inner) edge and of the rest of the loudspeaker from the upper driver can be entered. All the data is in millimetres.

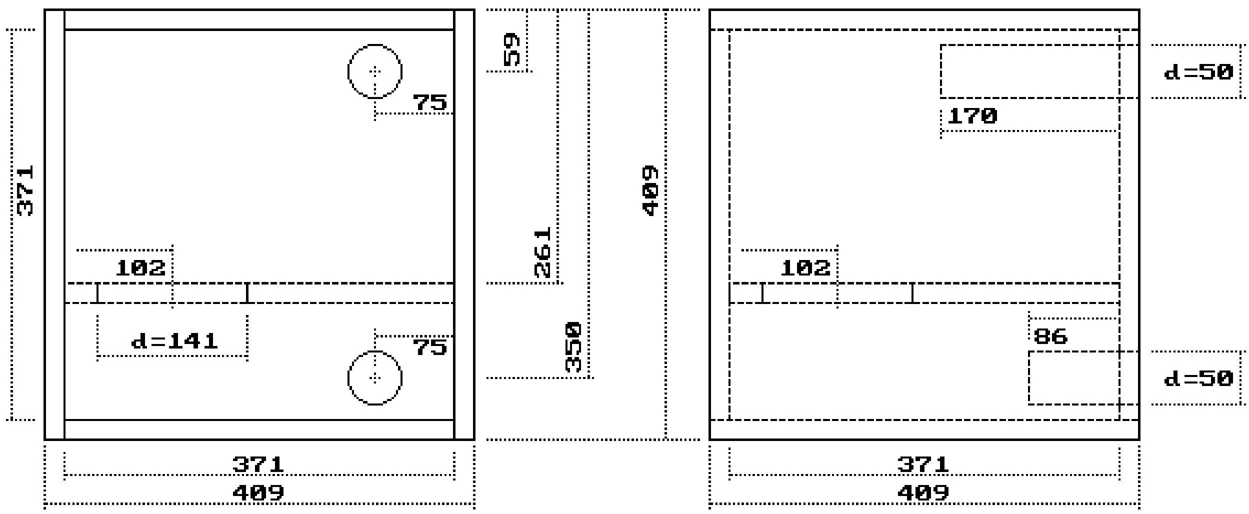
You can type in **R** for round or **Q** for rectangular as vent form.

Take into account for the making of the enclosure that the program distributes the inside diameter of the vent. The hole for the vent is therefore, depending on the used vent, always larger as indicated in the graphics.

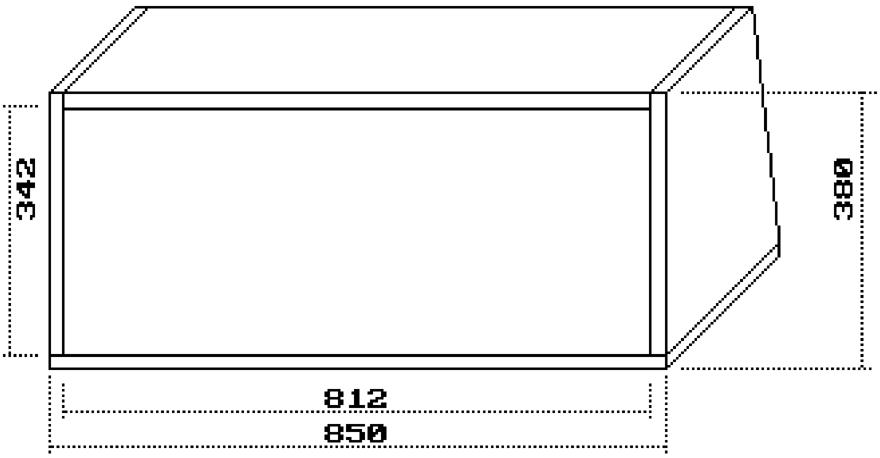
The menu-item Plot speakers:
Here you can switch the plot of drivers on or off.

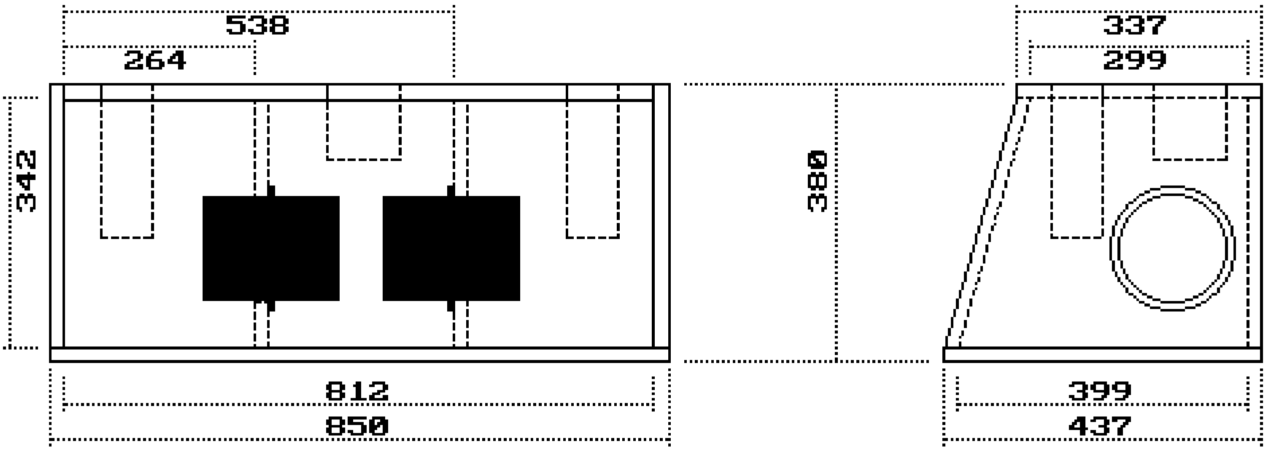
The **cabinet-draw bandpass** is served correspondingly just the same like the cabinet-draw of closed and vented boxes.





The **Cabinet-draw Car-Subwoofer** works like above described.





5.4 The menu Crossover

It is possible to calculate crossover networks from the first to forth order (6 to 24 dB) with the Butterworth, Bessel, Tschebyscheff, Linkwitz and compromise characteristics. Differences in the degree of efficiency between single drivers can be equalled out by the usage of voltage dividers. The impedance paths of drivers can be equailzed by using RC- and RCL links. The value of each single constructional element can be changed in the graphic presentation of the crossover network circuit, so that nearly any circuit can be simulated. Comfortable optimisation routines complete the variety of functions.

Note concerning the message Swapping: Through calling the crossover network mask the program temporarily stores date which is not needed for the network construction on the hard disk. After leaving the simulation the program will retrieve this data. This is called **Swapping**.

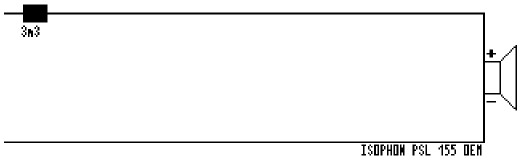
5.4.1 The menu Edit

Here the initial tuning of the network will be calculated. It is the basis for the fine tuning of the simulation and optimisation part.

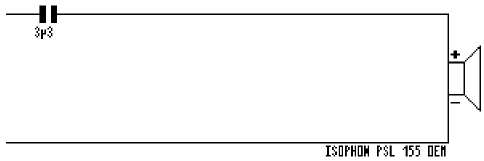
5.4.1.1 The menu item Cut off Frequency

After selecting **Cut off Frequency** the program will ask through a selection box about the driver to be worked on. Select woofer for example. Now the cursor is in the input field for the cut off frequency of the low pass for the woofer. Enter the desired frequency. Next select the slew rate (6, 12, 18 or 24 dB) in the selection box which appears. Specify the network characteristic (Butterworth, Bessel, Tschebyscheff, Linkwitz or Compromise). Repeat this procedure for all speakers. If you want only to change the cut off frequency for a driver hit the page down key. Then you will not have to enter the slew rate and filter characteristic again. In the following some basic crossover network circuits will be sketched.

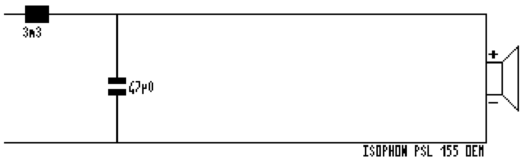
6 dB low pass



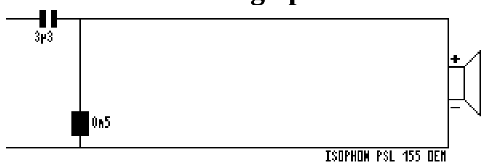
6 dB high pass



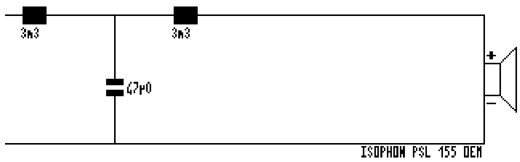
12 dB low pass



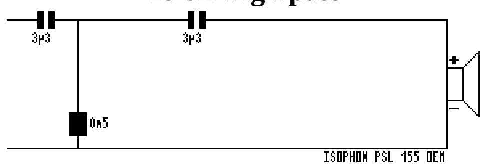
12 dB high pass



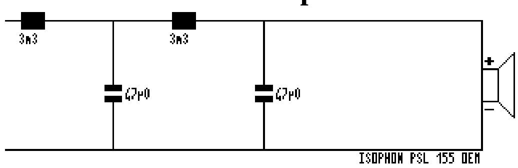
18 dB low pass



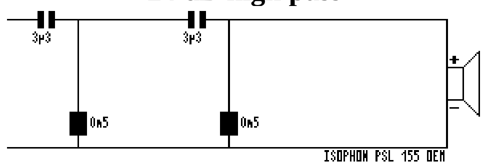
18 dB high pass



24 dB low pass

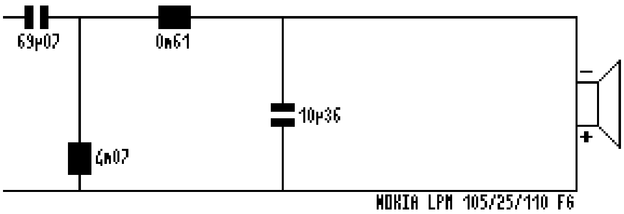


24 dB high pass

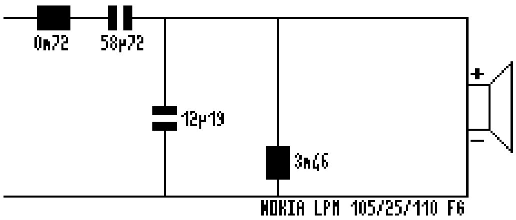


For a mid-range speaker a high and low pass will be connected in series or a band pass layout is used.

12 dB - high- and low pass



12 dB- bandpass



5.4.1.2 The menu item Impedance Emphasis & Attenuation

For the impedance equalization the following data has to be entered in the loudspeaker database for the respective driver:

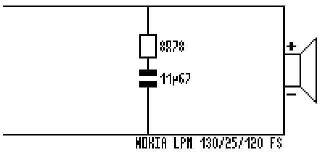
Direct current resistance (Re)
Resonance frequency (fs)
Mechanical quality (Qms)
Electrical quality (Qes) and
Moving coil inductance (Le)

If only the impedance and electrical phase curve are available and not the above mentioned Thiele/Small parameters, then the equalization can be empirically determined through the impedance optimization. In order to do this enter the desired equalization link (RC- or RCL-link) in the crossover network circuit, allocate the standard values to it (e.g. RCL-link,: 10 Ohm, 100 Micro-Farad, 50 MiliHenry, RC-link: 10 Ohm, 50 Micro-Farad) and optimize the impedance path in the desired area.

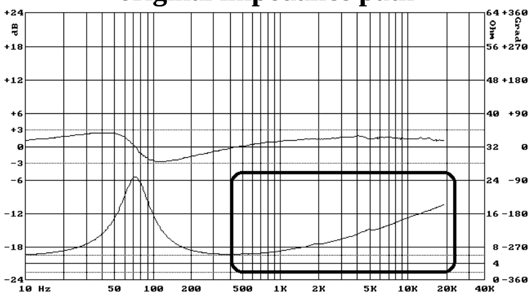
After selecting the menu item the input box will appear. In the upper area of the input box choose the desired crossover network branch. In the lower area chose the kind(s) of equalization.

RC-link

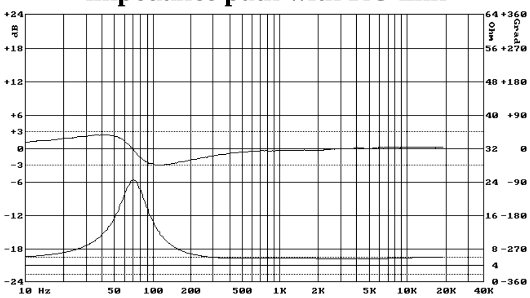
An RC-link will be calculated for the correction of the impedance rise caused by the moving coil inductance towards higher frequencies. For this calculation the direct current resistance (Re) and the moving coil inductance (Le) is required.



original impedance path



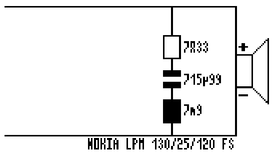
impedance path with RC-link



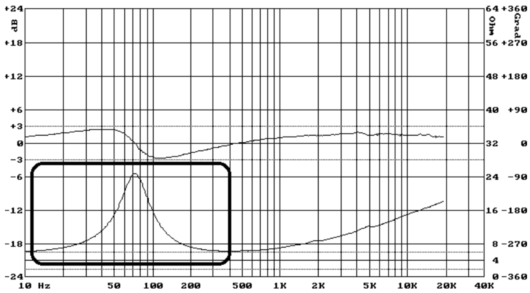
RC-links will usually only be used for woofers and mid-range speakers. The equalization of the inductive impedance rise for tweeters can in general be omitted.

RCL-link

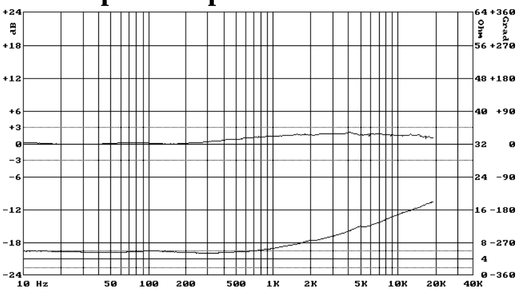
An RCL-link will be calculated for the correction of the impedance rise caused by the basic resonance, and the new impedance path will be shown. The RCL link will be calculated from the corner frequencies (f1, f0 and f2) according to Thiele/Small parameters. If this is not possible because of inconsistent data (impedance path not entered correctly) the program will determine the equalization link with the help of Thiele/Small parameters. The results produced by this however are in general less accurate.



original impedance path



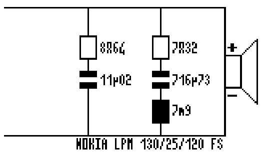
impedance path with RCL-link



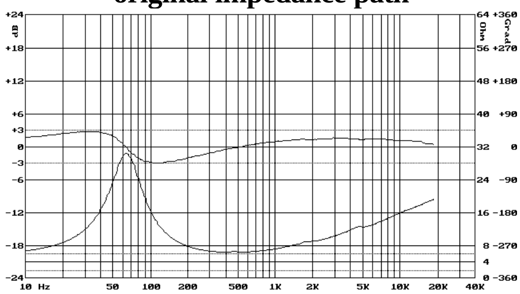
RCL-links will be implemented for mid-range speakers and tweeters. The basic resonance of the woofer has in general not to be equalized.

RC and RCL link

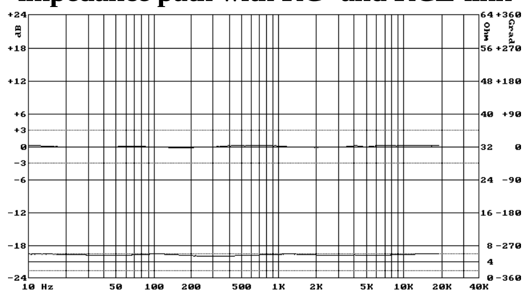
By the implementation of both equalization links complete linearisation of the impedance path is possible.



original impedance path



impedance path with RC- and RCL-link

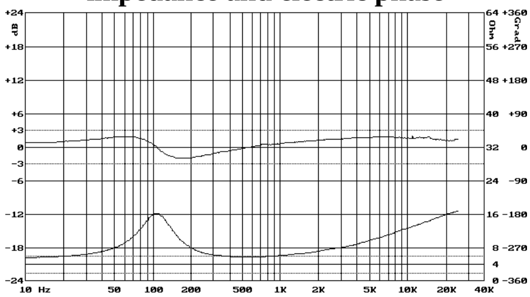


Attenuation

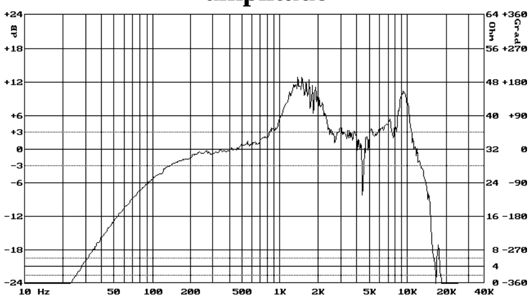
A mid-range speaker or tweeter which is too loud in comparison to the bass driver can be depressed in amplitude by the implementation of a voltage divider. To do this click on **[ ] Attenuation** and enter the desired level reduction in dB in the input box -dB. By using a voltage divider you can in general omit further impedance correction links.

Original paths

impedance and electric phase

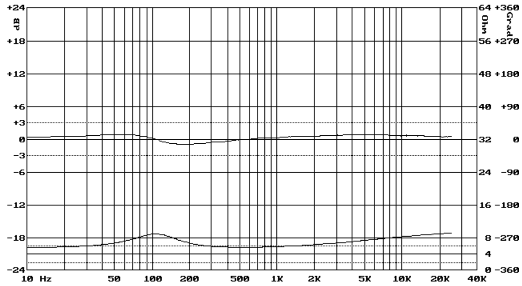


amplitude

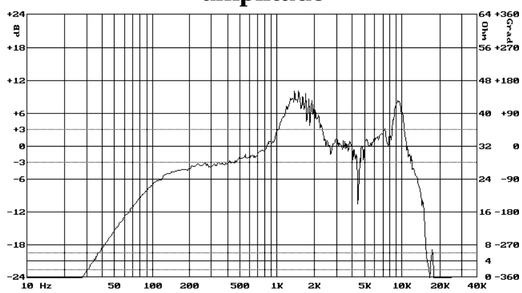


Paths with voltage divider

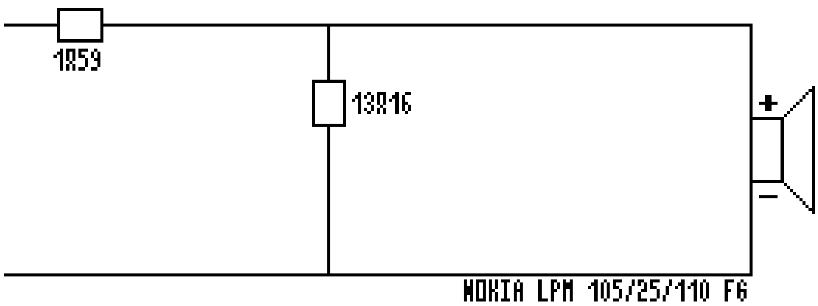
impedance and acoustic phase



amplitude



circuit



5.4.1.3 The menu item Acoustical Centre

The virtual place of sound origin is called the acoustical centre. This varies from driver to driver. You could say roughly that the acoustic centre lies somewhere near the moving coil.

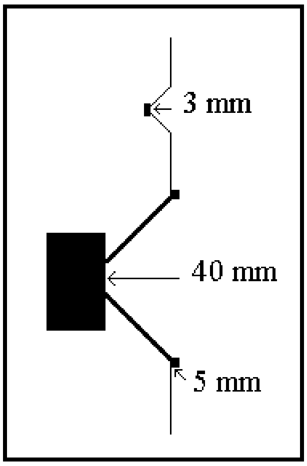
The relative position of the acoustic centre of each single driver when developing a crossover network must be taken into account for multi-way speakers. As shown for example in Appendix C, this can be managed either by measuring all loudspeakers in the case while maintaining the position of the microphone, or else by measuring the position of the acoustic centres for each individual driver. The later however is almost impossible with the normal PC protected audio measuring system in the listening room. Some manufacturers specify the acoustic centre in their specification leaflets. If the data is not available then you can do it yourself by following the above mentioned measuring methods or you suppose that the acoustic centre lies by the moving coil.

In AudioCad there are various input possibilities available:

- Place the acoustical centre of a driver in the loudspeaker database.
- Input the relative position of the loudspeakers in the crossover network mask.

The program works as follows:

The acoustic centre from the database and from the crossover network mask will be added per crossover network branch (driver). The smallest value (which in practice should be the tweeter) will be chosen as the reference point, i.e. it will be subtracted from all network branches. This data goes into the Simulation Summation curve (**add curves**). Under Simulation the effects of the acoustic centres will not be seen! Let's have an example for clarity:



- Input at the **Acoustical centre** field of the loudspeaker database:
You can measure the acoustic centre from the front edge of the basket. If the acoustic centre, seen from the front, lies in the direction of the moving coil a positive value has to be entered. If it quasi lies in front of the loudspeaker, which normally does not occur, a negative value will be entered.
Tweeter: 3mm (measured from the front of the basket)
Woofer: 40 mm (measured from the front of the basket)

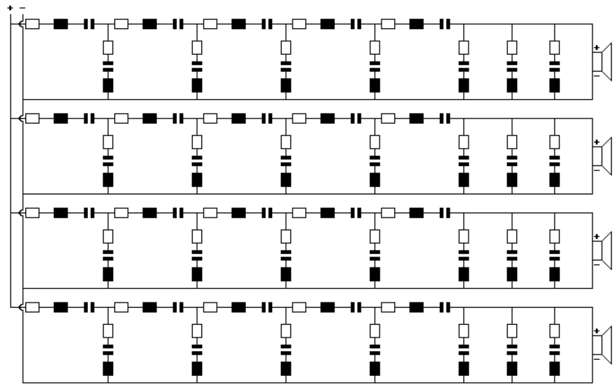
-Input in the **distance** field of the crossover network:
Tweeter: 0 mm (the tweeter is milled into the sound wall)
Woofer: -5 mm (the woofer is not milled in, the 5 mm refer to the strength of the basket, it is negatively entered as in this way the driver moves closer to the measuring microphone than the tweeter).

The program will add the values now:
Tweeter (3 + 0): 3 mm
Woofer (40-5): 35 mm

and subtracts the smallest value from all the values.
Tweeter: (3-0): 0 mm (reference level)
Woofer (35-3): 32 mm

The whole thing sounds very complicated, however in practice it is no more than that the acoustic centre of the woofer lies 32 mm behind the tweeter (as seen from the microphone position of the audio measurement system). The seemingly somewhat complicated way of input in different parts of the program (in the database and in the Crossover-mask) is necessary, otherwise data measured by the loud- speaker producer could not flow in, or else the user must always add it himself to the encasement position of the single drivers. If you estimate the acoustic centre of the single drivers, as described above, then you can of course enter the data immediately into the crossover network mask without having to deposit something for individual drivers in the database. In the above example you would enter 0 for the tweeter and 32 for the woofer in the transposition field.

5.4.2 The menu Simulation



Here you can realise almost any crossover network circuit, change constructional parts in the circuit, print the crossover network circuit and simulate the transmission functions of the single network branches or the overall system under ideal conditions or use of the measured data.

The menu item Circuit
Here you can view or print the circuit of the crossover network as a graphic, change the values of any constructional part and the polarity of the single drivers as well as insert new constructional elements into the circuit or delete single constructional parts. For the possibilities of the crossover network circuit graphic please see **Program operation in Graphics**.

The constructional part values will be displayed in the following format:

Resistance	: 8Ω2	corresponding to 8,2 ohms
Condensator....	: 3µ3	corresponding to 3,3 microfarad
Coils	: 1m5	corresponding to 1,5 millihenry

The menu item Sim Ideal
The amplitude, impedance and phase behaviour of the single network branches will be simulated without taking the measured data into account. This function will mainly be needed to define the target for the amplitude optimisation. It can be used to demonstrate the principle behaviour of crossover networks, however it will not produce usable results in the concrete construction of a crossover network! The graphic does not contain the influence of driver polarity and acoustic centres! For the possibilities of the graphic display please see **Program operation in Graphics**. A further possible application is the simulation of equivalent circuit diagrams. You can learn more about this in the Appendices.

The menu item Simulation
The amplitude, impedance and phase behaviour of the single crossover network branches will be simulated by taking the measured data into account. The graphic does not contain the influence of driver polarity and acoustic centres! For the possibilities of the graphic display please see **Program operation in Graphics**.

The menu item Add Curves
This menu item simulates the behaviour of the entire system. The result depends on whether the menu item **Sim Ideal** or the menu item **Simulation** has previously been chosen. The influences of the driver polarities and the acoustical centres will be taken into account in the simulation. For the possibilities of the graphic display please see **Program operation in Graphics**.

The menu item Export
Here the simulated data (curve paths) can be exported in AudioCad format. These files can be read in the AudioCad measured value database for further editing.
An application for this, is for example, the "overall impedance equalization" of a complete speaker including crossover network, which has been constructed with AudioCad. Develop your crossover network, export the impedance and electrical phase curves (summation curves), place a pseudo loudspeaker in the database which represents the whole box, and retrieve the exported curves for this, and initialise it as a woofer, for example. Then with the help of the optimizer (see the examples in the Appendices) you can determine the equalization.

5.4.3 4.4.3. The menu Optimize

In this menu the impedance and amplitude paths can be optimized with various target functions. Due to the possibilities for free definition for each target function respectively, almost no limits exist for the application of the optimizers.

The main application fields are:
-The optimisation of impedance correction links (RC or RCL-links)
-The 'overall - impedance equalization' of an already existing box
-The optimisation of the amplitude behaviour of single crossver network branches and of the entire system.

Furthermore there are many more imaginable application possibilities, especially when using data export (menu item Export in the main menu and in the **Crossover** mask), which should not be explained in more detail at this stage as the contents of the program description would dramatically increase. If you should have any specific application problems please feel free to call the author of the program during the support times.

The optimisation application is not always self explanatory in all points and assumes good previous knowledge of crossover network construction. **Please read this chapter completely before the first application.** Only then you can effectively use the numerous possibilities of optimisation. Before using the optimisation application for the first time you should also read the Appendices with the corresponding examples or carry out the examples on your PC.

For the working method of the optimizer: A non-linear optimisation will follow in practice, through the step-by-step changing of the input data (constructional part values) and the assessment of the results achieved by doing this. The program tests whether this causes an improvement or a deterioration. If it brings out an improvement then the optimisation will proceed in this direction. If a deterioration is caused then the change will be retracted. This assessment is being carried out taking into account the sum of all error squares (negative excursions should be dealt with the same importance as positive ones) in the observed frequency area. The smaller the sum of the error square the closer the actual curve path is to the objective function.
The AudioCad optimiser works out the crossover network circuit diagram from left (amplifier side) to right (loudspeaker side) and from bottom (bass) to top (tweeter). The constructional parts to be optimised can be freely chosen by marking them in the crossover network circuit diagram. Constructional elements which are not marked will not be changed by the optimiser!
Equally the frequency area to be observed can be pre-allocated. By a skilled selection of the frequency area (this should not be unnecessarily large) you can save on calculation time. On principle an optimisation will lead much quicker to the goal if only a few constructional elements are changed by the optimiser. If for example an RC and RCL are used in a network branch the optimisation for both equalization links should happen separately. The result will in general be the same, however by this procedure you can save a lot on calculation time!

5.4.3.1 Impedance optimization

The impedance correction networks obtained via the calculation formulas are in general in need of improvement. For example with RC-links this is due to the fact that the moving coil inductance is frequency dependent. With RCL-links inaccurate Thiele/Small parameters are mostly responsible for the non-optimal equalization. The impedance optimization also allows among other things the determination of an "overall-impedance equalization of the complete speaker".
The impedance optimisation will only happen for one crossover network branch (loudspeaker) respectively. If you would like to optimise several crossover network branches then carry out this procedure successively. If you insert an RC and an RCL-link for a loudspeaker then optimise the two correction links in succession rather than in one go. If the initial values are correct it will mostly work out in one go, however the optimisation will take much longer than if you execute the optimisations individually. As the optimiser involves the complete network branch it is also of importance that in the network branch concerned only voltage dividers, RC and RCL-links (parallel to the loudspeaker!) are "built in" at the time of the impedance optimisation. Possible existing high and low passes will hinder an efficient impedance optimisation!

Either a linear impedance path (allocated with Re and arbitrarily changeable) or an impedance path (measured and changed with the interface program to AudioCad format) which has been recorded with the measured data editor or measured with an audio measurement system or created by one of the AudioCad export functions can be taken as a basis for a target function.

For detailed application examples see Appendices.

5.4.3.2 Amplitude optimization

The amplitude optimisation can be done for one or more drivers of the system depending on the respective target function.

Target function linear amplitude:
The optimisation should be executed for all drivers of the system. Theoretically you can optimise a single, or for example two drivers, in a three way system. For the optimisation of a driver choose **<Single Driver>**, for the optimisation of several drivers choose **<All Drivers>** . "All Drivers" does not necessarily mean all drivers of the system. If you mark for example only constructional elements of the woofer and mid-range speaker in a crossover network circuit diagram of a three way speaker, only these two drivers will be optimised.

Target function amplitude path of measurement data editor:
The target function will be defined by a measured data file in AudioCad format. This can be

- created with the measurement data editor,
- measured with an audio measuring system and changed with the interface program into the AudioCad format or
- created with the help of an AudioCad export function.

Due to the free definition possibilities of the objectice function nearly all optimisations are possible. The description of all the possibilities would go beyond the purpose of this documentation. If you have a specific problem which you think could be solved with the help of the optimiser, then please call the author of this program during the support times. You can also fax your question under the same number.

Target function amplitude of Simulation Ideal:
With this target function you can determine the crossover network cut off frequencies yourself (with a linear amplitude path the optimiser will determine the cut off frequencies). As preparation enter the desired cut off frequency before the start of the optimiser as described above and execute **Sim Ideal**. This kind of optimisation will be carried out only for one crossover network branch. Optimise all crossover network branches of the speaker in succession.

Please see Appendices for the application examples of the amplitude optimization.

5.5 The menu Project

What should one do if one does not have the time to completely construct a speaker with the program in one go? You save a project file and leave the program. All the parameters necessary for the new automatic calculation of the already executed construction steps are saved in the project file. The functions are self-explanatory therefore only a short description is given here.

The menu item Load:
With this menu item a previously saved project is retrieved. Several project files are delivered for the program test. With these you can test the functions of the project editing.

The menu item Save:
This menu item saves a project file.

The menu item Delete:
This menu item deletes a project file after a previous security check.

The menu item Parts List:
A parts list for the actual project can be created here. The parts lists have the file extension **.STL** and will be saved in the projects directory as an ASCII file. Thus they can be changed with a word processing program or the MS-DOS Editor and can be used as an order list for construction parts.

5.6 The menu Utility

5.6.1 TS-Parameter-measurement

Thiele/Small parameters can be most easily measured with an audio measurement system. In case such a system is not available with a bit more work they can also be measured with the conventional method. For the measurement of the Thiele/Small parameters fs, Qes, Qms, Qts and Vas you will need the following measuring equipment.

Function generator (Sine frequency generator): The generator should be able to deliver voltages up to 10 volt undistorted. If your generator cannot do that you connect your amplifier between the generator and the loudspeaker. In this case of course if calculating the resistance you have to enter the output impedance of the amplifier instead of the one of the generator.

Voltmeter or multimeter with satisfactory display precision in the relevant frequency range (20-200 Hz alternating current).

Frequency meter: Not necessarily required if the generator has a satisfactory precision of the frequency adjustment scale.

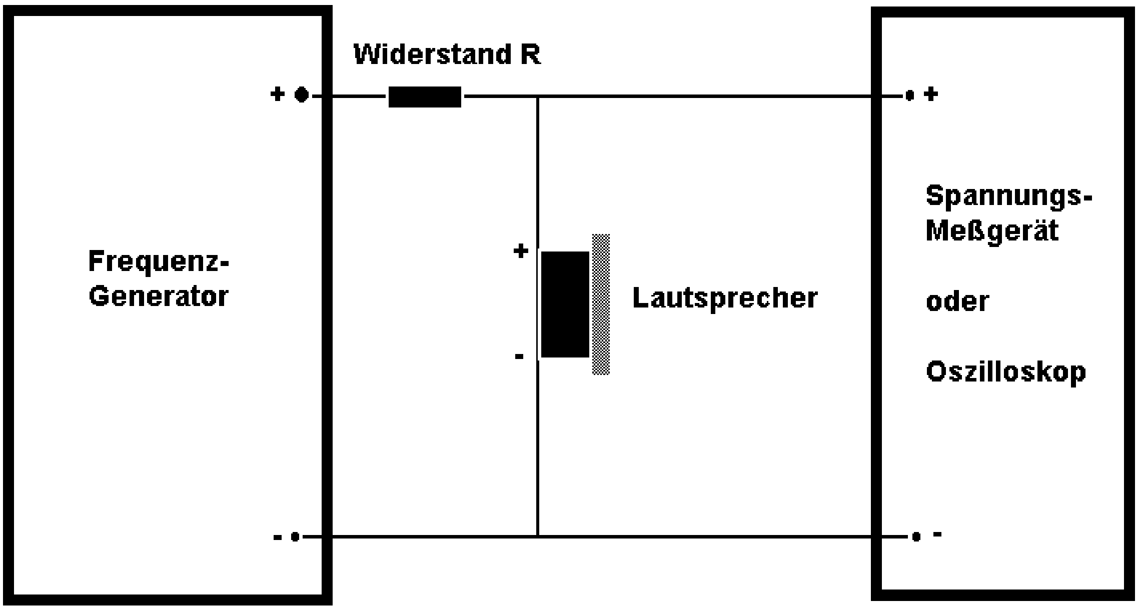
Instead of a voltmeter you can of course use an oscilloscope. You can also use it to check the measured frequencies if it has a calibrated time-slot pattern.

Description of the procedure:

The measurement can be done according to the weight or the case method. The case method is described in the following. When using the weight method the membrane will be loaded with a defined mass instead of building it into a case. For this the author uses a ring of plastacine which will be situated around the dustcap.

1. Measuring the direct current resistance (already possible with a very good multi-meter)

2. Create the following measurement circuit.



The resistance will be dimensioned according to the following formula:

R = 1000 ohms - Output resistance of the generator in ohms

E.G. The output resistance of the generator = 50 ohms - > R=953 ohms (950 ohms does not exist). A metal film resistance with 1% tolerance should be used.
Select a field of up to 200 mV at the voltmeter. The measured voltage drop which has been measured by the multi-metre over the loudspeaker will then correspond to the absolute value of the impedance measured in tenths of an ohm (200 mV correspond to 20 ohms). If the impedance to be measured exceeds 20 ohms exchange the resistor for a 10 KOhm resistor (10,000 ohms) and thereby you will achieve a measurement field of up to 200 ohms. The measurement circuit should be calibrated by a metal film precision resistor of 10 ohms with 0.1% tolerance. An example of such is available at the company
Radio RIM in Munich/Germany
tel: 049 89 55 17 02-0
order number 20-35-023 (Catalogue 1989) for about 5.-DM.

Now determine the resonance frequency of the loudspeaker in the free field, i.e. not built in. The resonance frequency is the frequency at which the loudspeaker will reach its maximum impedance. Typically the impedance paths increase until they reach the resonance frequency and then they drop again to the direct current resistance. Towards higher frequencies the moving coil inductance in general results in an impedance increase transgressing the value of the resonance frequency if this is not damped by appropriate means. (s. Impedance Emphasis and Attenuation).

3. Now find the frequency below the resonance frequency which has the impedance R1 previously given in the mask.

4. Now find the frequency above the resonance frequency which has the impedance **R1** previously given in the mask. The program will now calculate the parameters **Qes**, **Qms** and **Qts**. Thus the parameters fs, Qes, Qms and Qts are fixed. Print out the result by using the print key.

5. Build a closed test case with, for example, a net volume of 20 litres. Here you must pay special attention to the air permeability. The volume of the test box must be precisely known, as the calculation of the important parameter Vas is influenced by it. The driver is best built in with the magnet outwards. Thus, change the case-volume can be more easily calculated and you also save on the cable guidance. If the 20 litre case with very small loudspeakers does not lead to a significant rise of the resonance frequency the case volume can be reduced through a precisely defined brick or wood filling.

Carry out steps 1-4 again for the in built driver and print out the results. Calculate the comparative volume with the **Calculate Vas** program part.

5.6.2 Calculate Vas

After you have determined the Thiele/Small parameters fs, Qes, Qms and Qts then you can calculate the comparative volume Vas with this part of the program. The mask is self-explanatory. Therefore no further description is given here. After completing the Thiele/Small parameter measurement enter the determined values into the loudspeaker database.

5.7 The menu AudioCad

In this menu you will get information about the program version, copyright and the address of the author of the program.

6 Audio Measurement System Interface (KONVERT.EXE)

At the moment AudioCad Pro has the interfaces for the following audio measuring systems:

- AMS-PC	(Firm: Kemsonic / Bielefeld)
- AudioLab	(Firm: Kemsonic / Bielefeld)
- AudioTest Board	(Firm: Kirchner HiFi / Braunschweig)
- DAAS3	(Firm: HiFi Sound / Münster)
- DSA	(Firm: ADM-Engineerung / Nordhorn)
- LMS	(Firm: Audiomax / Heilbronn)
- MEPEG	(Firm: AES, Krefeld)
- MLSSA	(Firm: Harmonic Design / Vaihingen/Enz)
- IMP	(Firm: RCM Akustik)

These make it possible to automatically take over amplitude, impedance, and phase measured data (acoustic and electric) into AudioCad Pro measured-value databases. The interface program KONVERT.EXE functions as follows: The ASCII Export files generated by the measuring system will be read by KONVERT.EXE and converted into AudioCad compatible ASCII data format. This data format can be loaded in the ACPRO.EXE program (menu item **Import** in the loudspeaker database mask) in the measured values database. The AudioCad ASCII files with the file extensions **.AMP** (amplitude), **.IMP** (impedance), **.PHA** (acoustic phase) and **.PHE** (electric phase), will no longer be needed after the transfer has been completed to the measured value database, and can be deleted by the user. In case they are needed again for any particular reason then they can be regenerated from the ACPRO.EXE program by using the menu item Export. Here nevertheless the AudioCad internal frequency raster (512 support points from 10Hz -40 KHz, with logarithmic divisions) will be taken as a basis, instead of the proper format for the respective measuring system.

6.1 Connecting any audio measuring system

Since AudioCad Pro has no rigid frequency raster for the input of measured data, the quantity of measured data to be input is of no importance, and the AudioCad ASCII files are constructed as easily as you could imagine, so that theoretically it is possible to connect any PC supported audio measuring system. The interfaces mentioned above cover the complete German audio measuring system market for PC supported systems, at the moment. Should a new system be added, you are welcome to call the author (see Support and Updates). If you provide him with all the necessary information he will program the AudioCad interface for your measuring system.

6.2 Installation of the interface program

Open the menu item **Installation** in the menu **Utility**. A window will open in which you next select your audio measuring system, with either the mouse or cursor keys. Now click on **<Forward>** or tab across until **<Forward>** appears lighter, and then press the return key.

Tip for DAAS users: There are a few special points with the DAAS audio measuring system. Please read about this below.

Specify the paths and the microphone sensitivity in the window which follows.

Measuring system directory (all types): Enter the directory in which your measuring system leaves the ASCII Export files. E.g. With the Audio Test Board **C:\ATB\AC**

Database name (DAAS3 only): This information will at this time only be necessary for the DAAS3 measuring system. Please read about this below.

Micro-sensitivity (mV/Pa): Enter the microphone sensitivity in millivolts/pascal in the input field Micro-Sensitivity. This input will, at the moment, only be necessary for the AMS-PC measuring system (Firm: Kemsonic). When using another measuring system you can leave the field blank. Please note that you must use a decimal point instead of a comma. You should also only enter numbers and the decimal point, as the field content will not be checked, entering special symbols and letters will only lead to a program run full of errors. E.g. 42.80

AudioCad directory: Finally define the path, in the input field AudioCad directory, in which AudioCad should search for the measurement files. That is the path which you entered in the input field **Measurement-directory** in the AudioCad installation mask. Normally this is C:\AC8\MESS\

The data will be saved onto hard disk when you select **<Save>.** The menu item **<Backward>** goes to the selection menu Audio Measurement System Type, and **<Abort>** discards your inputs.

6.3 General description of file conversion

The menu **Convert** has various menu items, depending on which type of measuring system you are using, from which the files to be converted will be selected. Next the program asks for the file name (without the file extension - this will be added by the program), where the AudioCad compatible file will be placed. Please make sure that you always give in the same file name for the different types of measured values of a loudspeaker. Then you can retrieve all the measured data in the main program at the same time.

6.4 Generating measuring system ASCII format

The procedure depends on which measuring system you are using and will be described briefly below.

6.4.1 AMS-PC (Kemsonic)

The amplitude, impedance and phase measured data will be measured with the Kemtec measuring system and saved in ASCII (delimited) format with the menu item F1-F5 "File Export". Those with files saved in AMSPC have the file extensions: .PEC (amplitude), .IMC (impedance) and .PHC (acoustic phase). Please make note of which AMS-PC software version you have when you configure KONVERT.EXE. The AMS-PC version 1.32 and 1.33, which differ in the data format of amplitude and phase measurement files, will be supported by the program.

6.4.2 AudioTestBoard

To export the measurements from ATB use the function key F2, select either Export even or Export uneven and AudioCad.

6.4.3 DAAS3

The Thiele/Small parameters will be measured by DAAS and saved in a dBase-compatible database. DAAS saves amplitude, impedance and phase measured data in .FRQ format. This contains the amplitude, impedance and phase measured data (acoustic and electrical) of a loudspeaker, in one file. The generation of the database and the FRQ format is given in more detail in the DAAS manual.

Hints for the configuration of KONVERT.EXE: The DAAS interface is the most comfortable to handle at the present. This is mainly due to the fact that DAAS saves the Thiele/Small-parameters in a dBase-compatible database. DAAS can manage more of these databases in addition to the respective FRQ files. KONVERT.EXE is to be configured as follows:

Measuring system directory: names the directory in which the FRQ files are saved.
E.g. C:\HFS\FRQ\PROJECTS\

Database name: specifies the DBF file names including the complete path.
E.g. C\HFS\DBF\PROJECTS.DBF

AudioCad Directory: Finally define the path in the input field AudioCad directory, in which AudioCad should look for the measured data. That is the path which you entered in the input field Measurement-directory, in the AudioCad installation mask. Normally this is **C:**\AC8**MESS\.**

The rest of the input fields are not necessary. If after entering the configuration data you get an error message please correct your input data.

6.4.4 LMS

The user must save the files in APS1-format by using the menu item **Export** in the LMS **Utility** menu. As LMS uses the same file extension (.DAT) for amplitude and impedance measured data the user should, while assigning the file names, think up designations so that he can recognize which type of measured data they contain. The author recommends the following scheme:

A_<Driver name>.DAT	e.g. : A_ISO155.DAT	: Amplitude
I_<Driver name>.DAT	e.g. : I_ISO155.DAT	: Impedance

The conversion program KONVERT.EXE recognizes the format (amplitude and impedance) regardless of the file name and generates the corresponding output files which can be loaded in the measured values database of the AudioCad main program.
LMS only measures amplitude and impedance and no phases. Through the process functions contained in LMS the acoustic (amplitude phase) and the electric (impedance phase) phase can be calculated using a Hilbert transformation, and then also saved in the above file format. The following gives a short description of the procedure for calculating the acoustic phase for the amplitude. You get the electrical phase information analogously, whereas here naturally an impedance measurement curve will be used as a basis. This short description is for the LMS software version 3.05. In the following only the menu items to be select have been mentioned.

[L]ibrary Menu. Move to the measured curve with the cursor keys and mark it using F1.
Esc.-key
[P]rocess Menu
[F4] Move to the minimum curve with the cursor keys and then hit the **return** key.
Esc.-key
[U]tility menu
[E]xport Data
[F2] Move to APS1(.DAT) curve with the cursor keys and then hit the return-key.
Return-key
Give in the file names (for allocating names see above)
Hit the **escape** key twice to return to the main menu.

6.4.5 MEPEG

The measurement files with the file extension .MPG are created with MEPEG software. This is the normal MEPEG data format for saving amplitude and impedance measured data. Thiele/Small measured data are not processed by the interface!

As you cannot recognise from the file extension (.MPG), whether the file contains amplitude, impedance or Thiele/Small measurements, the author recommends getting used to a definite scheme when allocating file names. It could look like this for example:

I_<Driver name> .MPG	e.g.:I-ISO155.MPG	:impedance
A_<Driver name< .MPG	e.g.:A-ISO155.MPG	:amplitude
T_<Driver name> .MPG	e.g.:T-ISO155.MPG	:Thiele/Small

By doing this the files in the file selection box can easily be differentiated by KONVERT.EXE.

To measure the amplitude use the MEPEG menu item 2: SPL measurement. The measurement range must be between 20-20,000 Hz for all MEPEG measurements which are to be taken over by AudioCad Pro!

6.4.6 MLSSA

Amplitude and acoustic phase: For this file the user must enter the file extension .PEC in the MLSSA software, as KONVERT.EXE searches for files with this extension. If any file extension were to be used then KONVERT.EXE could not tell amplitude and impedance files apart. E.g. MESS.PEC

Example file:

"Sensitivity Bode Plot" dB SPL/watt (4.0 ohm load)"
"Hz" "Mag (dB)" "deg"
384.7064, 65.89388, -1.922165
399.5028, 69.28891, 9.898952
414.2992, 72.132, 10.57795

Impedance and electric phase: For this file the user must pre-allocate the file extension .IMC in the MLSSA software, as KONVERT.EXE searches for files with this extension. If any file extension were to be used then KONVERT.EXE could not tell amplitude and impedance files apart. E.g. MESS.IMC

Example file:

"Impedance Bode Plot - ohms"
"Hz" "Mag" "deg"
9.765625, 3.472225, 0.07227466
19.53125, 3.475806, 0.2204187
29.29688, 3.472461, 0.2132831

The generation of this MLSSA-ASCII Export format is explained in more detail in the MLSSA manual.

6.4.7 IMP

Amplitude and acoustic phase: Create a file in **.FRD** - format (see chapter **frequency** response of your IMP-Manual) using your IMP-Software.

Impedance and electric phase: Create a file in **.ZMA** - format (see chapter **impedance** of your IMP-manual) using your IMP-Software.

Impedance, electric phase, amplitude and acoustic phase: Create a file in **.ZFR** - format (see chapter **driver data** of your IMP-Manual) using your IMP-Software.

The generation of this IMP-ASCII Export format is explained in more detail in the IMP manual.

6.4.8 CLIO

Using your CLIO- audio measurement system create an ASCII-export file with the file extension ".TXT". The procedure is descripted in your CLIO manual in the chapter "file import export". The file should look approximately so:

Freq	Ohm	Phase
10.0	4.39	11.2
10.3	4.39	11.4

Install the AudioCad Audio-Measurement-System-Interface - Program:
- Choose as Audio-Measurement-System **CLIO**.
- In the menu **CONVERT** only one menu-item **CONVERT** appears. The program automatically recognizes the CLIO-file type (amplitude + acoustical phase or impedance + electrical phase) and converts it into the AudioCad format.

7 The measurement data editor (ACEDITOR.EXE)

The main program does not permit any manual input of measured data but can only retrieve measured amplitude, impedance and phase paths with loudspeaker measuring systems. Thus the measurement data editor has been created so that users without loudspeaker measuring systems can also use the program. This makes it possible to manually input measured data and save it on the hard disk in AudioCad compatible format. The curve paths installed in this way can be read by AudioCad as if they had been installed by a measuring system.

The measured data editor is called up via the main menu. After the program has been loaded a selection menu appears in which you click on the desired type of measured value. Thereupon you will be asked whether you want to create a new file. If you still have not taken any data for the actual loudspeaker then select <Yes> here. The program will now ask you for a file name. Enter any file name here without the extension (that which lies after the point). The file extension will be added by the editor. The editor now has a look to see whether such a file already exists. If this is the case you will be asked to input another file name so that previous data is not accidently overwritten.

If you have already taken measured data for the loudspeaker and would like to fine tune it select <No>. Thereupon a file selection box will appear in which you can mark the file to be edited. This will then be loaded in the editor.

Now the editor itself will appear. The editor contains the following windows.

The measured data window: Your inputs will be displayed here, sorted by decreasing frequencies for control reasons.

The frequency window: Enter the desired frequency of the measured values here. Then use the tab key to position the cursor in the measured value window.

The value window: Here you enter the measured value (amplitude in dB, phase in +/- 180 degrees or +/- 360 degrees, impedance in ohm). The input of amplitude values should be entered relative to the degree of efficiency (SPL). If for example the degree of efficiency (SPL) of the chassis is 90 dB then the measured value of 87 dB will be taken as -3 dB.

For inputting the measured values you are not bound to a fixed frequency raster. All intermediate values which are not entered will be interpolated from the existing values. The data for 10 Hz and 40 kHz should however be entered. If you do not do that they will be preallocated as described below. If you do not stick to the measurement area (you measure for example from 20 Hz to 20 KHz) the program will use the following pre-allocations for the missing frequencies (10-20 Hz and 20-40 kHz).

Amplitude	: -99dB
Phase	: 0 degrees
Impedance	:Direct current resistance (Re)

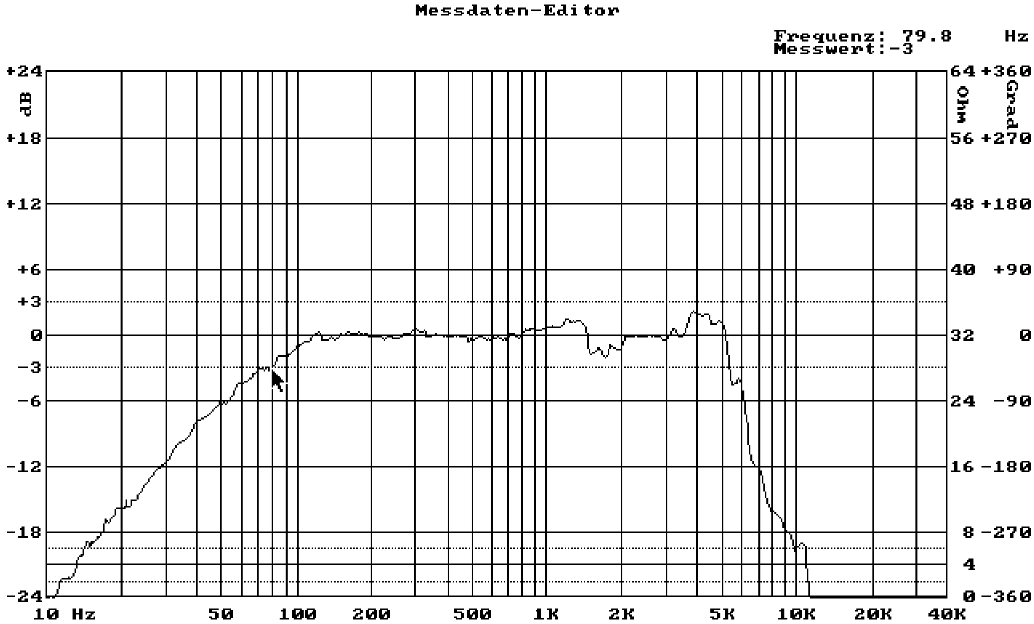
The menu item New: After inputting the measured value simply hit the return key or click on the menu item <NEW> with the mouse. By doing this the measured values The measured value which have just been taken will be entered in the MEASURED DATA window.

The menu item Delete: Click on the measured value which is to be deleted, with the mouse, or select it with the cursor control keys. Select the menu item Delete using the **Tab** key or the mouse. The measured value will be deleted from the measured data window.

The menu item Forw: turn forwards one page in the measured data window.

The menu item Backw: turn backwards one page in the measured data window.

The menu item Graph: Your input data will be graphically displayed. By clicking with the right mouse key in the graphic the frequency and the corresponding measured value will be displayed. With amplitude measured data you can move the curve up or down by entering the corresponding SPL. This however will in general only be necessary when re-editing a curve determined by the audio measurement system.



The menu item Save: Your input data will be saved in the previously selected file.

The menu item Save as: After inputting a new file name the measured data will be saved in this file.

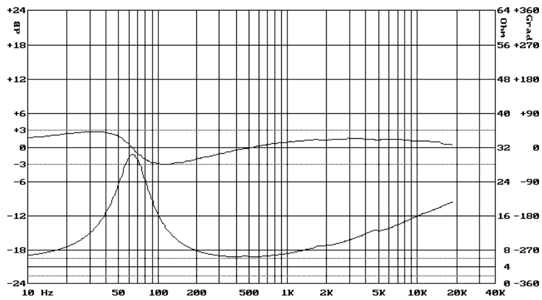
The menu item Cancel: With this item you throw away your input data. The selected file will remain unchanged.

8 Abbreviations table

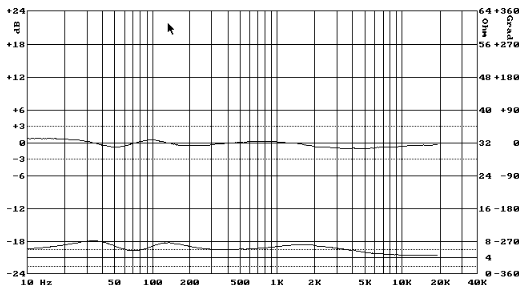
Bl:	Power factor in Newtons/Ampere
7cm:	centimetres
Cms:	Compliance in mm/Newton
EBP:	Efficiency-Bandwidth-Product after Small in hertz: Through this the suitability of a driver for a certain cabinet construction can be determined. Closed cabinets and bass reflex boxes need EBP between 50 and 100 hz. With a good horn driver the value should be over 150 Hz.
f3:	Frequency at which the sound level lies approximately 3 decibels below the average level.
Fb:	Tuning frequency of the bass reflex channel in Hz.
fc:	Resonance frequency of the speaker built into the box.
fs:	Resonance frequency in hz of the non-built in loudspeaker.
gr:	gram
Hz:	hertz (oscillation per second)
l:	litre
Le:	Moving coil inductance of the loudspeaker in mh
mH:	milli-henry (10 E-3 henry): Unit of inductance of coils
mm:	millimetre
Mms:	Mass of the loudspeaker membrane in grams.
No:	Reference efficiency in percent
Qes:	Electrical quality of the loudspeaker
QL:	Overall quality of the case and bass reflex channel (depending on the box size) up to 30 litres: QL=10: up to 70 litres: QL=7 above 70 litres: QL=5
Qms:	Mechanical quality of the loudspeaker
Qts:	Quality factor of the loudspeaker
Qtc:	Overall quality of the loudspeaker built into the box (try to get 0,707)
Re:	Direct current resistance in ohms
SPL:	Reference sound pressure (Sound pressure level) in dB, with the initial power of 1 watt measured at a distance of 1 metre
Vas:	Equivalence volume of the loudspeaker in litres
VB:	Net internal volume of the box in litres
µF:	microfarad (10 E-6 farad): Unit for the capacity of condensers.

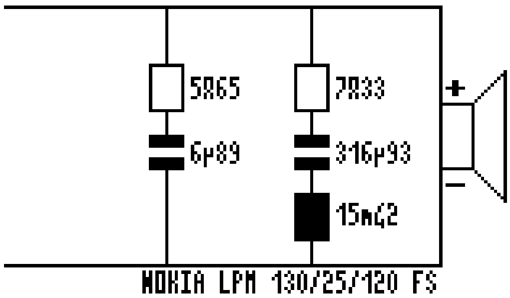
The equalization of the basic resonance and the inductive impedance increase of a mid-range speaker will be described here.

The following graphics show the impedance and the electrical phase respectively. The following is the initial situation:



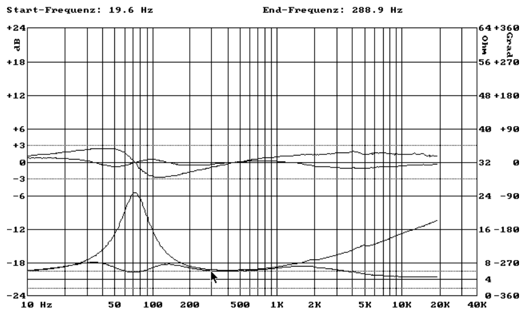
After the calculation of the impedance correction circuit (RC- and RCL-link) with the help of the menu item **Impedance Emphasis & Attenuation** in the menu **Edit** the impedance path will be as follows:



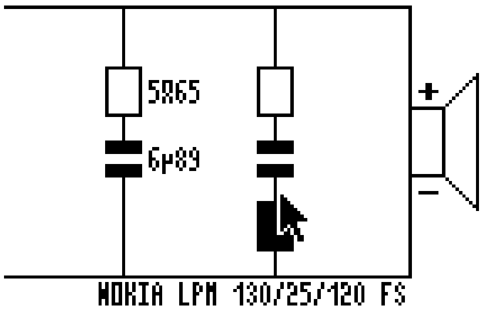


Now activate the menu item **Optimize** in the **Optimize** menu. The program will ask for the type of optimization. We chose **<impedance>** and the target function **linear impedance**. In the following input field enter the impedance value for which the optimization should take place. This is usually the direct current resistance of the driver, which has been preallocated.

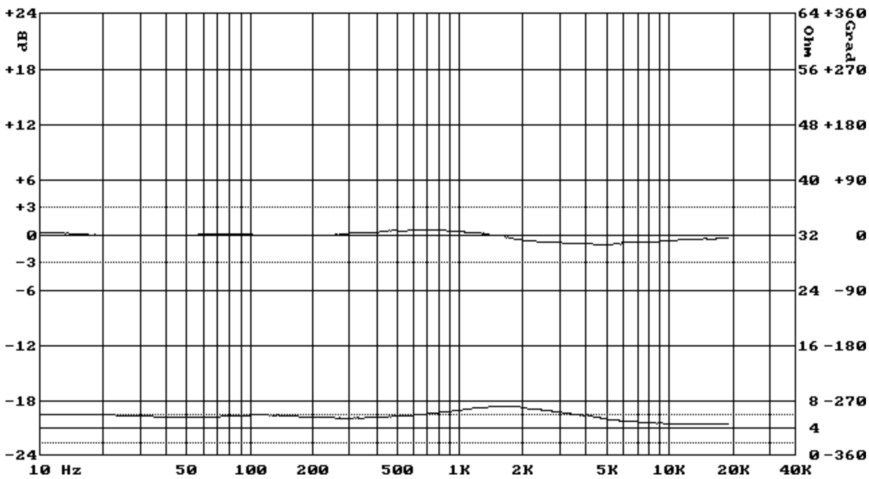
Now a graphic appears in which you can mark the frequency range for optimization using the right mouse button. For better recognition of the relevant ranges the non-equalized curves will apppear in grey, in addition to the actual impedance- and electrical phase path. As initially we want to optimize the RCL - link we chose the frequency range between 20 and 300 hz.



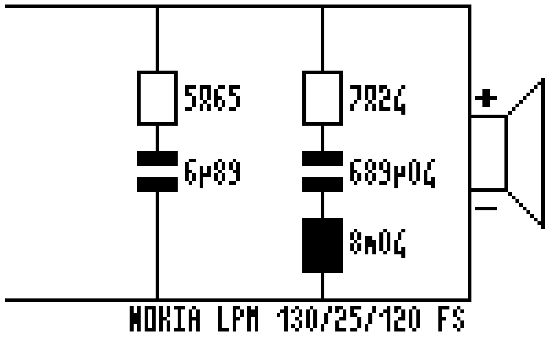
In the crossover circuit diagram which now appears click on all the constructional elements of the RCL - link. The marked constructional parts will appear red and the value will disappear. All the non-marked constructional parts will not be altered by the optimizer!



After clicking on the switch field **Exit** the program will ask for the number of sim-points. If you enter 8 then 64 dots per curve will be simulated, similarly if 4, 2 or 1 are entered then the resulting dots per curve will be 128, 256 and 512 respectively. An optimization with 64 dots is already so precise that in general a longer calculation time needed to achieve a higher resolution is not worth the trouble. Therefore keep the default value, click on **<OK>** or press the return key. The optimization process will now be activated. At the beginning the sum of the error squares of the initial situation will be shown. This is a measure for the degree of the optimization. After the optimization has finished this figure should be as small as possible. However, in general the value 0 will never be reached! When the optimizer is ready it will display in a message box the sum of the error squares. Confirm this by clicking on **<OK>**. Now the impedance path will be as follows:

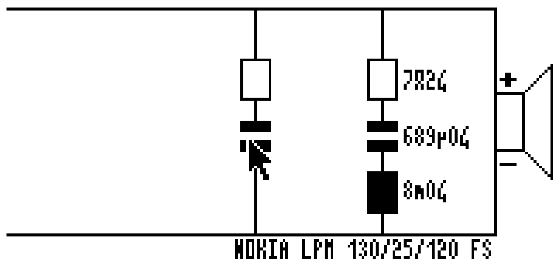


Circuit

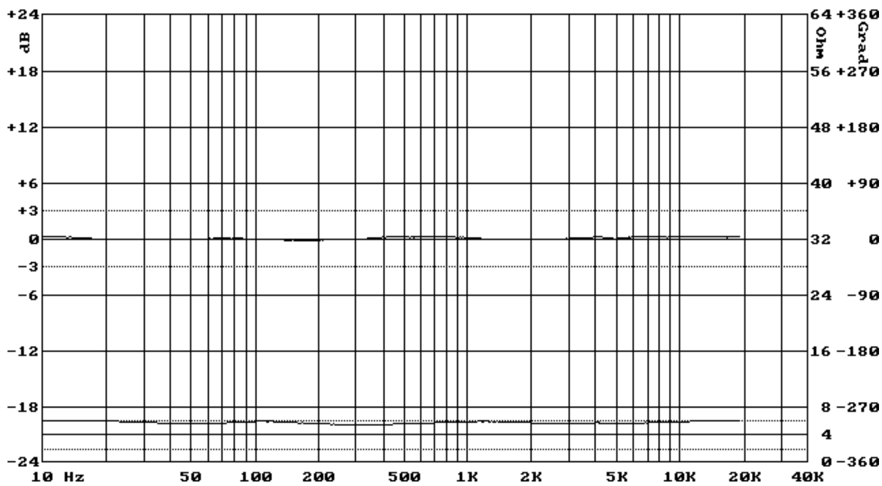


In the next step we will optimize the RC-link. Only the steps differing from the ones previously mentioned will be described below.

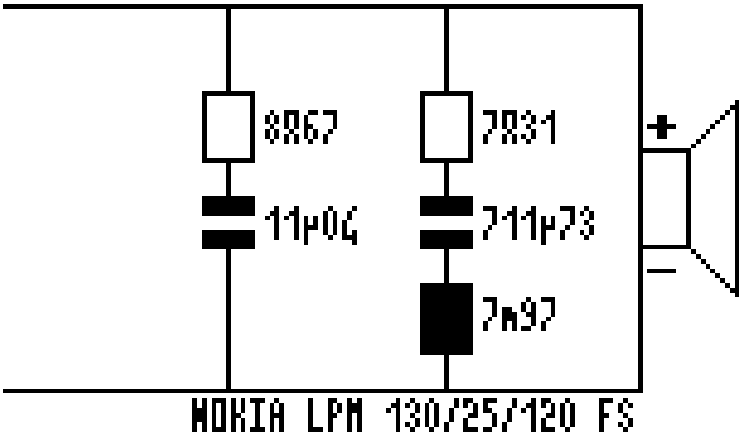
Selection of the frequency range: For the RC - optimization mark the range from 400 to 15000 Hz.
Marking of the constructional elements: Mark the resistance and the condenser of the RC - link.



At the end of the optimization process the following impedance path will be displayed.



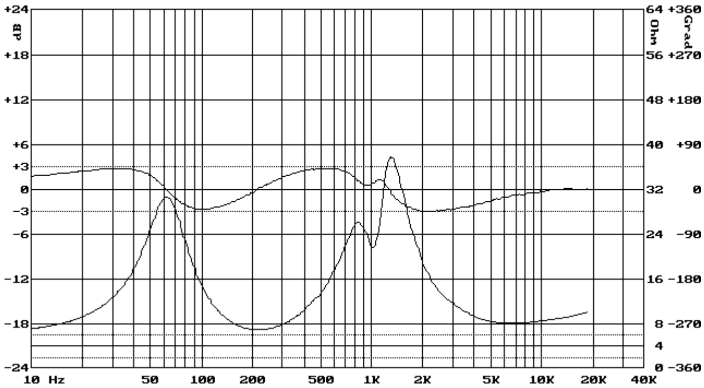
Circuit



The constructional part values determined by the optimizer will usually not be available on the market. However, you can change the constructional part values in the crossover network circuit diagram accordingly and for example optimize the coil again for the modified constructional parts, as the inductance is changeable for any value by unwinding the coil. The required resistances and condensers can be built together by joining several smaller values (resistances in series, condensers in parallel).

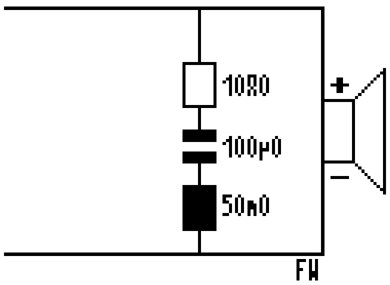
This seems to be very complicated and time consuming but with a little exercise the procedure can be done on an 80486 PC in about 2 minutes.

Low priced amplifiers normally have problems with impedance and phase angles which change over the frequency range. This problem can be solved by a subsequent impedance linearisation of the whole box including the crossover network. In the example a closed two-way box will be equelized with the help of the impedance optimizer. The resulting circuit can also be built externally onto a small platine which has to be connected with loudspeaker cables on the input side, and with the box on the output side. The impedance- and electrical phase measured data will be shown as follows:

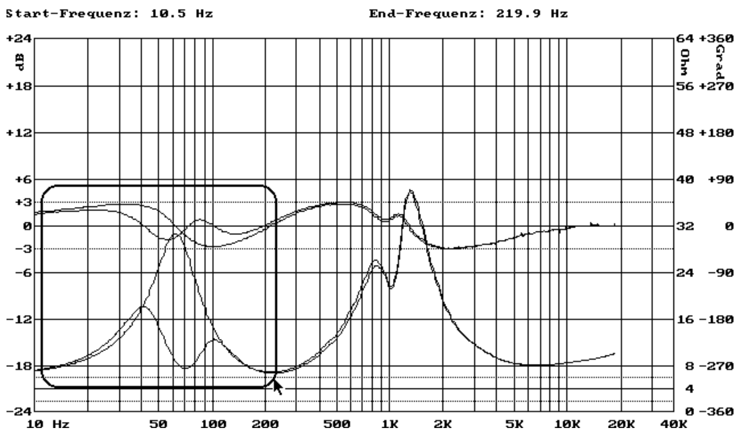


The data can be measured with an audio measuring system. If however the crossover network was constructed with AudioCad then the data can be generated with the help of the menu item **Export** in the **simulation** menu of the crossover network **(**click on **[ ] impedance**, **[ ] el. phase**, **[ ] crossover-simulation: sum of all speakers** and enter a file name). In the following create a pseudo-driver in the loudspeaker data base which represents the whole box. Transfer the measured data to this driver and initialise it for further calculations e.g. as a woofer.

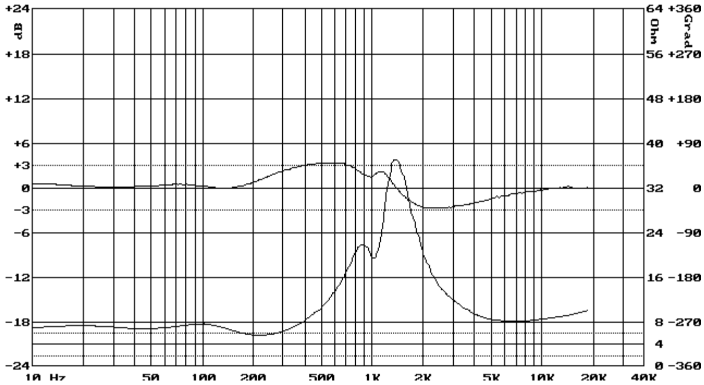
Then call the crossover network simulation and allow the crossover network circuit diagramm to be shown. Click on the switch field **Show All**. Now enter an RCL-link, whereby you can pre-allocate for example the following values: R:10 ohms, C: 100 microfarad, L: 50 millihenry. Next click on **Hide**. The connection diagram should now look like this:



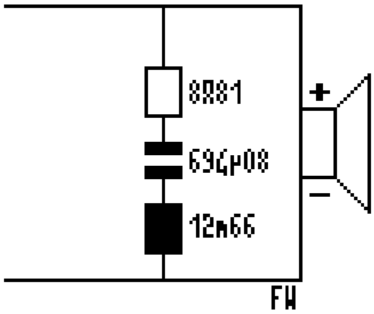
Go now as described above to the impedance optimization, enter a linear impedance path of the example 7 ohms as a target function and mark the first impedance peaks of the original curve (grey) as shown in the following picture:



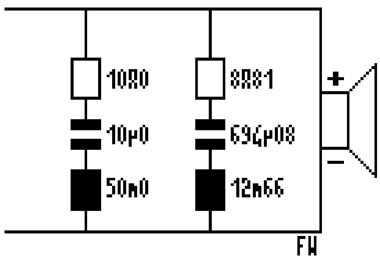
Mark all the constructional elements of the RCL-link in the crossover circuit diagram. The subsequent optimization produces the result:



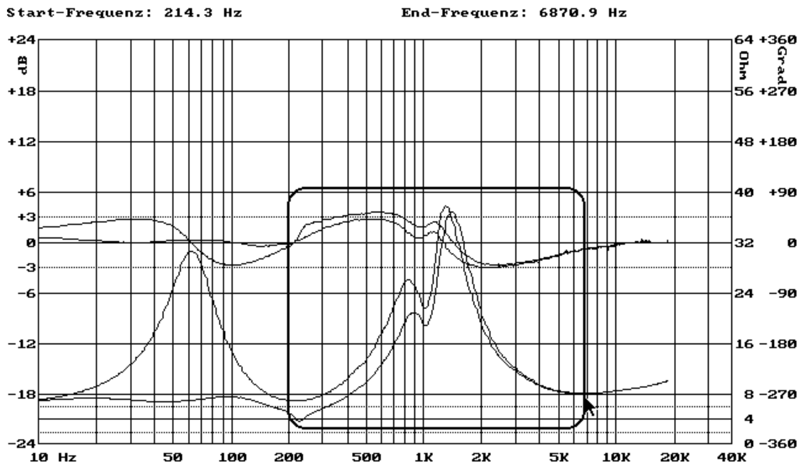
The first 'peak' has now disappeared. The circuit diagram of the equalization looks like this:



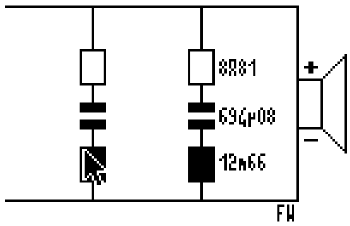
Now enter a further RCL-link into the switch diagram display. As the resonance peak, which is to be equalized, lies at a higher frequency enter for R 10 ohm, for C 10 microfarad and for L 50 millihenry for example.



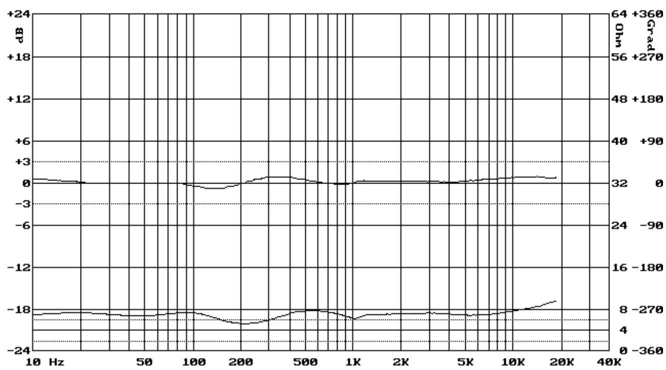
Now we go back to the optimization. Mark the following frequency range:

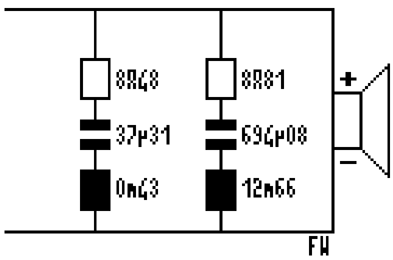


Now only mark the second RCL-link for the optimization:

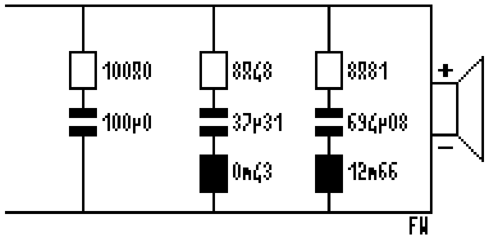


The result now looks pretty good:

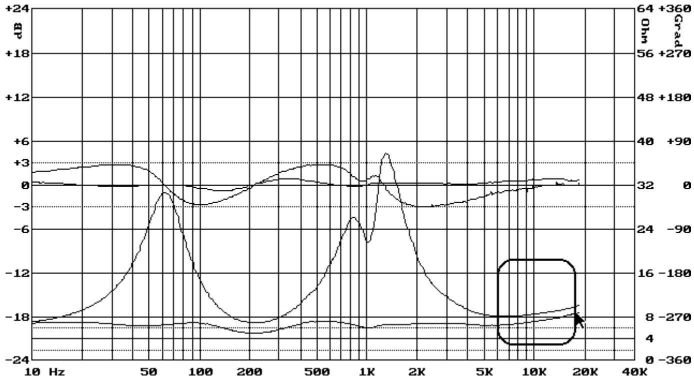




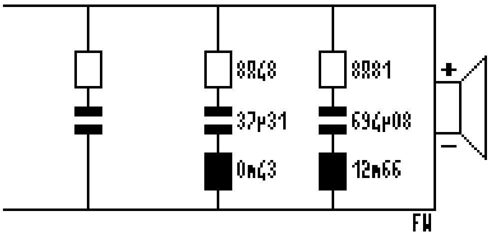
Perfectionists can now equalize the small impedance rise which begins at 6 KHz. To do this insert a RC-link in the switch diagramm. You can take 100 Ohm and 100 Microfarad as default values:



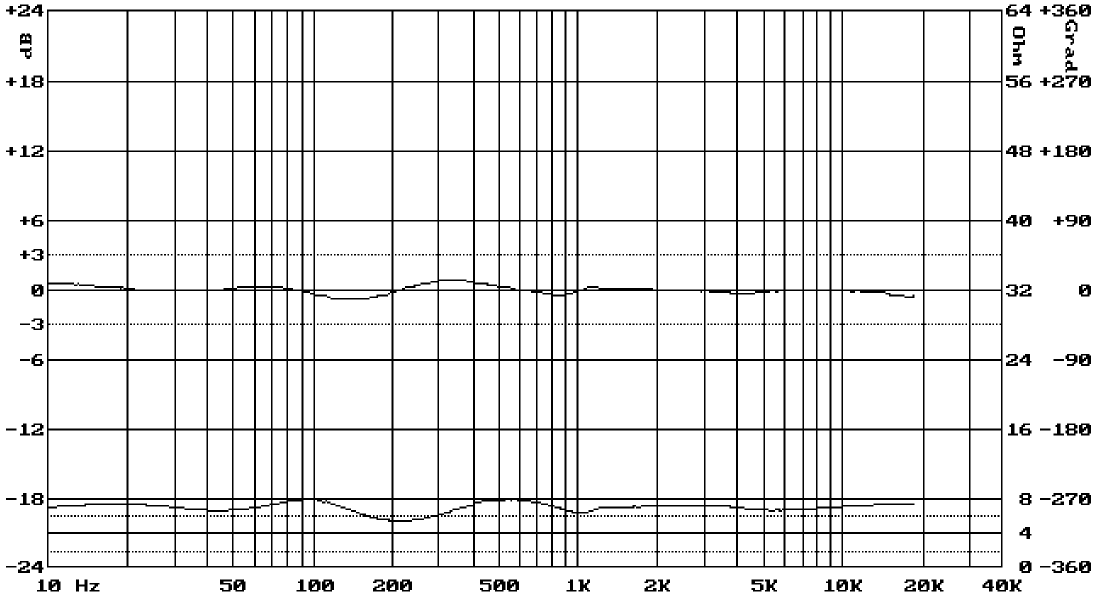
After this optimize again and select the frequency range:



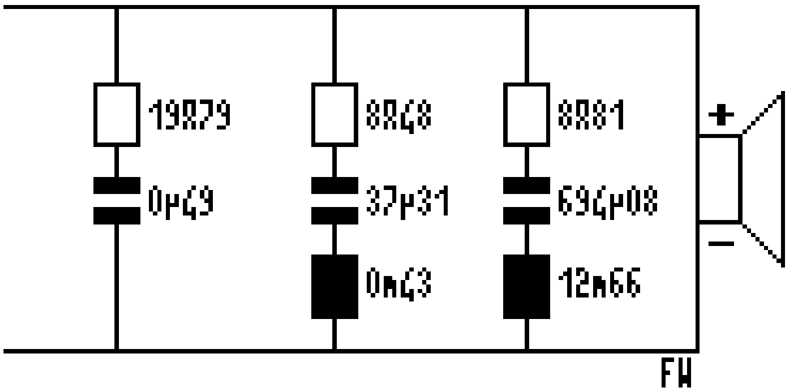
Only mark the RC-link in the switch diagramm:



A perfect equalized impedance path will now be the result, which should no longer cause problems even for a 'feeble' amplifier:



The box will be equalized with the circuit shown in the following diagram:



This procedure requires about 2,5 minutes! A further tip for the starting values of the equalization links: Should the optimization not produce an improvement after about 20 seconds then your starting values are quite wrong. In this case quit the optimization using the function key F9 and change the starting values accordingly. The specified calculation times refer to a PC with an 80486 DX-33 MHz processor. 'Smaller' PCs require a correspondingly longer time.

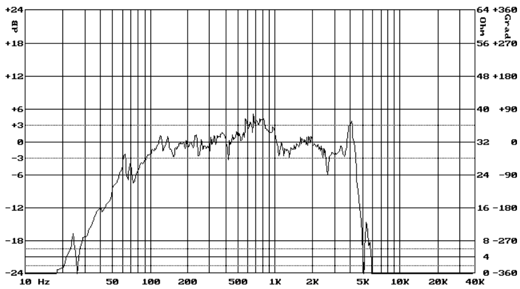
The development of a crossover network for a two-way speaker system will be described here. Before the construction of the crossover network the loudspeakers were built into the cabinet without a crossover network and connected in such a way that they can be controlled individually from the outside. The measuring microphone was placed about 30 cm away just below the tweeter membrane. It is important that the position of the microphone is not changed during the whole measuring procedure and that while importing the data into AudioCad the **SPL** value and the **0 dB point** for both loudspeakers are taken the same values. This procedure does not require the input of the acoustic centres of the individual drivers, as they automatically come in through the measuring with a stationary microphone position.

Remarks about the microphone position: As the author does not possess a reflectionfree room (as will be the case with most AudioCad users) the microphone position during measuring had to be determined empirically. All positions are possible in which the measured acoustic phase data of both loudspeakers are more or less plausible. In the transmission range of a loudspeaker (see below) no large phase jumps must appear during measuring. However, the whole thing must also not be exaggerated: Phase jumps outside the transmission range of the chassis do not play a big role, as there is no transmission of recognizable sound anyway.

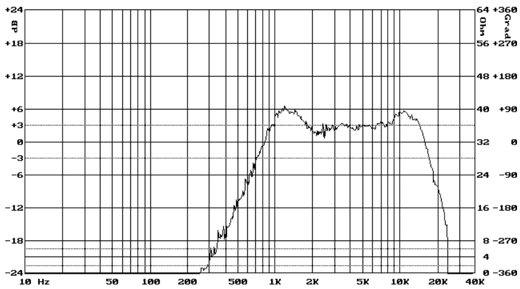
Measured Data

Woofer	Tweeter

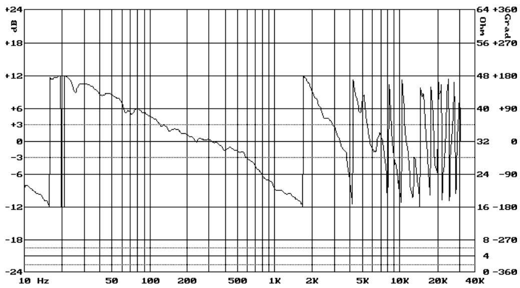
Woofer — Amplitude



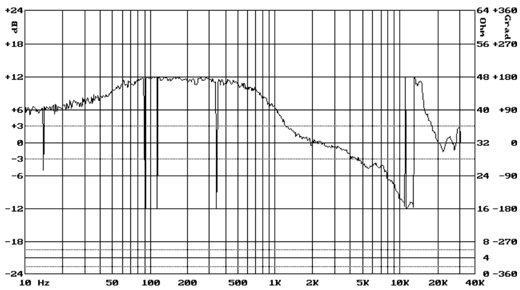
Tweeter — Amplitude



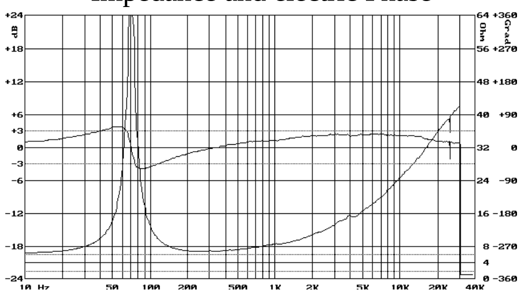
Woofer — Acoustic Phase



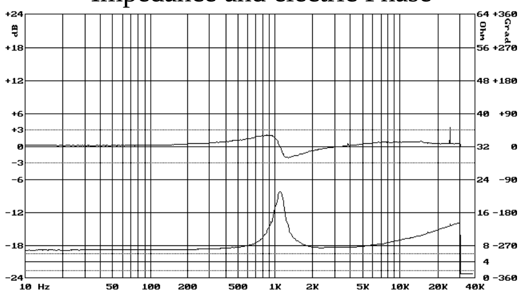
Tweeter — Acoustic Phase



Woofer — Impedance and electric Phase

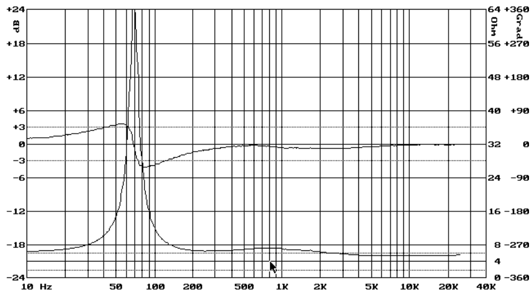


Tweeter — Impedance and electric Phase

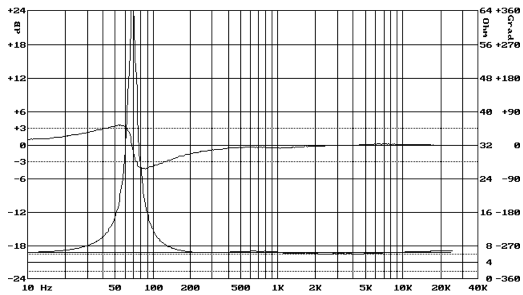


The two drivers are now being initialised for the calculation (chose the respective driver in the data base mask, leave the mask via **Exit** and answer the question which will follow with **Yes**) and then start the item crossover in the crossover menu.

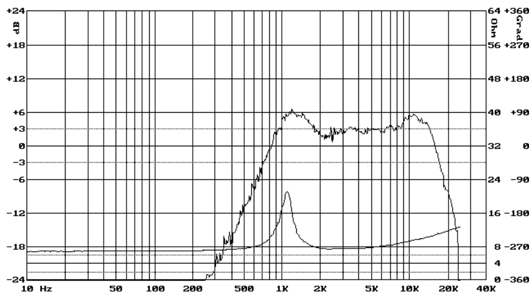
Now call the menu item **Impedance Emphasis & Attenuation** in the menu **Edit**. Click on the woofer and on **<OK>**. The program will now show the measured data of the woofer as shown above. By switching on and off the respective curve paths and clicking on the box **Plot** they can be viewed individually which facilitates the overall handling. The impedance path which rises towards the high frequencies is disturbing with regard to the woofer. Therefore leave now the graphic and re-start the **Impedance Emphasis & Attenuation** while again marking the woofer and **RC** as well as clicking on **<OK>**. The following picture will appear:



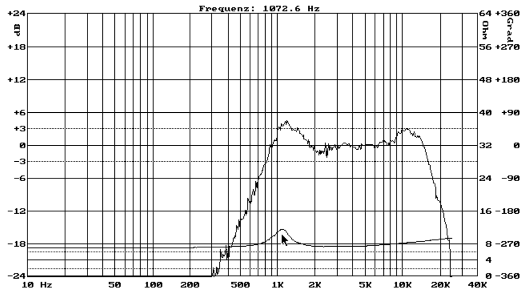
Of course this is not satisfactory as the equalization was not very good (see arrow). Therefore optimize the impedance path of woofer as described above. After the optimization the following result will appear:



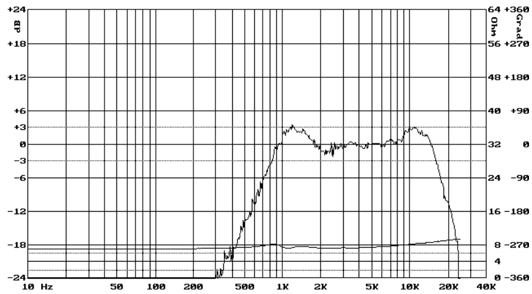
Next equalize the impedance of the tweeter. The measured amplitude and impedance data will appear as follows:



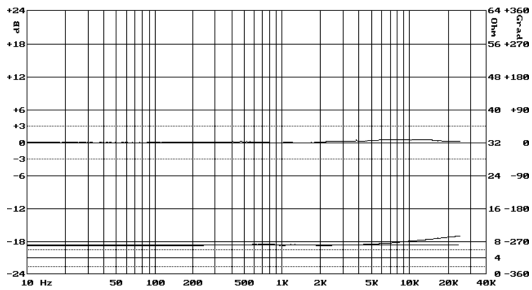
Note that the tweeters sound pressure is approxiamately 3 dB higher than that of the woofer. Therefore initially select **[ ] Attenuation** and enter 3 in the input field **-dB**. The program does not mind whether this is entered as a positive or negative value. Do not forget the tweeter in the selection box otherwise you execute the whole procedure for the woofer and will override the previously optimized data. After the attenuation the amplitude and impedance of the tweeter will be as follows:



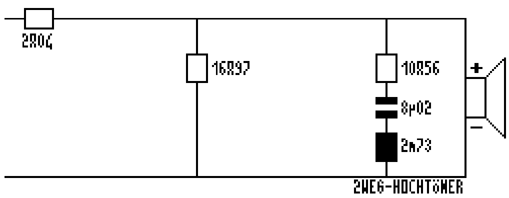
The impedance peak at 1100 Hz is still a little bit disturbing. Therfore call the **Impedance Emphasis & Attenuation** again, mark the **tweeter**, **RCL**, [ ] **Attenuation** and enter 3 again in the field **-dB**. The result:

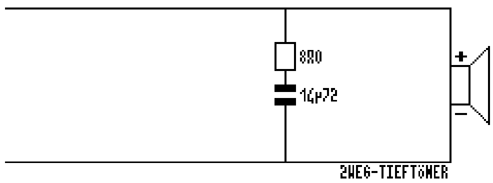


As you are a perfectionist now you will optimize the RCL-link which will lead to the following impedance path:

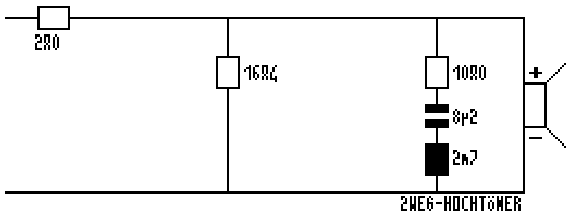


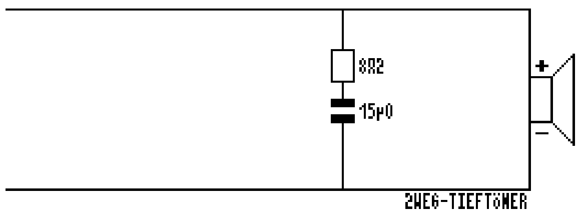
Circuits



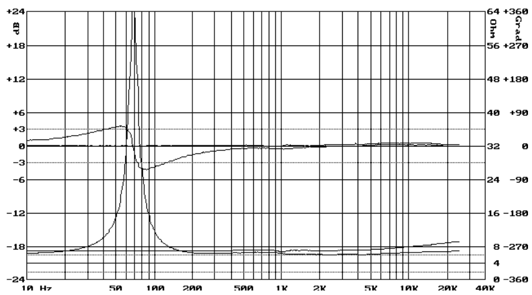


As the constructional part values are not customary in trade alter these as shown below. By doing this the values can be combined by using several individual values (resistors connected in series and condensers connected in parallel):

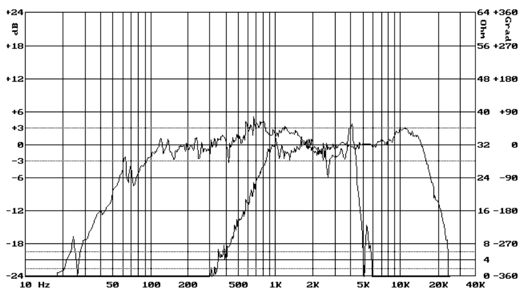


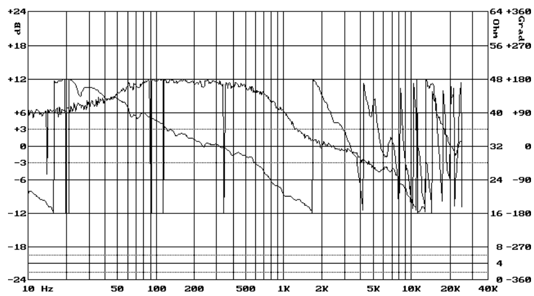


The curve paths resulting from this alteration can be seen with the menu item **Simulation** in the **Simulation** menu:

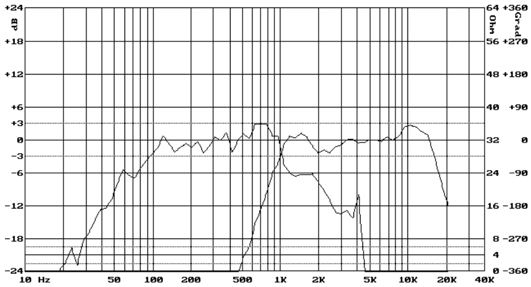


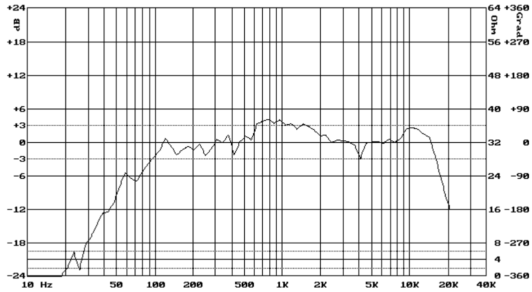
Only the impedance and the electric phase paths were displayed here. Of course you can also look at the amplitude and acoustic phase:



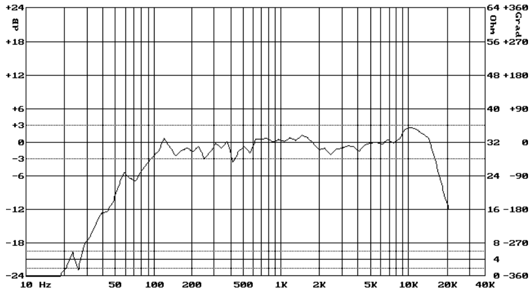


Hereby the impedance equalization and the attenuation is finished. A practised user can complete everything in about 5 minutes. Now turn to the crossover network itself, that is the high and low passes of the crossover network.
The box constructed at this point will be connected by the author to a stereo TV with a 2 times 20 Watt-amplifier. Therefore it does not have to be especially stable, but it should however have a linear transmission range. Under these presuppositions a very low cut-off frequency can be chosen for the tweeter without causing risk of damage. However, if this box is connected to a stronger amplifier with higher levels then this is not advised! Therefore a simple 6dB circuit will be selected as a crossover network topology. Activate the menu item **Cut-Off Frequency** on the edit menu and enter for example 1000 (Hz) as the cut-off frequency for the woofer, **<6 dB>** as the flank and **<Butterworth>** as the characteristic. Repeat this for the tweeter and look at the **simulation**.

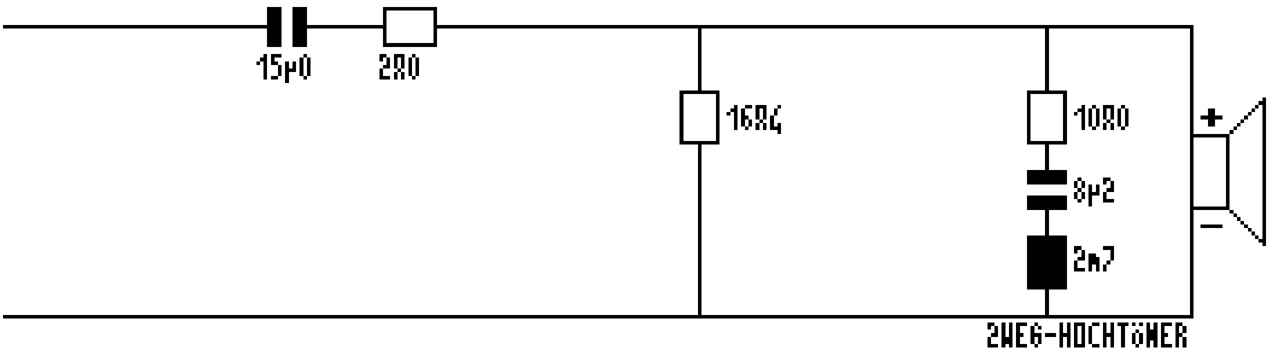


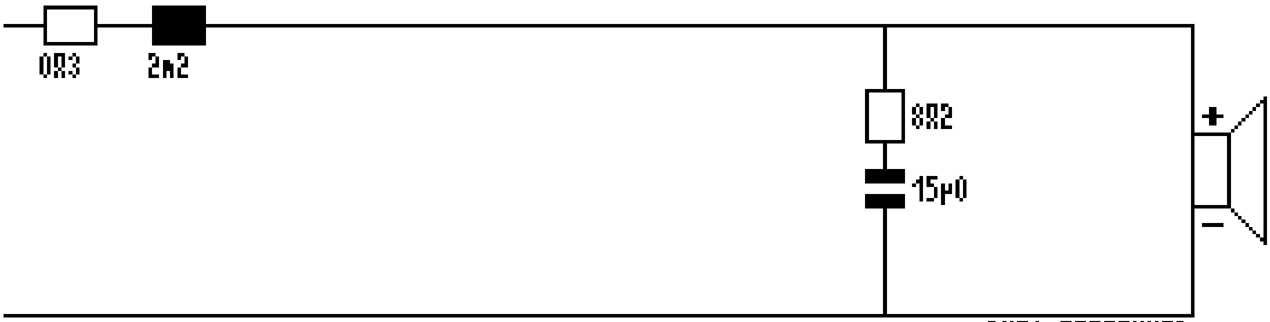


Now start the amplitude optimization. Select the menu item **optimize** in the **optimize** menu, click on **<amplitude >**, **( ) linear amplitude**, **<all speakers>**, mark the frequency range from about 500 to 8000 Hz and click in the crossover network diagram on the condenser (high pass of the tweeter) and the coil (low pass of the woofer). The optimization will now start and after a short period will produce the following result:



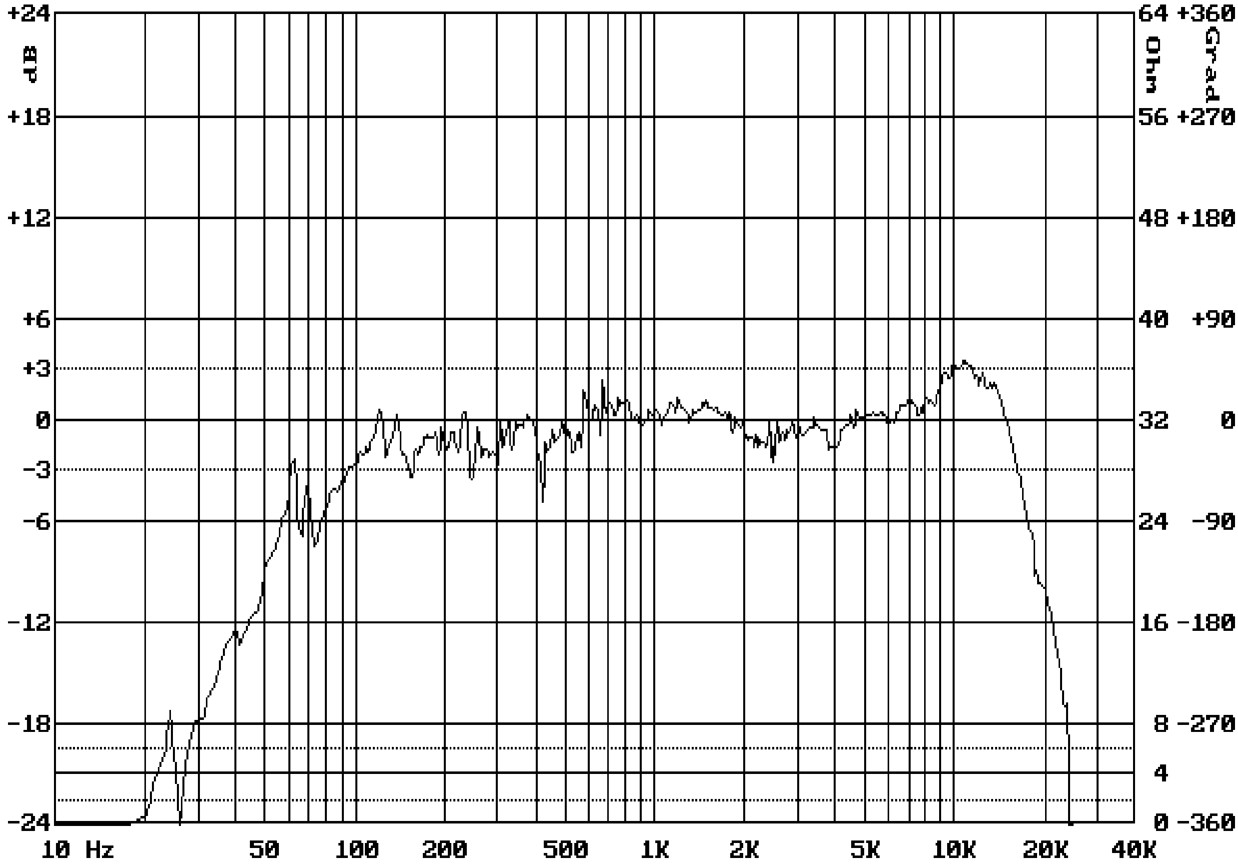
Now call up the crossover network diagram and alter the constructional part values as follows:





Now the crossover network construction is complete. Save this as a project and print the results. You should notice that this is actually not an especially good speaker and that the crossover network was also not created in this form. In reality a 12 dB crossover network was created. However, the representation of it will not be done here as the above construction is simply an example. It is now up to you to develop a better crossover

network for the two-way speaker. The measured data of the individual drivers in the cabinet is contained in your database.
Finally the amplitude path of the above system will be shown in high resolution (512 simulation points per curve):



Okay- lets go ! Start your PC and do it better!